Background Story

CORE Processing

Dual-path compression system achieves best clarity, comfort and sound quality

Introduction

It has long been known that amplification alone is not enough to enable someone with hearing loss to not only hear everything clearly again, but also as naturally as possible. The human hearing system has a distinct dynamic range within which sounds can be detected and remain comfortable. In people with impaired hearing, this dynamic range is reduced. This means that they perceive loudness differently. The result of simple (linear) amplification is that some sounds are still too soft, stay below the hearing threshold level and therefore remain inaudible. Conversely, other signals may be amplified too much, causing discomfort or even pain. While many signals will indeed be reproduced within the residual hearing range of someone with hearing loss, they may not be perceived correctly in relation to the person's changed dynamic range nor sound natural. In order to resolve this problem, modern digital hearing systems use complex compression systems. These change the input signals so that the amplified sounds are audible, remain comfortable and the person with hearing loss perceives sounds as naturally as possible. Phonak has continued to develop and improve the compression systems used in its hearing systems. Now, products based on the CORE platform employ the industry's most advanced adaptive "dual-path" compression system to ensure audibility, intelligibility and natural sound quality.

Why compression?

The human ear has a clearly defined dynamic range within which it is able to detect and manage acoustic signals. This dynamic range extends from the lowest detectable frequency to the highest that is audible, and from the softest input signal that is only just audible, to the pain threshold (Figure 1). Within certain limits, the range can vary greatly from one person to another. In the scientific literature, this range is usually defined from about 20 Hz to 20 kHz in the frequency domain and with intensities ranging from 0 dB to 120 dB. The sounds of speech lie within this dynamic field.

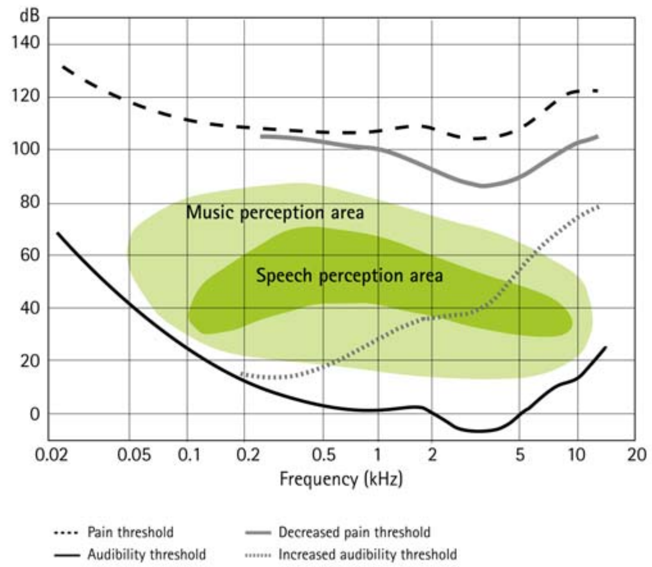


Fig. 1 Representation of the normal human dynamic range and reduced dynamic range resulting from hearing impairment

Figure 1 also shows how for a hearing impaired ear, the dynamic range can be reduced as a result of increased hearing thresholds and, often, lowered discomfort limits. With hearing loss, the way in which loudness increases is also different from a normally hearing ear (figure 2).

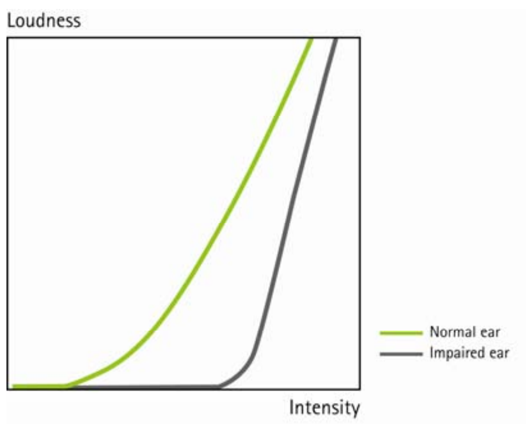


Fig. 2 Schematic of loudness perception for a normal hearing ear and for an impaired ear

PHONAK life is on

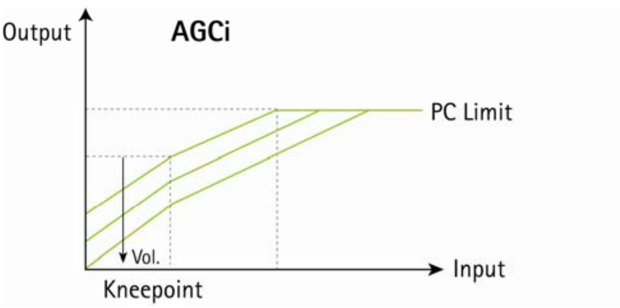
The result is that, with linear amplification, only a few signals can be properly amplified to suit the individual residual hearing range of someone with hearing loss. All other sounds are either too loud or too soft. They may not be heard at all, be masked or may even cause discomfort. A modern hearing system therefore requires a suitable amplitude compression system which brings as many of the incoming signals as possible into a person's residual dynamic hearing range.

History of compression in hearing systems

Methods for changing the dynamic range in hearing instruments have existed for more than 50 years. The first instruments of this kind were equipped with simple limiters, which were only able to limit the maximum output level. Because they simply eliminated the components of the amplified signal which exceeded the defined limit, this method was called **peak clipping**. However, simply cutting off the peaks results in a significant change to the signal which causes a high level of distortion. Users could perceive this sudden signal degradation.

Some improvement came with the first real amplitude **compression** circuits. By controlling gain based on the intensity of the input signal, it is possible to continuously adjust the amplification so that large portions of the input signal fall within the user's residual hearing range. Depending on whether the signal adjustment is made before or after the volume control, the system is called automatic gain control for input (AGCi) or automatic gain control for output (AGCo). Each of these methods has its own advantages and disadvantages, partly related to the threshold kneepoints which, in early compression systems, were generally quite high (60 dB or more). With AGCi, the user can adjust the volume control himself to set the most comfortable sound, while with AGCo, the volume control can be used to improve the signal-to-noise ratio for certain situations.

Historically, the various manufacturers of hearing instruments often used quite different audiological models and had different objectives in compensating for hearing loss. The result of this was that some manufacturers, depending on their philosophy, had only AGCi models in their products and others only AGCo. Figures 3 and 4 provide show the fundamental principles of these two methods.



.Fig. 3 Schematic shows the fundamental principles of input dependent AGCi. In an AGCi circuit typical compression ratios are 2:1 or 3:1 and the maximum output is limited by peak clipping or AGCo.

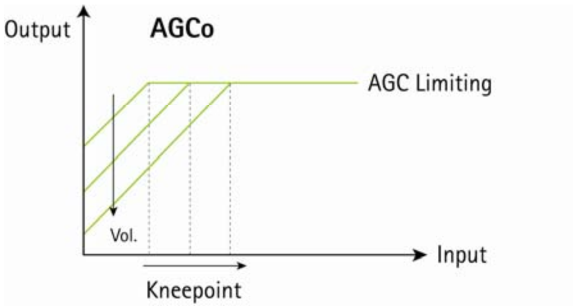


Fig. 4 Schematic shows the fundamental principles of AGCo. An AGCo circuit typically has a compression ratio of 10:1 and is largely used for output limiting.

As already mentioned, early systems used rather high threshold kneepoints, usually around 60dB. To further improve audibility of very soft sounds, systems with very low knee-points were introduced. This approach is often referred to as wide dynamic range compression (WDRC). Frequently used for cases where the residual dynamic range is severely restricted, WDRC became widespread across most manufacturers. The first instruments with WDRC separated the low and high frequencies into two separate channels. Modern hearing systems now work with many different frequency channels and each channel has its own compressor. If these multiple compressors each worked independently, sound quality would be compromised because of distortion and spectral artifacts. To avoid this undesirable distortion and ensure that the spectral contrast of the signals is not smeared by the compression, most systems now link the individual frequency channels in some way.

With time, the only constant is change

To achieve appropriate compression, it is not enough to continuously adjust the amplification. This is due to the fact that for all signals, and for human hearing in general, timing plays an important role. If, for instance, a compression system made instantaneous adjustments, this would result in changes to the signal which would cause substantial and unacceptable harmonic distortion. This is why all compression systems – even today – have specified time constants within which they react. These time constants prevent major distortion and should also ensure a smooth sound pattern with no dramatic variations in the volume.

In early compression systems, time constants were defined according to what was then known about the reaction times of the ear. The attack time was derived from the time it takes the brain to fully register variations in stimuli. Since, in those days, this time was believed to be about 20 ms, compression attack times were typically about 20 ms or a little less. The release time of about 200 ms was based on the fact that hearing thresholds for shorter duration signals are unstable and tend to increase. 200 ms is roughly the time within which the brain tends to integrate individual brief stimuli into an overall sensation or sound (Zwicker, 1974). In the early days of compression systems, this was regarded as the best solulution.

As a result of a great deal of research, knowledge about the timing of responses in the hearing-impaired ear steadily grew. This led to new models for optimizing the time constants in compression systems. With the launch of digital hearing instruments, improved compression strategies quickly became established on the market.

Various compression timing strategies

One widely-used compression strategy is to work with relatively short attack times of about 5 ms and long release times of 200 ms or even longer. The advantage of this is that the short attack time can fairly reliably reduce even suddenly occurring loud sounds, while the long release times ensure a smooth, comfortable sound pattern, with little distortion. This means that the compression knee-points can be reduced to 30 dB or less without disturbing the hearing instrument user.

Another strategy works with very short attack **and** release times, well below 50 ms. Because of these short time constants, variations in level can be controlled even within syllables, so this type of compression is also referred to as syllabic compression. It has the advantage that the overall dynamic range of the signal is greatly reduced, so that it can be reproduced in the limited residual dynamic hearing range of the user. This is generally not possible with a system that works slowly. Nevertheless, the rapid control also causes distortion, as a result of which the sound can be significantly modified, and during breaks in the desired signal there is more background noise compared to slow compression systems.

A more sophisticated approach works on the basis of frequencies, with time constants varying. Because of the risk of distortion, signals at low frequencies have longer time constants than those at high frequencies.

Some people with hearing loss prefer slow-acting compression, while others prefer fast-acting. While there is a tendency for people with more severe hearing losses to prefer slower time constants, the connection between severity of hearing impairment and timing preferences is not clear-cut. It is also not necessarily true that younger patients prefer faster compression and older patients slower, as is sometimes claimed or assumed. (Gatehouse et al, 2006) Research indicates that cognitive function is also a factor relevant for selecting time constants. (Pichora Fuller, 2006) For example, faster acting compression is useful for individuals with intact cognitive processing abilities, while those with diminished cognitive function perform better with slow-acting compression. (Lunner, 2003)

So, since each type of compression system has its advantages and disadvantages, it is not surprising that some advanced hearing instruments now employ a form of **dual-path compression.** A slow- or fast-acting compression strategy is automatically activated, depending on the situation. The time constants can vary across a wide range. Adaptive compression systems that are available in the most sophisticated instruments, use not only the level of the signal but also other characteristics of the signal as relevant parameters to determine the control of overall amplification.

(R)Evolution of Phonak processing systems

Leading up to the introduction of the first Phonak digital hearing instrument, the focus for development, particularly for compression systems, was on the cochlea. Claro was the first entirely digital Phonak hearing instrument with a perception based signal processing strategy which incorporated the psycho-acoustic and auditory processes of the inner ear in its signal processing. Digital Perception Processing (DPP) in Claro continuously evaluated the loudness of incoming signals and used this information to select the correct amount of amplification. At the same time, it also considered the dynamic properties of sounds, together with spectral masking effects and the bandwidth of the channels.

Claro™

Compression was, of course, a fundamental element of this processing system and two different adaptive strategies were available. The hearing care professional could select which strategy the client needed or wanted. One option employed short time constants (Fast Adaptive DPP) and one used long time constants (Slow Adaptive DPP). The time constants were symmetrical, meaning the attack and release times were the same. To make sure that the internal noise of the microphone could not be heard with very soft input signals, expansion (Soft Squelch) was also applied. Depending on the frequency, expansion regulates amplification below the knee-point, and so maintains the balance between sufficient amplification of quiet signals and suppression of internal noise.

Perseo

In Perseo, the enhanced signal processing strategy DPP² was developed so that a single but improved compression system with different attack and release times was used. The attack time was 10 ms and the release time 80 ms. The relatively fast attack time ensured good protection for sudden loud signals, while the slower release time made for a balanced sound pattern with no pumping effects. This avoided blurring of syllables by maintaining the proper amplitude fluctuations of the original signal. Together with more accurate DPP² fitting precalculation, these adjustments to the time constants meant that effective compression was more beneficial to the user. The compression knee-points in Perseo were also improved in that they were set lower, and were frequency- and signal-dependent. For narrow band signals they were in the range of 30–35 dB and for broadband signals 45–50 dB. The application of expansion was also enhanced in Perseo. By achieving smoother transitions between expansion and compression, timbre artifacts that could be perceived by the user as unpleasant were eliminated.

Further learning about how to better incorporate the intricate processes of the cochlea was applied in the next generation system known as Bionic Perception Processing (BPP). First introduced in Savia, BPP optimized time constants and tuned the frequency filter banks to further minimize subtle artifacts still possible with earlier systems. Additionally, a new spectral output limitation system was incorporated. This allowed for an optimum combination of control over loud inputs to maintain comfort and accurate reproduction of amplitude fluctuations needed to ensure intelligibility. This is particularly beneficial in noisy situations where both loud sounds and speech need to be effectively managed.

CORE Processing with the most sophisticated dual-path compression system available today

Modern digital signal processors in hearing systems manage and control hundreds of parameters. In many cases these affect each other, sometimes with negative results. What distinguishes an effective signal processor is the art of harmonizing hundreds of different parameters to avoid artifacts and internal interference signals. The need to react quickly and accurately to different situations means that compression systems often employ a dual-path compression strategy, with one path providing slow-acting, the other path providing fast-acting gain control. However, the fast-acting path with its short time constants can blur the amplitude fluctuations of the original signal, as discussed above, and the slow-acting path can make internal noise audible.

Some competitive systems currently available on the market (and even new systems now being introduced) use a process whereby both compression paths are directly controlled by averaging the input signal over time. Since an input average is used to determine compression settings, depending on the situation, neither may be ideal. An example illustrates some drawbacks of time-average input gain control. Consider a situation where an abrupt change in the sound environment occurs, for instance, starting a loud conversation in a previously quiet room. In this scenario, a time-input average control system would first increase the gain, causing a substantial overshoot, before finally applying the appropriate compression. Figure 5 illustrates this example and behavior.

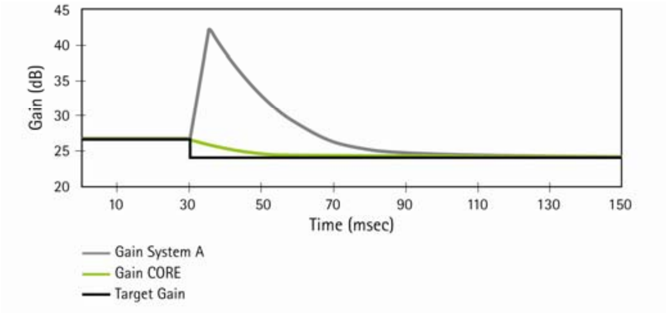


Figure 5 This example shows the adjustments to gain over time in response to an abrupt increase in signal intensity (20dB to 70dB, shown occurring at about 30msec) using a time-input average gain control system (yellow curve) which overshoots the target gain, and CORE processing, (green curve) which accurately achieves the target quickly, smoothly and without overshoot.

The **CORE** (Communcation Optimized Real-audio Engine) audioprocessor, available in all new Phonak products since Exélia, avoids these pitfalls that plague other systems by clearly separating input level detection from the calculation of spectral amplification. By calculating these parameters many hundreds of times per second in each path, the spectral cues are more accurately reproduced to keep pace with the ever-changing soundscape. So, in the same example described before, the appropriate level of compression is instantly applied resulting in no gain overshoot. Figure 5 also illustrates this preferable behavior.

Furthermore, this unique CORE adaptive dual-path compression system is integrated into the sound-class sensitive SoundFlow automatic, so amplification control is both time and situation-sensitive. As appropriate for the situation, the time constants which provide best signal transmission with the least distortion are automatically selected. This achieves virtually instantaneous, yet smooth and automatic adjustments, so even the most abrupt changes to the scoundscape are efficiently managed without distortion or artifacts.

Ultimately, CORE Processing, which incorporates the industry's most advanced dual-path compression system achieves, better than ever before, the fundamental objectives of amplification – clarity, comfort and natural sound quality.

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