Sonic Spotlight

Speech Variable Processing Revisited

Phoneme Focus vs. Envelope Focus

Just like snowflakes, no two hearing losses are ever quite the same. Even when audiograms exhibit identical thresholds, patients with equivalent hearing impairment will understand speech in their own unique way. Depending on patients' auditory resolution abilities, speech understanding – especially in noise – may differ mildly to wildly. Read on to find out how Sonic helps address differing auditory resolution abilities, with the digital signal processing options found in the Speech Variable Processing platform.

Through observations from clinical experience, hearing care professionals know that two patients with nearly identical audiograms can vary in the benefit they receive from amplification. While both may report great benefit listening in quiet, they may differ in their ability to understand in noise. What underlies such differences when thresholds appear to be the same? To answer that question, we must look deeper into the physiological processes of audition, and also understand how amplification strategies relate to the underlying physiology of individual patients.

Peripheral sound processing

The perception of hearing begins when sound vibrations travel upstream through the outer and middle ear. Behind the tympanic membrane, the footplate of the stapes pushes on the oval window, which creates a pressure difference on the cochlear fluid contained within the bony structure of the inner ear. This change in pressure causes the basilar membrane within the snail-shaped cochlea to move. The organ of Corti, which lies on the basilar membrane, holds sensory receptors – bundles of stereocilia made up of outer and inner hair cells (Figure 1). As sound initiates a traveling wave moving down the basilar membrane, the shearing action between it and the overlying tectorial membrane cause the hair cells on the organ of Corti to bend in response (Zemlin, 1988).

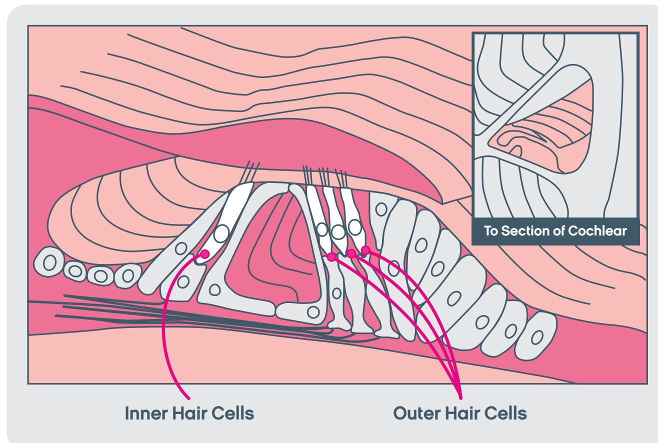


Figure 1: Organ of Corti

The basilar membrane does more than just move randomly. Similar to a key on a piano that resonates precisely for one note, the basilar membrane will naturally vibrate in a specific manner according to the particular frequency or frequencies it receives. In that respect, it acts as a sharply tuned band-pass filter. It gives a strong response to a limited range of frequencies for a specific location along the membrane. This gives rise to the frequency selectivity for which this sensory organ is known. The membrane's characteristic frequency (CF) is the place of maximum displacement. Here the amplitude of the vibration will exhibit a very sharp response peak in a healthy system, as shown in Figure 2 (Darwin, 1994). Additionally, the characteristics of the peak will correspond to the amplitude of the signal. Due to the tonotopic arrangement of hair cells, low-frequency sounds maximally stimulate the basilar membrane at the apical end of the cochlea, whereas high-frequency sounds stimulate it at the basal end. Ultimately, disturbances of the hair cells transform mechanical energy into electrical impulses that stimulate the auditory nerve.

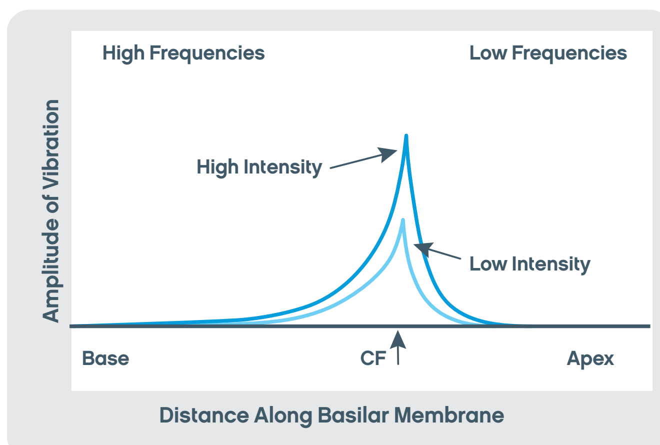


Figure 2: Representation of sound vibration on the basilar membrane; used with permission from Darwin, C. 1994.

Hair cell damage

Like most things in nature, the healthier the system, the better it functions. Intact outer hair cells on the organ of Corti act as a cochlear amplifier and transmit sound in a non-linear fashion to the auditory nerve: a stronger response occurs to softer input; and a weaker response occurs to louder input (Canlon, 2010). However in an unhealthy system (i.e., sensorineural hearing loss), the degree of hair cell loss affects the signals it receives. Damaged hair cells – due to noise, age, ototoxic medications, etc. – will cause broader frequency tuning in response to the frequency of the input (Ginsberg and White, 1994; Glasberg and Moore, 1986). Or, they may fail to receive any input and unsuccessfully stimulate the auditory nerve, resulting in a complete loss of sensitivity for the corresponding frequencies (cochlear dead regions) (Moore, 2004). Finally, the non-linear nature of the cochlear amplifier changes, and the auditory system becomes more linear in response to sound, causing recruitment in some cases.

Hearing aids and individual performance variability

No one will argue that hearing aids are the best remedy for sensorineural hearing loss. Most current amplification strategies employ wide dynamic range compression (WDRC) to increase the audibility of soft sounds, yet keep loud sounds comfortable. However, as stated earlier, hearing aid benefit still varies for hearing-impaired patients, especially in noise. A major contributing reason is because individual variation exists regarding the ability to extract speech signals embedded in background noise – an ability otherwise known as 'listening in the dips' (Lorenzi et al., 2006). Normal hearing listeners and those with mild to moderate losses are better able to use auditory information during background noise fluctuations, or 'dips', compared to listeners with more progressive hearing loss (Hopkins and Moore, 2007; Strelcyk and Dau, 2009; Ardoint, 2010).

In addition to hearing loss, both age and cognition have been shown to affect one's ability to listen in the dips (Duqesnoy, 1983; Gatehouse et al., 2003). Finally, digital signal processing (DSP) has its own effect on listening performance, in quiet and in noise (Moore, 2008). As such, DSP becomes an important variable like directional microphones or noise reduction that influences the listening benefit for individuals with varying degrees of auditory resolution.

The following sections explain the important factors related to speech signal components, auditory coding of speech signals, and DSP optimization with Speech Variable Processing to better address the individual amplification needs of patients.

Acoustic signal components – spectral and temporal properties

In order to optimize amplification for patients, it is important to first understand the components of an acoustic signal that we can process or manipulate with DSP. Speech sounds are a perfect model to illustrate the two main structural components – the spectral and temporal properties – of a signal.

Spectral properties: Spectral, or frequency-based features identify the pitch or timbre of sound. Most of our basic knowledge about speech frequencies centers on the audiogram from .25 – 8 kHz, the frequency content of phonemes, formants in vowels, or the relative location where consonants and vowels fall on the audiogram. Conveniently, we explain to patients how the cochlea follows the same organization, acting as a frequency analyzer that filters speech from low to high frequencies along the basilar membrane: a loss of sensitivity at a particular frequency directly correlates to the need for amplification at the same frequency location to make the sound audible. Shown in Figure 3, a spectrogram represents the spectral properties of sound in the frequency domain, where the horizontal axis represents time, the vertical axis represents frequency, and color intensity represents the amplitude of a particular frequency at a particular time.

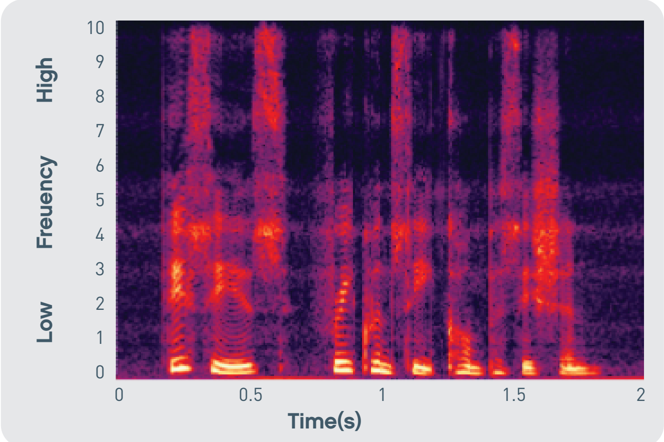


Figure 3: Spectrogram example

Temporal properties: In addition to the spectral properties, the temporal structure of sound deserves equal attention, since it influences speech perception as well (Rosen, 1992). In fact, understanding the time-domain of speech is an emerging factor for selecting individualized amplification strategies to benefit the needs of patients (Gatehouse et al., 2006a). A waveform graphically represents the temporal properties of sound in the time domain. Figure 4a shows the format: a number of amplitude-modulated signals representing the output of a set of narrow frequency channels is distributed across the acoustic spectrum over time (Swaminathan, 2012). Furthermore, we can extract additional data from the waveform. The output of the channels can be mathematically factored into two products: a slowly varying modulation signal, the temporal envelope (ENV); and a rapidly varying carrier signal, the temporal fine structure (TFS), shown in Figures 4b and 4c, respectively. The TFS refers to rapidly fluctuating variations in amplitude of the waveform at a rate close to the center frequency of the band. The ENV refers to slower amplitude modulations superimposed on the TFS (Flanagan, 1980).

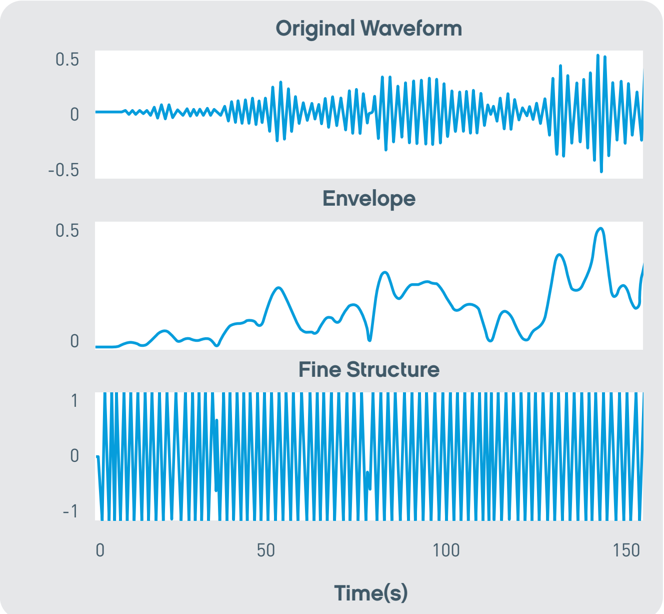


Figure 4a: The original waveform of a speech signal;
Figure 4b: The envelope (ENV) of the waveform;
Figure 4c: The temporal fine structure (TFS) of the waveform.
The contour of the ENV closely follows the outline of the original waveform, whereas the TFS contains all the fluctuations found in the original signal. Used with permission from Delgutte, B.: https://research.meei.harvard.edu/chimera/motivation.html

We will now focus on the temporal properties of speech signals, in order to reveal their relationship with Speech Variable Processing.

Envelope and temporal fine structure

As stated above, the temporal components of a signal contain both envelope (ENV) and temporal fine structure (TFS) information. Each component provides unique acoustic cues to the auditory system (Figure 5).

ENV: The temporal envelope conveys information about the spectral shape of a signal, and it also indicates how the signal's short-term spectrum changes over time. The slowly varying temporal variations correspond to salient acoustic features we perceive in speech, like intensity and duration. It also conveys linguistic information such as manner of articulation, voicing, place cues, vowel identity, and prosody (Rosen, 1992; Swaminathan, 2012). Listeners with or without hearing impairment easily perceive these robust cues, and rely on them to hear in quiet listening situations (Shannon et al., 1995).

TFS: The fast fluctuations of a signal's TFS are carrier waves containing useful information for sound identification. These rapidly varying oscillations carry details about the signal's fundamental frequency, and about its short-term spectrum. TFS comprises the small variations that occur between pauses of a periodic signal (e.g., speech) or within short time intervals of aperiodic sound (e.g., noise) (Rosen, 1992). Not every listener can benefit from TFS, specifically in complex listening situations. Whereas listeners with normal hearing and milder losses retain the ability to use the small amount of auditory information in the short time intervals of background noise ('listening in the dips'), listeners with increasing hearing impairment do not. They have reduced sensitivity to TFS (Hopkins et al., 2008).

While studies have shown that ENV cues are sufficient for understanding speech in quiet, TFS is essential for understanding speech in fluctuating background noise (Festen and Plomp, 1990; Lorenzi et al., 2006). Listeners with progressive hearing loss and other associated factors like age and cognitive issues, lose their ability to use TFS and fail to benefit from the information it provides during fluctuating background noise (Hopkins et al., 2008). The ability to understand speech in complex noise, therefore, strongly relates with one's ability to utilize TFS (Lorenzi et al., 2006; Ardoint, 2010).

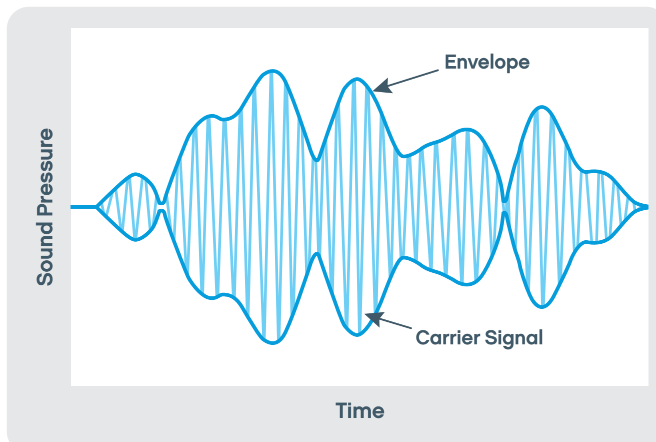


Figure 5: Envelope and carrier signal of a speech waveform

Auditory coding

Studying how sound is coded within our auditory system helps us to understand why sensitivity to envelope cues remains as the ability to use TFS declines. In his book "Auditory Processing of Temporal Fine Structure: Effects of Age and Hearing Loss", author Brian Moore describes how the ENV and TFS of an input signal relates to auditory coding within the ear and brain. Moore explains that ENV and TFS are both represented on the basilar membrane and in the auditory nerve. For example, he states that the basilar membrane breaks down broadband signals (e.g., speech) into narrowband signals. The resulting waveform at each location on the basilar membrane corresponds to the ENV, which is superimposed on top of a more rapidly fluctuating carrier, the TFS (Figure 6). The auditory nerve translates ENV and TFS information coming from the basilar membrane into the neural ENV and TFS representation of sound.

Thus, the coding process begins. An input signal causes the basilar membrane to move. Inner hair cells transduce the mechanical movements of the outer hair cells into action potentials that stimulate afferent neurons of the auditory nerve. ENV and TFS information is transmitted via timing and the rate of nerve spikes in the auditory nerve. Neural responses to the amplitude-modulated cues lock into the signal's ENV and TFS, a process known as envelope locking and phase locking, respectively (He, 2008).

Studies show that ENV information via envelope locking is well preserved – even enhanced – in neural responses at all levels of the auditory system (Joris, 2004). Phase locking, which occurs when auditory nerve fibers fire in sync with a particular phase of the signal, is not as well preserved. It requires the neural response to faithfully represent the instantaneous phase contained within the cochlear information it receives (Shamma & Lorenzi, 2013). The ability to understand speech, therefore, largely depends on efficient auditory coding of all ENV and TFS information, from the signal to the basilar membrane, and from the basilar membrane to the auditory nerve.

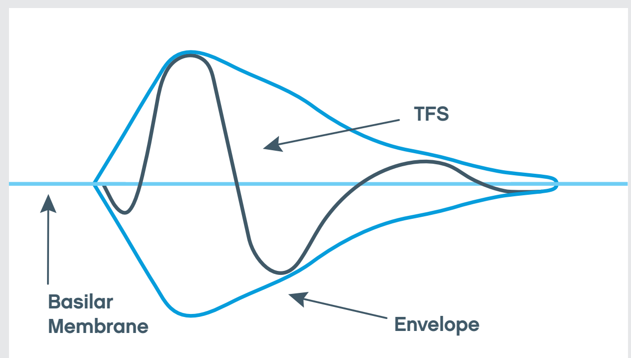


Figure 6: ENV and TFS representation on the basilar membrane; adapted with permission from Venema, T. Compression for Clinicians, 2nd ed., Cengage (2006).

Factors that influence auditory processing

The proficiency of the auditory nerve to process slow ENV and fast TFS information begins to form the framework of an individual's auditory resolution abilities. A healthy auditory system reliably conveys both slow ENV and fast TFS information contained in the signal. However, compared to the more robust ENV cues, TFS becomes vulnerable to degraded processing when the auditory system is compromised. Many physiological factors can contribute to a decreased ability to use TFS. For example:

Age-related changes: Older listeners often report increased difficulty hearing in noise than younger listeners. Temporal auditory processing abilities have been found to decline with age, even in the absence of hearing loss (Fullgrabe, 2013). Studies have shown that older listeners with normal hearing have less ability to use TFS cues, compared to younger listeners without hearing loss, which may be the result of perceptual cochlear or retro-cochlear deficits not captured in an audiometric assessment (He et al., 2008; Moore et al., 2012; King et al., 2014). Therefore, aging alone can have an impact on TFS, due to physiological changes in the inner ear.

Hearing loss: As stated earlier, progressive cochlear hearing loss can impair TFS, but it can do it in a few different ways. Research by Shamma in 1985 found that a loss of outer hair cells can cause phase-response changes to the signal which then affect the ability to extract TFS information. Similarly, Woolf et al. (1981) suggested that cochlear damage may lead to a deficit in phase locking at the level of the auditory nerve, which would reduce the ability to use TFS. Glasberg and Moore (1986) concluded that broader auditory filters may make the output of stimulus waveforms uninterpretable by the auditory system, also contributing to impaired TFS ability.

Age and hearing loss: As one might expect, the common occurrence of presbycusis, or age-related hearing loss could complicate the interpretation of the factors that affect auditory processing. However, Gallun and colleagues (2014) examined the role of TFS in relation to both age and hearing loss. Their research supports the fact that age and hearing loss are independent factors responsible for temporal processing ability. They found that when the amount of hearing loss is similar between groups, that TFS-related tasks are impaired for older listeners, compared to younger listeners.

Cognition: An interaction between hearing loss and cognition has also been established. Specifically, compared to hearing-impaired listeners with intact cognitive ability, hearing-impaired listeners with lower cognitive ability are generally less able to benefit from TFS. Gatehouse et al. (2003) reported that listeners with greater cognitive ability benefit more from TFS in background noise. This suggests that more central neural processes may be involved in patients presenting with poorer cognitive function.

The relevance of these factors – alone or combined – along with other influences like memory, attention and motivation, is to emphasize that auditory processing relies on information flowing upstream and downstream, from the afferent and efferent neural pathways between the ear and brain (Tremblay and Miller, 2014). When the integrity of the peripheral or higher auditory system breaks down, temporal fine structure information becomes susceptible to the adverse effects of a damaged system.

Effect of DSP on auditory resolution abilities

Now that we have seen how healthy versus impaired auditory systems receive acoustic signals, we can begin to understand how a hearing instrument's digital signal processing can have an influence on listener benefit. Tremblay (2015) reminds us that with nonlinear, amplitude-based compression systems used in most hearing aids today, the output signal exiting the hearing instrument into the ear canal will not be the same as the signal entering it. For example, amplification strategies such as WDRC apply more gain to soft sounds and less gain to loud sounds. In this way, it can keep loud sounds comfortable, and also reduce the need for manual volume changes. Amplitude compression uses three important parameters to alter the incoming signal. They are the compression kneepoint, compression ratio and compression speed. The remainder of this paper will focus on the effects of compression speed, showing how time constants alter the incoming signal and how it affects ENV and TFS information.

Fast vs. slow compression

Compression speed generally falls into two groups, fast-acting or slow-acting, depending on how long it takes the input level of the stimulus to reach its prescribed gain (Pittman et al., 2014). Fast time constants allow a system to rapidly apply gain to low-level phonemes (consonants) and then immediately reduce gain for high-level phonemes (vowels) that follow. Conversely, slow time constants adjust amplification based on the long-term changes of the auditory environment. Concerning speech, gain does not change from one phoneme to the next. Rather, the natural peak-to-peak differences between louder vowels and softer consonants are preserved over time and retain a greater contrast to each other, compared to a faster system.

Various studies report a number of pros and cons for each strategy. Specific to ENV and TFS however, fast compression can improve the ability to detect a weak consonant following an intense vowel, but it can also create over- and undershoot effects in the temporal envelope of sound due to fast attack and release times (Stone and Moore, 2008). Slow compression preserves envelope fluctuations, but it does not significantly increase gain for target speech found in the dips of fluctuating background noise (Bacon et al., 1998). Determining an optimal amplification strategy, therefore, requires us to know one more piece of the puzzle. That is, to understand the effect that compression speed has on differing auditory resolution abilities.

DSP related to TFS ability

The main factor that determines an optimal amplification strategy for listeners is their ability – or inability – to process TFS information. Moore (2008) explains that fast time constants may be necessary to preserve TFS to improve speech perception in complex noise – but, listeners must have the ability to use TFS for it to be effective. In these cases, he reports that fast-acting compression can help restore the audibility of low-level portions of signals, and information derived from TFS can be used to stream components of the target speech during gaps in background noise. As such, fast-acting compression may improve speech intelligibility in noise, but only for listeners who can process TFS (Moore et al., 1999).

Again, Moore (2008) goes on to explain that listeners with decreased ability to process TFS information will mainly rely on envelope cues to understand speech. As previously stated, fast-acting compression can disrupt envelope cues which would lead to reduced speech intelligibility in these cases (Stone and Moore; 2004, 2008). Therefore, listeners with reduced ability to use TFS may find slow-acting compression to be more effective.

Predicting TFS ability in patients

The studies described above show that fast time constants preserve TFS information and slow time constants preserve ENV information. We can see that one isn't 'right' or 'wrong' – but that one may be more optimal than another depending on the individual needs of a patient.

At this point, it becomes important to look for characteristic signs that may predict an individual's ability to use TFS. As specified earlier, listeners who are generally younger, presenting with mild to moderately-severe losses, or who have intact cognitive function[1] are predicted to have the ability to use TFS and rely on it to hear speech in background noise. Listeners with these characteristics may find fast compression to be optimal.

However, listeners who are elderly, with greater hearing impairment, or who have cognitive issues are expected to have less ability to use TFS and instead benefit from the contrasts in the amplitude envelope. They may find slow compression to be optimal for speech recognition in noise.

Gatehouse et al. (2006b) makes it clear that one set of time constants won't work for everyone. Therefore, hearing care professionals must look beyond the audiogram and learn to differentiate which amplification strategy works best for individual needs and abilities – then select the optimal speech setting that will provide the most benefit.

Speech Variable Processing options

In order to provide the optimum form of amplification for individual needs, Speech Variable Processing (SVP) now offers the choice of two amplification strategies[2] that places emphasis on either envelope or temporal fine structure cues. Phoneme Focus aims to preserve TFS, by providing maximum audibility of all the fine details of the speech signal. Using fast-acting compression, this strategy measures the incoming wideband signal and rapidly adjusts the gain to apply the precise amount of amplification to each phoneme. Envelope Focus is designed to support hearing-impaired people who rely on the information from the envelope for speech understanding. Using slow-acting compression, this strategy measures the wideband signal and adjusts the amplification based on the long-term changes of the auditory environment in order to preserve the envelope information. The following describes them in more detail:

Phoneme Focus – The main approach of Phoneme Focus is to apply the desired gain as fast as possible. Phoneme Focus behaves the same as the traditional Speech Variable Processing compression system has previously behaved: a broadband level estimator is used to achieve an accurate input level estimation so that the correct amount of gain will be applied to soft and loud phonemes. This method aims to preserve TFS information contained in speech in order to provide the necessary cues for understanding speech in noise.

Envelope Focus – Envelope Focus offers a slower compression speed that may give better results for some listeners. By slowing down the compression speed, the amount of gain will vary less and will have a more linear behavior. The contrast between loud and soft phonemes will be greater than with Phoneme Focus. Envelope Focus will be set as a default when HTL > 70 dB HL at 2000 Hz and higher frequencies, when the age is above 75 years, and/or when the air bone gap is > 20 dB at 3 consecutive frequencies.

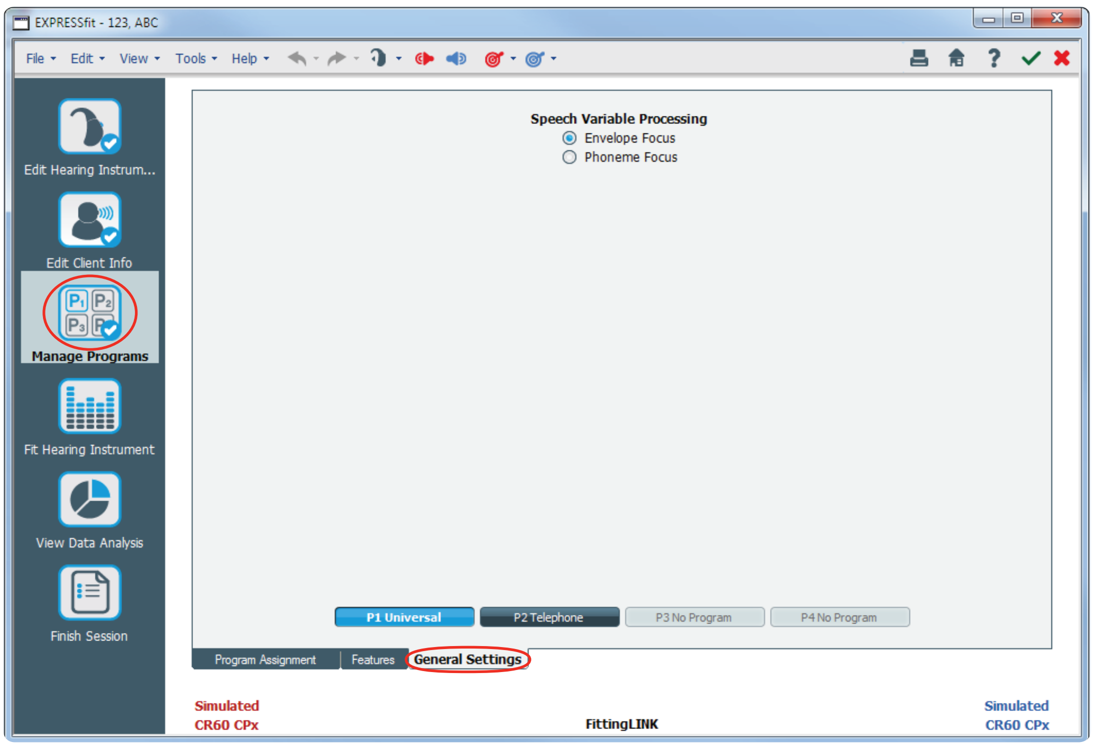
[1] Readers are referred to the Mini-Mental State Examination, the Mini-Cog Test, or Montreal Cognitive Assessment to explore further information on cognitive screening tests for audiological practice (Weinstein, 2015).
[2] Implementation details are proprietary and remain the property of Sonic Innovations, Inc.

SVP selection in EXPRESSfit

The EXPRESSfit fitting software (2015.2 and later) automatically determines the optimal SVP amplification strategy – Phoneme Focus or Envelope Focus. It selects the default based on age and audiogram information entered in the patient's record.

Hearing care professionals also have the option to manually select between the two speech settings, based on professional assessment of individual needs and abilities of patients. Phoneme Focus and Envelope Focus are conveniently located in the Manage Programs screen, under the new tab labeled General Settings.

Again, Phoneme Focus promotes maximum audibility of all the fine details of the speech signal. This strategy rapidly adjusts the gain to apply the precise amount of amplification to each phoneme. Envelope Focus supports listeners who rely on the information from the envelope for speech understanding. This strategy is designed to preserve envelope information. Remember that EXPRESSfit applies the default strategy based on information entered for the patient. However, it can be changed at any point in the fitting process. Choose either Phoneme Focus or Envelope Focus to best meet the individual needs of your patient.



Benefit

As we have learned, listeners wearing hearing instruments are different in many ways, even if hearing loss thresholds appear to be the same. The ability to hear speech in background noise varies from patient to patient. In addition to degree of hearing loss, other intrinsic factors such age, cognition, even motivation, attention or memory will affect auditory resolution abilities from listener to listener. Some patients depend on the temporal fine structure, or TFS, of sound to understand speech in background noise. Others have lost the ability to use TFS and rely on envelope information instead.

In addition, hearing aids introduce a new variable to the auditory system. Digital signal processing has its own effect on speech, depending on whether fast or slow time constants are used. Studies have shown that listeners who benefit from TFS information in speech prefer fast time constants, and those with decreased ability to use TFS prefer slow time constants, especially in noisy conditions, since it preserves the envelope cues found in speech. Therefore, selecting the signal processing for the underlying physiology of the patient becomes an important factor for providing individual benefit on a case-by-case basis.

Speech Variable Processing now offers the chance to guide the hearing aid selection process, with the option of two compression speeds:

- *Phoneme Focus with fast-acting compression aims to apply the precise amount of gain to soft and loud phonemes to enhance TFS information*

- *Envelope Focus with slow-acting compression aims to preserve the natural amplitude variations of the input speech signal to enhance envelope information*

When patients can enjoy their hearing instruments in even more situations, they experience a world where every day sounds better. Phoneme Focus and Envelope Focus is available starting with the Cheer product line, and future hearing instruments from Sonic.

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

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Sonic Innovations, Inc.
2501 Cottontail Lane
Somerset, NJ 08873 USA
+1 888 423 7834

Sonic AG
Morgenstrasse 131B
3018 Bern, Switzerland
+41 31 560 21 21