Minimizing Signal Distortion through a Novel Amplification Scheme in Bone Anchored Sound Processors

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Bone anchored sound processors have been used successfully for many years now, but with technology that lags behind that of modern air conduction hearing instruments. With continual developments in signal processing technologies in air conduction hearing instruments, there is a strong need for incorporating newer technologies into bone anchored sound processors so that patients wearing these processors can benefit from their superior performance to the same extent that wearers of advanced air conduction hearing instruments currently do. An amplification scheme that is designed to minimize distortion of speech signal even during the occurrence of a sudden extraneous noise signal has been shown to benefit users of Oticon's advanced hearing instruments in terms of speech understanding and sound quality (Bruun Hansen et al., 2010). This amplification scheme is known as Speech Guard. Given the reduced dynamic range of hearing of wearers of bone anchored sound processors and the relatively small dynamic range of the processors themselves, Speech Guard avoids the common pitfall of compression that is widely used in today's hearing aids and bone anchored processors: signal distortion. This paper describes how Speech Guard is designed to maintain the amplitude variations between sounds and preserve the natural details and nuances of speech.

Introduction

The heart of any hearing device is its amplification system. Though hearing devices are often simply considered to be amplifiers, there is a lot more than simple amplification that goes on in the amplification system. A complex series of signal processing takes place to ensure that the signal that reaches the user is audible, comfortable and intelligible. A fundamental task of the amplifier is to respond to intensity fluctuations in the input signal in such a way that the integrity and fidelity of the output signal is maximally preserved. Achieving this is rather challenging for designers of hearing devices as any signal processing that is employed to maximize audibility is bound to result in some sort of distortion. While there is no arguing that the audibility is the basic fundamental function of any hearing device, audible distortion in the signal can potentially influence the way end users accept and use hearing devices in the long term. The Mark Track Survey (Kochkin, 2010) has clearly revealed that there is more to patient satisfaction than merely making sounds audible. Just how natural or undistorted the signal from the device is can significantly impact patient satisfaction. There is also a cognitive argument for preserving the natural dynamics of speech as much as possible. It has been shown that a natural undistorted signal is the easiest for the brain to process (Lunner, 2010, Behrens, 2010). It is quite clear that preserving the integrity of the input signal is a key to achieving patient satisfaction and long term use.

There are several parameters of amplification that have to be taken into account when designing an amplification system. For one thing, the amount of gain prescribed for signal inputs across a range of frequencies is a key consideration. Typically, for sensorineural hearing loses, this is achieved through wide dynamic range compression (WDRC). WDRC is employed to varying extents in almost all hearing devices to squeeze a range of inputs into the patient's dynamic range of hearing. Weaker parts of the speech such as the consonants are turned up so that they are audible, while the louder parts of speech such as the vowels are turned down so that they are within comfortable loudness levels. Some form of compression is also used to keep the amplified signal below the saturation limit of the device to avoid distortion. Essentially, with WDRC, soft and loud sounds are kept within the working auditory space (dynamic range) of the user and of the device. However, there is a downside to WDRC; it always distorts the signal it compresses (Souza, 2002). Despite this, designers of amplification systems have almost entirely focused on achieving audibility. Maximizing speech fidelity was hardly a top priority. For the users of hearing devices, this basically meant accepting distortion as an inevitable side effect of the amplification. An alternative to wide dynamic range compression is linear gain whereby equal amounts of gain are applied to a range of inputs. Linear gain processing was the amplification strategy of choice before wide dynamic range came into being in hearing devices. It has a distinct advantage of preserving the amplitude variations inherent in the input speech signal as much as possible. In other word, there is minimal signal distortion with linear amplification (Arehart K, Kates J and Anderson, 2010).

Another key parameter that is often overlooked or given much less attention is the speed at which the amplification system responds to fluctuations in the input signal. This speed is commonly referred to as time constants in clinical audiology. A fast response system can make softer sounds that follow an intense sound in speech audible but it has a tendency to distort the speech signal through frequent gain change (Souza, 2002; Stone and Moore, 2007). A slow response system helps to preserve the amplitude-time structure of the original signal in the longer term by keeping the gain constant, and is less distorting to the speech signal (Plomp, 1988). Overall, better speech fidelity can be achieved with a slow response system (Schum and Sockalingam, 2010).

Achieving the required audibility with minimal distortion is a huge challenge from a signal processing standpoint. It requires an amplification system that is able to compute the right amount of gain and the right response speed for a given input at a given point in time. For developers of Oticon Medical's Ponto sound processors however, this challenge is now a thing of the past. Ponto, Ponto Pro and Ponto Pro Power now feature a novel amplification system called Speech Guard that is designed to achieve just that audibility without the level distortion that is commonly associated with WDRC systems.

The Structure and Function of Speech Guard

Speech Guard, as its name suggests, is designed to "protect" those intensity variations naturally inherent in speech as much as possible. The key to achieving this lies in the time constants Speech Guard uses to respond to fluctuations in the input signal. Speech Guard comprises two systems constantly interacting with each other and responding to fluctuations in the input signal in real time. It is a highly adaptive system, capable of responding either slowly or quickly depending on changes in input level. There is essentially a slow response system and a

Figure 1: Components of the Speech Guard Amplification System

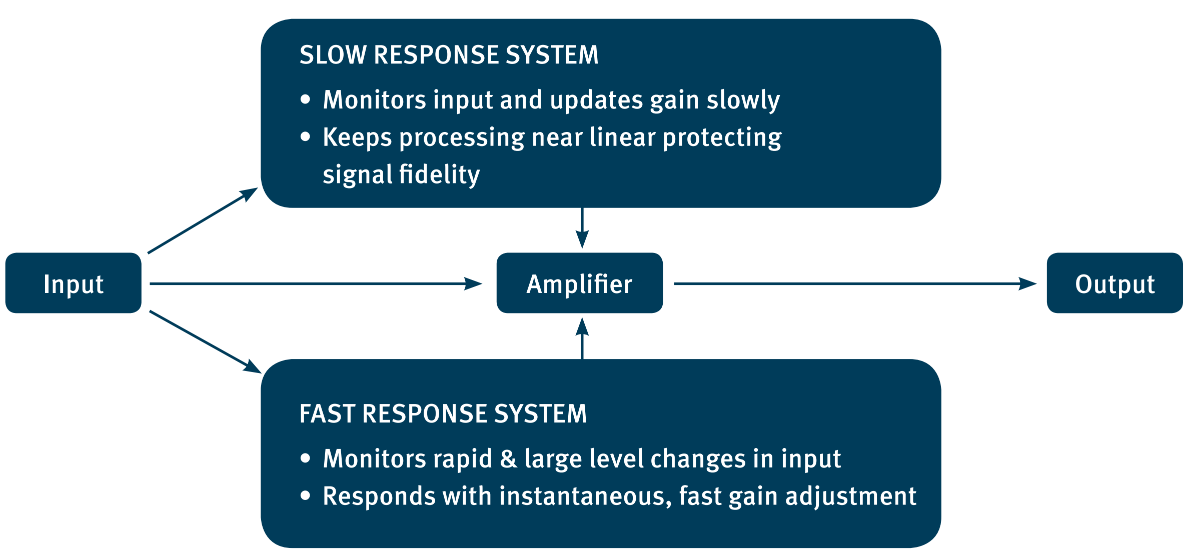

Figure 1 shows the two systems in Speech Guard working in tandem to respond at appropriate speeds to fluctuations in the input signal. The ultimate goal of these two systems is to ensure that the dynamics of the output speech signal resembles that of the input signal as much as possible.

fast response system. Together, the two systems ensure that gain is adjusted at a speed that is optimal for audibility and signal fidelity (see figure 1). When the overall signal level does not change much, the slow analyser keeps the gain more or less the same. In such situations, Speech Guard operates more or less in a linear fashion. What this means is that within a defined range of input levels the same amount of gain is applied. The advantage of using a slower response system in preserving the information embedded in amplitude variation between sounds relative to the original has been well documented (Schum and Sockalingam, 2010).

What happens if the input signal suddenly changes in intensity? If there is a sudden rise in the signal level, Speech Guard responds by changing the gain rapidly so that the output level is well within the dynamic range of the hearing device. By keeping the level within this range, it reduces distortion that may occur when the saturation point of the device is reached. When the sudden loud intrusive sound is gone, Speech Guard will quickly revert to a slow response mode so that the system can continue to process speech as linearly as possible. In doing so, the integrity of the ongoing speech signal following the loud sound is preserved.

Speech Guard Solution in Ponto Sound Processors

In Oticon Medical's Ponto family of sound processors, the slow system in Speech Guard ensures that the whole amplification system operates in a linear fashion as long as possible. Linear gain is known to be suitable for conductive hearing loss. There is no need for compression for this type of loss as the air-bone gap is simply overcome via direct stimulation of a healthy cochlea by the bone anchored transducer. Linear gain sufficiently compensates for the small loss of energy through transcranial attenuation. In unilateral profound sensorineural hearing loss, linear gain is also applied to compensate for transcranial attenuation as the signal is relayed through the skull to the cochlea on the contralateral side. For mixed conductive and sensorineural hearing loss, WDRC has been used in some

Figure 2: The Effect of Speech Guard on the Speech Envelope

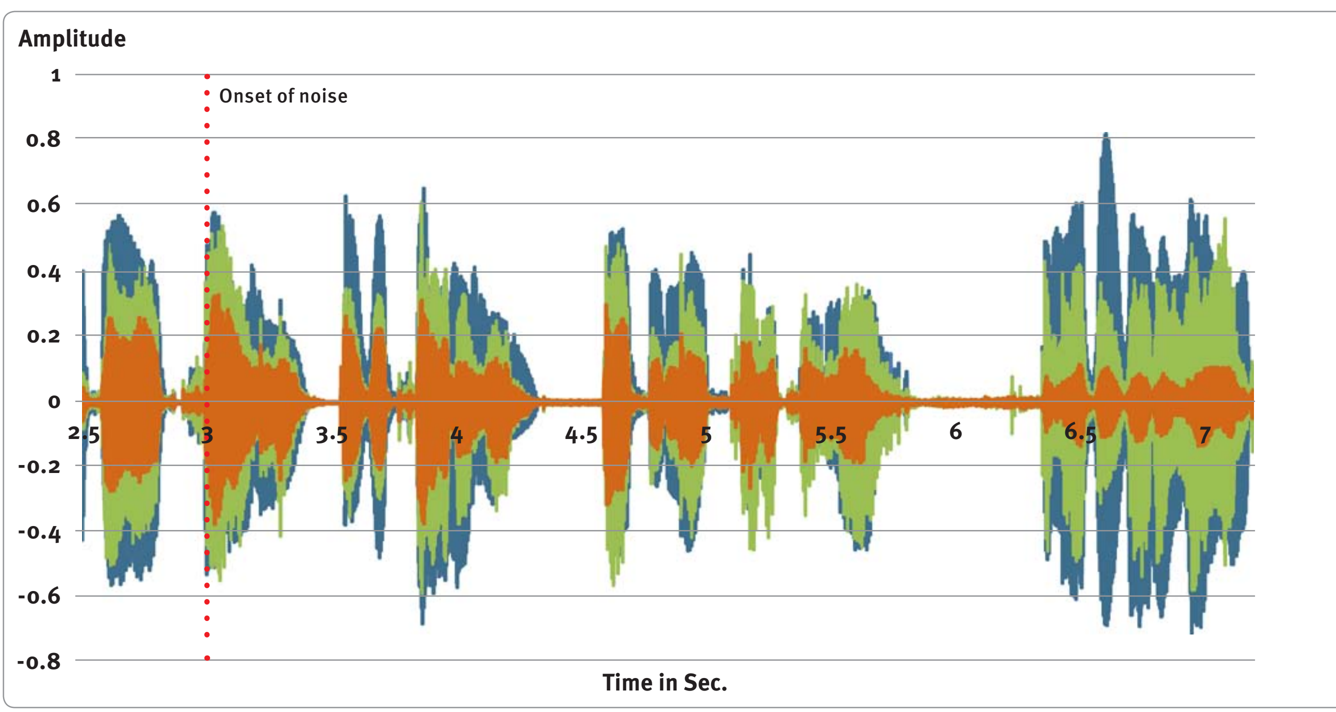

Figure 2: The envelope of the original speech signal (blue), speech processed by Speech Guard amplification system (green), and speech processed by an advanced amplification system without Speech Guard (orange). The vertical dotted line indicates the onset of high frequency noise at 3 seconds for the speech signals processed with (green) and without (orange) Speech Guard. The level of both speech and the noise were 70 db SPL. Note that the speech processed by Speech Guard matches the original speech signal in amplitude variation far better than speech processed by a system without Speech Guard. Speech Guard has the unique ability to better preserve signal integrity even in the presence of noise.

bone anchored processors to increase audibility within the limited dynamic range of the patient's sound processor. It is imperative therefore that the amplification system employed in these processors be able to handle sudden large increases in input level without causing distortion and discomfort. With the combination of fast response and slow response systems of Speech Guard, the negative side effect of compression – distortion of speech signal – can possibly be avoided.

Based on an amplification strategy that is predominantly linear, Speech Guard in the Ponto sound processors aims to preserve the natural amplitude-time envelope of speech in a unique way. It does so even if noise is introduced during ongoing speech in a unique way. In such situations, the distortion to ongoing speech that is so commonly associated with conventional compression systems is hardly noticeable with Speech Guard. This is clearly demonstrated in figure 2. The phoneme to phoneme intensity transitions as well as gaps between phonemes in speech are preserved thus keeping the output speech signal from as close as possible to the original speech signal. The performance benefits of Speech Guard in advanced Oticon hearing instruments have been clearly shown in laboratory conditions as well as in the real world (Bruun Hansen et al, 2010).

Final Comments

As a novel amplification scheme, Speech Guard has been well described (Simonsen and Behrens; Sockalingam and Simonsen 2010), and its audiological advantages and benefits, particularly in difficult listening environments, have been reported in studies on Oticon hearing instruments (Bruun Hansen et al, 2010; Schum and Sockalingam, 2010). The key to Speech Guard lies in its ability to respond to signal fluctuations using a system that circumvents the pitfalls of either a slow response or a fast response. By being in a slow response mode as long as possible in relatively stable listening environments, and in a fast response mode for sudden increases in signal level, Speech Guard in Oticon Medical Ponto hearing systems preserves the integrity and fidelity of the speech signal even when an intrusive extraneous sound interrupts speech.

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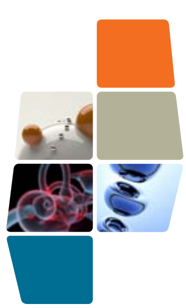

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