Canadian
Hearing Report
Revue canadienne d'audition
VOL. 10 NO. 6 2016

Raymond Carhart
The Father of Audiology
1912-1975

A TECHNICAL COMPARISON OF DIGITAL FREQUENCY-LOWERING ALGORITHMS AVAILABLE IN TWO CURRENT HEARING AIDS

YURI SOKOLOV: IN CONVERSATION...

SPEECH GUARD: PROVIDING AUDIBILITY AND CUES FOR AUDITORY PROCESSING

Publications Agreement Number 40025049 | ISSN1718 1860

Published by ANDREW JOHN PUBLISHING INC.
andrewjohnpublishing.com

Are we on for Christmas? In keeping with the season, this last issue for 2015 is a stocking stuffed with a variety of things inside. Here, we feature hearing aid technology and people with ideas. Our field is filled with both of these. Let's start with the people. Meet the memory of the "Father of Audiology," Raymond Carhart. I've been waiting for this one for a long time. We've all heard of Carhart's Notch, but what about the real person who wrote the classic on the notch back in 1950? It was called, "Clinical application of bone conduction audiometry," and it appeared in *Archives of Otolaryngology*, 51, pages 798–808. The American Academy of Audiology has several publications, one of them being their bimonthly *Audiology Today* (AT) magazine. With permission from AT we are reprinting an article called "Remembering Raymond Carhart." It was written by another giant in the field, James Jerger, who among many, many things, coined the names of the Jerger type tympangrams. Jerger was a student of Carhart's at Northwestern University located in Evanston Illinois, a suburban city just north of Chicago. Later on, Jerger himself taught at Northwestern as a colleague of Raymond Carhart. I think his article will make an interesting read, and it will certainly help us to better know the man behind the name.

The next person highlighted in this issue is an innovator, who has given us a lot when it comes to measuring Auditory Brainstem Responses (ABRs) and Oto-acoustic Emissions (OAEs). Meet Yuri Sokolov, the originator of Vivosonic Inc. If you've ever wanted to be able to record an ABR while doing backflips, try a system from Vivosonic. They have become well known in the field of audiology as a system that maximizes noise reduction in ABRs. While the new auditory steady state response (ASSR) became popular a decade ago, Vivosonic didn't stand still; I remember well doing contract work with them as they began developing their own ASSR approach. Yuri and I have known each other for a long time now, and one thing is consistent – he continues to innovate. Have a read of our interview with Dr Sokolov, the one who along with Vivosonic has pioneered, promoted, and presented some of Canada's most unique auditory test equipment for measuring non-behavioural responses.

You may recall from issue 4, an article I wrote on Adaptive Dynamic Range Optimization (ADRO): An Alternative Strategy to WDRC. In my editorial of that issue, I mentioned that Oticon presently uses a "floating linear compression" which has some similarities to ADRO. I also said that I hoped Oticon would possibly submit an article to CHR about this unique feature in the near future. That way, readers could do some comparisons themselves. Well, it being Christmas, I got my wish. I must have been a good boy (not)! Anyway, Thomas Behrens and Kamilla Angelo have weighed in with an article on "SpeechGuardTM. See if you can compare for yourselves the similarities and differences between their application of linear gain and that utilized by ADRO. It's nice to know that ongoing hearing aid technology is not confined to dongles, handheld devices and Bluetooth. There is lots to be said for the usage of linear gain to reduce distortion in aided speech, because this can result in rendering less client reliance upon alternate programs and other complexities.

Our final article by Hugh McDermott continues with a hearing aid technology that has become quite well known over the past few years; namely frequency lowering. Fewer hair cells cannot handle the complexities of speech of an acoustic signal nearly as well as can thousands of healthy hair cells. Witness the concept of cochlear dead regions and also, the fact that the NAL fitting method has for years made a distinction between "audibility" and "effective audibility." As we know, the technology of frequency lowering is used to shift high-frequency sound stimulation from cochlear dead regions to areas along the Basilar membrane with healthier hair cells. I first encountered frequency lowering while working at Unitron back in the mid 1990s; a company out of Israel (AVR Sonovations) introduced their hearing aid called the TransonicTM. I recall clients with "left-corner" audiograms could stand 10 feet away and turn their backs to me, and yet they were able to clearly distinguish whether /s/, /sh/ sounded "the same" or "different." True stuff, but I also remember that it made speech inputs sound a bit like Darth Vader! A few years ago, both Phonak and Widex made some huge advances in frequency lowering. Phonak does it with frequency compression, while Widex does it with frequency transposition. Have a read of an article that compares the two approaches (and be sure to be good for Christmas)!

Ted Venema, PhD,
Editor-in-Chief

CANADIAN INTERDISCIPLINARY HEARING SCIENCES, OTOLOGY AND AUDIOLOGY CONSORTIUM (CIHSOAC)

A new Canada-wide initiative in for research collaborations in audiology/ otology was set in motion this month (Nov 2015). This is the Canadian Interdisciplinary Hearing Sciences, Otology and Audiology Consortium (CIHSOAC). The Hearing Foundation of Canada aided with some industry sponsorship held the first workshop in Halifax NS, hosted by Drs. Manohar Bance and Steve Aikin. Leaders in audiology, otology, and hearing science from across the country discussed priority research questions, and plans to address them. The focus was on clinical problems rather than basic science issues. Four key areas were chosen, namely middle ear function, auditory neuropathy (SD), tinnitus, and presbyacusis.

In addition, and the reason for this note to hearing healthcare clinicians, is that we also discussed the limitations of clinical diagnostic assessments for patients with hearing difficulties. In a nutshell, the standard audiogram, and its interpretation is inadequate to address many forms of hearing problems that we now recognize.

Here are three examples:

1. We have long known that threshold measures of hearing (i.e., the audiogram) very often do not correlate with real hearing difficulty. This notion was put into sharp focus when we formally recognized auditory neuropathy spectrum disorder ANSD). More recently we have become aware of "hidden hearing loss," from studies showing that noise exposure can deplete more auditory neurons than cochlear haircells. Cochlear thresholds can look normal despite the retro-cochlear lesions. Standard audiometry has no chance of revealing any subtle threshold deficits. However if we used a higher definition testing with 1–2 dB accuracy instead of 10dB approximations, we might identify ANSD like problems.

2. Why do we still, in 2015, not measure the whole high-frequency range to our hearing? Why do we only measure up to 8 kHz when we know that most hearing problems typically start at the high frequencies? Ototoxic drug damage, presbyacusis, noise trauma can all cause hearing loss at frequencies above those that we routinely test. Some studies in patients with tinnitus that appear to have "normal audiograms" turn out to have very high frequency hearing loss (above 8 kHz), but not usually detected. This is crazy; we need to make high frequency audiometric testing standard.

3. Given the lack of level definition and frequency range of standard audiometry the interpretation of the audiogram for reporting purposes is very misleading. To suggest that a patient with a threshold elevation of 10–15 dB is in the "normal range" is not useful. Statistically, a 5dB threshold loss is significant! To report that a subject had normal hearing thresholds when at 16 kHz there may be an unmeasured 50 dB threshold elevation is clearly inaccurate. Our recognition of ANSD and "hidden hearing loss" etc. should, by now, have prompted us to revise our standard audiometric procedures. We need to see some evolution, and that will only come when audiologists and otologists agree that change is necessary.

Our consortium members would be very interested to hear your immediate feedback comments on this audiogram issue. Please feel free to contact the group (through myself at, rvh@sickkids.ca).

On behalf of CIHSOAC,
Robert V. Harrison

Professor, and Vice-Chair (research)
Department of Otolaryngology –
Head and Neck Surgery
University of Toronto.

Director, Auditory Science Laboratory,
Program in Neuroscience and
Mental Health
The Hospital for Sick Children,
Toronto, Canada.

Canadian
Hearing Report
Revue canadienne d'audition

Vol. 10 No 6 • 2016

EDITOR-IN-CHIEF / EDETEUR EN CHEF
Ted Venema, PhD, tvenema@shaw.ca

MANAGING EDITOR / DIRECTEUR DE LA RÉDACTION
Scott Bryant, scottbryant@andrewjohnpublishing.com

CONTRIBUTORS
Kamilla Angelo, Thomas Behrens, James Jerger, Hugh J McDermott, Ted Venema,

ART DIRECTOR/DESIGN / DIRECTEUR ARTISTIQUE/DESIGN
Amanda Zylstra, design@studio19.ca

SALES AND CIRCULATION COORDINATOR. / COORDONATRICE DES VENTES ET DE LA DIFFUSION
Brenda Robinson, brobinson@andrewjohnpublishing.com

ACCOUNTING / COMPTABILITÉ
Susan McClung

GROUP PUBLISHER / CHEF DE LA DIRECTION
John D. Birkby, jbirkby@andrewjohnpublishing.com

Canadian Hearing Report is published six times annually by Andrew John Publishing Inc. with offices at 115 King Street West, Dundas, On, Canada L9H 1V1.

We welcome editorial submissions but cannot assume responsibility or commitment for unsolicited material. Any editorial material, including photographs that are accepted from an unsolicited contributor, will become the property of Andrew John Publishing Inc.

FEEDBACK

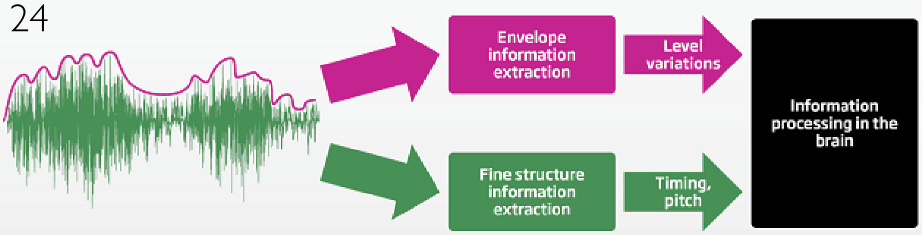
We welcome your views and comments. Please send them to Andrew John Publishing Inc., 115 King Street West, Dundas, ON, Canada L9H 1V1.

INDIVIDUAL COPIES

Individual copies may be purchased for a price of $19.95 Canadian. Bulk orders may be purchased at a discounted price with a minimum order of 25 copies. Please contact Ms. Brenda Robinson at (905) 628-4309 or brobinson@andrewjohnpublishing.com for more information and specific pricing.

Vol. 10 No 6 | 2016

contents



BACK ISSUES CAN BE FOUND AT WWW.CHR-SEARCH.COM

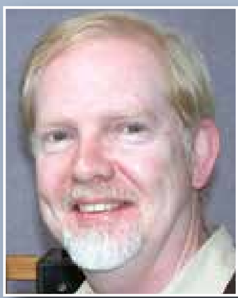
Follow us on Twitter @chr_info

Publications Agreement Number 40025049 • ISSN 1718 1860

Return undeliverable Canadian Addresses to:
Andrew John Publishing Inc. 115 King Street West,
Dundas, ON, Canada L9H 1V1

A Technical Comparison of Digital Frequency-Lowering Algorithms Available in Two Current Hearing Aids

By Hugh J. McDermott



About the Author
Hugh J. McDermott is with The Bionic Ear Institute and the Department of Otolaryngology, The University of Melbourne, Victoria, Australia.

ABSTRACT

Background: Recently two major manufacturers of hearing aids introduced two distinct frequency-lowering techniques that were designed to compensate in part for the perceptual effects of high-frequency hearing impairments. The Widex "Audibility Extender" is a linear frequency transposition scheme, whereas the Phonak "SoundRecover" scheme employs nonlinear frequency compression. Although these schemes process sound signals in very different ways, studies investigating their use by both adults and children with hearing impairment have reported significant perceptual benefits. However, the modifications that these innovative schemes apply to sound signals have not previously been described or compared in detail.

Methods: The main aim of the present study was to analyze these schemes'technical performance by measuring outputs from each type of hearing aid with the frequency-lowering functions enabled and disabled. The input signals included sinusoids, flute sounds, and speech material. Spectral analyses were carried out on the output signals produced by the hearing aids in each condition.

Conclusions: The results of the analyses confirmed that each scheme was effective at lowering certain high-frequency acoustic signals, although both techniques also distorted some signals. Most importantly, the application of either frequency-lowering scheme would be expected to improve the audibility of many sounds having salient high-frequency components. Nevertheless, considerably different perceptual effects would be expected from these schemes, even when each hearing aid is fitted in accordance with the same audiometric configuration of hearing impairment. In general, these findings reinforce the need for appropriate selection and fitting of sound-processing schemes in modern hearing aids to suit the characteristics and preferences of individual listeners.

Citation: McDermott HJ (2011) A Technical Comparison of Digital Frequency-Lowering Algorithms Available in Two Current Hearing Aids. PLoS ONE 6(7): e22358. doi:10.1371/journal.pone.0022358

Funding: The author has no support or funding to report. Competing Interests: The author is an employee of The Bionic Ear Institute and has an honorary appointment in the Department of Otolaryngology, The University of Melbourne, Victoria, Australia. The Bionic Ear Institute (BEI) (www.bionicear.org) is an independent, non-profit, research organisation. The author is the originator of Phonak' SoundRecover scheme, as disclosed in Australian Patent Application number 2002300314 ("Apparatus And Method For Frequency Transposition In Hearing Aids"). There are no other patents, products in development or marketed products to declare. This does not alter the author's adherence to all the PLoS ONE policies on sharing data and materials, as detailed online in the guide for authors. **E-mail: hmcdermott@bionicear.org*

INTRODUCTION

Two major hearing-aid (HA) manufacturers have recently introduced frequency-lowering sound processing schemes. Although these schemes are technically dissimilar, they are both intended for HA users who have relatively poor hearing at high frequencies. Lowering selected high-frequency components of sound has been shown to help some people with hearing impairment to perceive them [1,2]. The perceptual benefits potentially include improved ability to resolve and discriminate between sounds as well as to detect them. As is well known, many people with sensorineural impairment have poorer hearing at high frequencies than at lower frequencies, as indicated by hearing sensitivity recorded on a pure-tone audiogram. In such cases, other aspects of auditory perception in addition to sound sensitivity are often affected. For example, frequency resolution, which is related to a listener' ability to separate a signal of interest such as speech from a background noise, is generally found to be poorer at frequencies having worse thresholds [3]. As a consequence, amplification by a HA may fail to enable every hearing-impaired listener to identify all sounds reliably, even though the audibility of those sounds is usually improved. Although various frequency lowering schemes have been developed over several decades in attempts to address these problems, only two schemes are presently in widespread use.

The purpose of the present study was to measure and report the technical characteristics of these recently introduced digital frequency-lowering schemes. The first scheme was devised by Widex, a company based in Denmark, and is known as the Audibility Extender. It is a linear frequency transposition (LFT) scheme that has been reported to improve the understanding of some phonemes in speech, at least after training. For example, identification of fricative consonants increased by about 14 percentage points, on average, for eight adults after two months of use [2]. The second scheme, called SoundRecover, is available from Phonak, a company based in Switzerland. It is a nonlinear frequency compression (NLFC) scheme that was developed after promising perceptual results were reported for an experimental prototype [4]. Similar results have been published more recently [1]. They showed, for instance, that activation of the NLFC scheme increased mean scores by about 15 percentage points for 13 adults and 11 children in a test of plural-noun identification based on detection of a final /s/. The findings of the present study provide technical explanations for the perceptual benefits reported with use of both the LFT and NLFC frequency-lowering schemes.

FREQUENCY-LOWERING TECHNIQUES

The Widex LFT scheme functions by shifting components of sounds present within a source octave into a predetermined target octave [2]. As described in the Materials and Methods section, the settings chosen for the measurements reported below defined the source octave to encompass 2.5–5.0 kHz, and the target octave to be one octave lower (i.e., 1.25–2.5 kHz). In the LFT scheme, the contents of the source octave are analyzed periodically to identify a dominant spectral peak. The frequency of that peak is determined, and the amount of lowering is calculated such that the selected frequency is shifted down by one octave. Other frequency components in the source octave are shifted by an equal number of hertz. For example, if the peak frequency is 4 kHz, the extent of the downward shift is 2 kHz, resulting in the peak component being lowered to 2 kHz. At the same time, a source component at 5 kHz would be lowered by 2 kHz to 3 kHz. Note that, in general, only the frequency of the peak is shifted by exactly one octave. Consequently, it is possible that some components in the source octave would fall outside the target octave after shifting. For instance, in the above example a source component at 3 kHz would be lowered to 1 kHz. However, the signals resulting from the shifting process are filtered to ensure that they remain within the boundaries of the target octave. Thus a source component at 3 kHz would be discarded if the amount of lowering was 2 Hz (or any amount greater than about 1.75 kHz). After transposition, the contents of the target octave are mixed with any sound components already present in the same frequency region. Subsequently the usual processes of amplification, such as amplitude compression, are applied to the composite signal. An important characteristic of the LFT scheme is that the amount of frequency shifting generally varies over time in accordance with the frequency of the dominant peak in the source octave.

The Phonak NLFC scheme is based on different principles [4]. The processing has two adjustable parameters: the cut-off frequency and the frequency-compression ratio. For the present study, a cutoff of 2.3 kHz was chosen. This means that frequencies below 2.3 kHz are unaffected by the NLFC processing, whereas those above are compressed in frequency. The amount of lowering is progressive, such that frequencies much higher than the cut-off are shifted by a larger amount than frequencies only slightly above the cut-off. For example, the selected frequency-compression ratio of 1.7:1 would result in a component at 1.7 oct above 2.3 kHz (i.e., 7.47 kHz) being lowered to a frequency 1 oct above 2.3 kHz (i.e., 4.6 kHz). The transfer function relating input to output frequencies is completely determined during fitting by selection of the above two parameters; it does not change in response to any signal characteristics. Signal components processed by the NLFC scheme do not overlap any other components present at the same time. Together with components below the cut-off frequency, signals that have been compressed in frequency are amplified and additionally processed as usual.

RESULTS

To obtain the measurements reported below, each HA was programmed according to the manufacturer' guidelines to provide an appropriate fitting for a sloping, severe-to-profound hearing loss (see Table 1). The input signals delivered to each HA comprised a sinusoid with slowly increasing frequency, a sequence of notes played on a flute, and four words chosen to contain many phonemes with dominant high-frequency components. Recordings from the Widex HA with and without LFT are available as Audio S1 and Audio S2 respectively, and the corresponding recordings for the Phonak HA are in files Audio S3 and Audio S4. Measurements with Sinusoid Measurements on each HA with the frequency-lowering functions disabled confirmed, as expected, that the gains and output levels were very similar. Therefore, the spectrum for this condition shown in Figure 1 (dashed curve, right panel) is an average of the spectra obtained for each HA separately. The output of each HA for the swept sinusoid (not shown in the figures) conformed generally to expectations of the LFT and NLFC processing functions. For the Widex HA with LFT, the maximum output frequency was approximately 2.5 kHz, corresponding to a 1-oct lowering of the highest frequency in the source octave. For the Phonak HA with NLFC, the maximum output frequency was approximately 4.4 kHz, corresponding to an input frequency of about 6.8 kHz.

The short-term spectra for a brief portion of the swept sinusoid at which the input frequency to the HAs was around 3 kHz are shown in the left panel of Figure 1. The output of the Widex HA with LFT (gray) showed a high-level component at 1.5 kHz, which is 1 oct below the input frequency, as anticipated. Also evident were two lower-level components at 3 and 4.5 kHz which may have been at least partly artifacts of the processing. In comparison, the output of the Phonak HA with NLFC (black) had a single dominant peak at approximately 2.7 kHz, which is the output frequency expected for an input tone at 3 kHz with the selected parameter settings.

MEASUREMENTS WITH FLUTE SOUNDS

The right panel of Figure 1 shows the averaged spectra from each HA with and without frequency-lowering for one of the notes produced by the flute (C5). A 300-ms steady portion of this note was analyzed. As the fundamental frequency was approximately 523.3 Hz, and the signal waveform was essentially periodic,

Table 1. Hearing threshold levels used to program the two hearing aids.

Frequency (kHz)	0.25	0.5	1	1.5	2	4	8
Hearing Threshold Level (dB HL)	50	60	70	80	90	100	100

doi:10.1371/journal.pone.0022358.t001

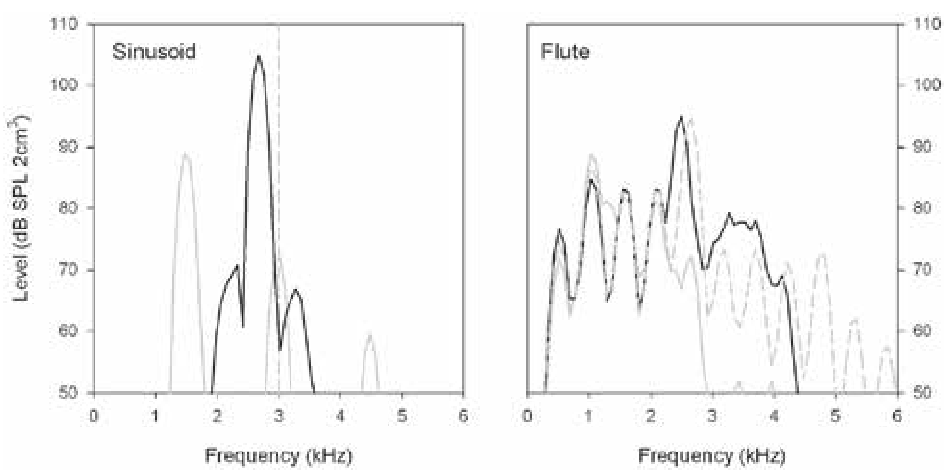


Figure 1. Output spectra of each hearing aid (HA) for inputs consisting of a sinusoid (left) and a tone produced by a flute (right). The sinusoid had a frequency of 3 kHz (vertical dashed line), corresponding to a brief portion of a sweep encompassing the frequency range 0.1–10 kHz. The flute note was C5 (fundamental frequency: 523.3 Hz). In both panels, the black line shows the spectrum from the Phonak HA with NLFC, and the gray line shows the spectrum from the Widex HA with LFT. The gray dashed line in the right panel shows the averaged spectrum from both has without frequency lowering.

harmonics were present at frequencies of 1046.5, 1569.8, 2093.0 Hz, and so on. In both HAs, the first four harmonics produced almost identical outputs for the conditions with frequency-lowering disabled, and, for the Phonak HA, with NLFC enabled. With LFT, the same four frequency components were evident at similar levels, but the fifth harmonic (approximately 2.6 kHz) would have fallen into the source octave. As it was apparently identified as the dominant peak, it was shifted down by 1 oct to about 1.3 kHz. It therefore appeared between the second and third harmonics. There is evidence that a shift of the same amount (i.e., 1.3 kHz) was applied to the seventh harmonic (3.7 kHz) to produce an output component near 2.4 kHz. The unshifted fifth harmonic was also present in the output signal, but Frequency-Lowering Hearing Aids higher frequency components were at much lower levels. With NLFC, the fifth harmonic was shifted down to approximately 2.5 kHz, while the higher harmonics were shifted further and output at lower levels, corresponding to the relatively low level of harmonics above the fifth in the input signal.

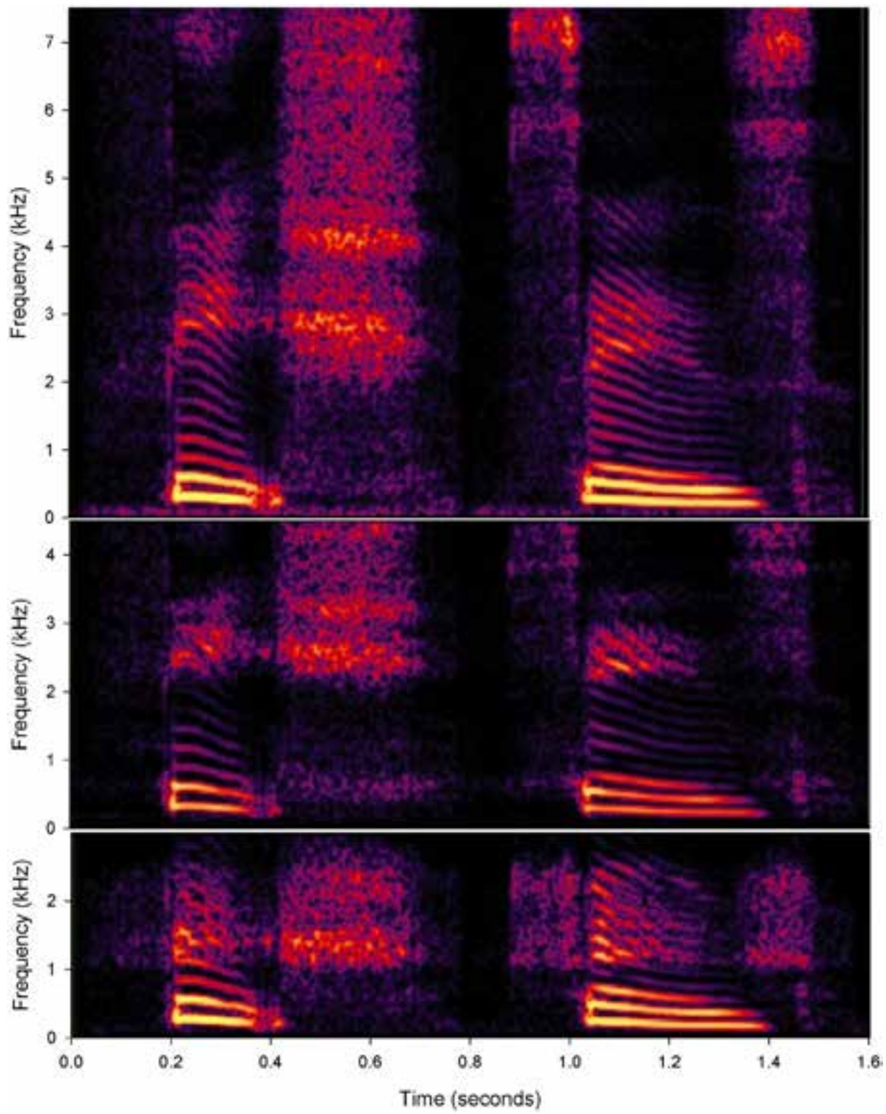


Figure 2. Spectrograms of the word sequence fish, says (top panel) and the corresponding outputs from each frequency-lowering hearing aid (middle: Phonak Nonlinear Frequency Compression; bottom: Widex Linear Frequency Transposition).

MEASUREMENTS WITH SPEECH

Figure 2 shows spectrograms of two of the words used in the tests (i.e., fish, says). The upper panel shows a spectrogram of the original signal, whereas the two lower panels show the outputs of the HAs with NLFC and LFT activated, respectively. The main effect of each type of processing is most evident in a comparison of a vowel sound, such as /i/ in approximately the 0.2–0.4 s portion of the spectrograms, and a consonant sound, such as /#/ in the following portion up to about 0.7 s. Averaged spectra estimated from these two signals are shown in Figure 3. The spectra for /i/ (left panel) were obtained from a 50-ms steady portion near the vowel onset, whereas those for /#/ (right) were obtained from a 200-ms steady portion within the consonant sound. As in Figure 1, the dashed curves show averages of the spectra for each HA with the frequency-lowering functions disabled.

For the vowel, a comparison of the spectra with the frequency lowering functions enabled and disabled shows minimal effect for signal components near the first formant frequency (i.e., around 0.5 kHz). With LFT, components near the second formant (about 2.9 kHz) were lowered to approximately 1.5 kHz. The general effect of linear frequency transposition is clearly evident in that the shape of the spectrum in the source octave above 2.5 kHz with LFT disabled is similar to that with LFT enabled in the target octave below 2.5 kHz. With NLFC, the second-formant spectral peak was lowered to approximately 2.6 kHz, while higher-frequency components were lowered by progressively larger amounts.

For the consonant, the spectrum without frequency lowering shows two local peaks at about 2.8 and 4.1 kHz. With LFT, a peak is evident near 1.4 kHz, presumably corresponding to the lower input peak shifted down by 1 oct. There is also a second, relatively broad peak around 2.2 kHz which seems to have resulted from some combination of shifted and unshifted input components. With NLFC, the two input peaks were shifted to approximately 2.6 and 3.2 kHz, respectively. Some interpretations and implications of these results are discussed next.

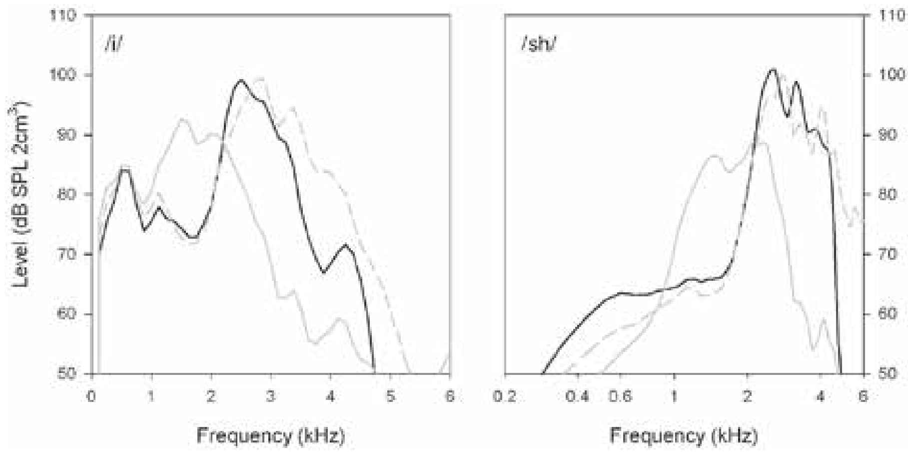


Figure 3. Output spectra of each frequency-lowering hearing aid for inputs consisting of the vowel /i/ (left) and the consonant /#/ (right). Note that the abscissa in the right panel shows frequency on a log axis. Other details are as for Figure 1.

DISCUSSION

In general, the above measurements are consistent with most expectations of the function of both LFT and NLFC processing. The effect of each scheme to reduce the bandwidth of output signals from the HAs is evident particularly in the spectrograms of Figure 2 and the spectra of Figure 3 (right). For LFT, the maximum output frequency was limited by the upper boundary of the target octave (i.e., 2.5 kHz), whereas that for NLFC was approximately 4.4 kHz. Note, however, that the output bandwidth of each HA is effectively adjustable by changing the parameter values of the frequency-lowering functions.

The tests with the swept sinusoid indicated that the Widex HA with LFT enabled produced at least two additional frequency components higher than the one expected from transposition of the input signal. Although this suggests some distortion in the LFT processing, it is likely that the levels of the extra components would be lower than the audibility threshold of a HA user with the audiogram used to program both devices (see Table 1). The tests with the flute sounds suggested that both HAs could provide accurate pitch information to listeners within the lowest four harmonics (including the fundamental) of the signal; see Figure 1 (right panel). Psychophysical studies have found that this frequency range tends to dominate listeners'perception of pitch for complex sounds [5]. Neither frequency-lowering scheme preserved accurate frequency differences between all of the harmonics. However, it seems plausible that the relatively small shift in the frequency of Figure 1. Output spectra of each hearing aid (HA) for inputs consisting of a sinusoid (left) and a tone produced by a flute (right). The sinusoid had a frequency of 3 kHz (vertical dashed line), corresponding to a brief portion of a sweep encompassing the frequency range 0.1–10 kHz. The flute note was C5 (fundamental frequency: 523.3 Hz). In both panels, the black line shows the spectrum from the Phonak HA with NLFC, and the gray line shows the spectrum from the Widex HA with LFT. The gray dashed line in the right panel shows the averaged spectrum from both HAs without frequency lowering the fifth harmonic caused by NLFC processing would be less salient perceptually than the production by LFT of the component near 1.3 kHz. That component is not harmonically related to other components present in the input signal, and, given its comparatively high level, might reduce the ability of some hearing-impaired listeners to resolve the adjacent second and third harmonics.

The spectra obtained using the vowel sound also showed that some frequency ratios (or differences) between spectral peaks were altered by both LFT and NLFC; see Figure 3 (left). As expected, neither scheme changed frequencies near the first formant, but LFT shifted the peak near the second formant to about 1.5 kHz. In contrast, NLFC lowered that peak only slightly, with the result that it remained well within the overall range of second-formant frequencies for this vowel reported from measurements involving many different speakers [6].

Similar observations apply to the spectra of the consonant sounds (right panel). The relatively small effect of NLFC compared to LFT suggests that it might be easier for inexperienced listeners to adapt to the frequency-shifted signals, particularly when listening to speech, at least for the settings applied in the present tests.

In conclusion, both frequency-lowering schemes may provide perceptual benefits to HA users with hearing impairment at high frequencies. Although only one audiogram configuration was applied in the experiments, it is likely that the findings would be generally similar for other audiogram shapes, provided that they represented types of hearing impairment that would be suitable for fitting of either type of frequency-lowering HA. The technical test results suggest that the Phonak NLFC processing may preserve more details of the overall spectral shape than the Widex LFT scheme, at least for the selected signals and settings. However, the LFT scheme may be more suitable than NLFC for HA users with minimal usable hearing at frequencies above approximately 1.5– 2 kHz. This is because the NLFC cut-off frequency is limited to a minimum setting of 1.5 kHz; thus, NLFC is unable to modify lower frequencies. In any case, selection of the optimum fitting for each HA user should depend ultimately on perceptual assessments, including tests of speech understanding in particular.

MATERIALS AND METHODS

The hearing aids used for the present study were the Widex mind440 m4-19 and the Phonak Naı´da V SP. Each was programmed to suit the audiogram shown in Table 1, based on default settings of the fitting software. This audiogram is well within each manufacturer' fitting guidelines for these HAs. Furthermore, it is close to the average audiogram of the subjects who participated in an evaluation of a prototype of the NLFC processing [4], and is within the range of audiograms of the subjects who participated in a published evaluation of the LFT scheme [2]. To ensure that the technical performance of each HA was not inadvertently affected by irrelevant aspects of the fitting, both HAs were programmed to match as closely as possible the gain and amplitude-compression characteristics recommended for this audiogram by the NAL-NL1 prescription [7]. In addition, signal-processing features such as feedback cancelation, noise reduction, and occlusion compensation were disabled, and an omni-directional microphone configuration was selected. These settings were not altered during measurements in which the LFT or NLFC schemes were either enabled or disabled. The selected settings of the frequency-lowering parameters for each HA are shown in Table 2.

Output signals were recorded from each HA in each condition for three types of input signal: (1) a sinusoid swept from 0.1 to Figure 3. Output spectra of each frequency-lowering hearing aid for inputs consisting of the vowel /i/ (left) and the consonant /#/ (right). Note that the abscissa in the right panel shows frequency on a log axis. Other details are as for Figure 1. 10 kHz logarithmically over 10 s; (2) a succession of notes played on a flute; and (3) speech, comprising four monosyllabic words recorded by a female speaker. The average level of all signals was 65 dB SPL. The sounds were delivered to each HA in a Bru¨el & Kjær Type 4222 anechoic test chamber, and the output signals were recorded via a 2- cm3 coupler for later analysis using Adobe Audition 3.0 software. The swept sinusoid, which was used to verify the function of each HA with and without each frequency-lowering scheme, was passed through a low-pass filter with frequency response similar to the longterm average speech spectrum [8] before delivery to the HAs. This ensured that the level across frequency was well within the range at which optimal processing could be expected for each HA. The flute sounds were included to investigate the potential effects of frequency lowering on musical pitch, and comprised a sequence of notes ranging from G4 to G5 (i.e., fundamental frequencies 392–784 Hz). The words in the speech material (thatch, fish, says, verge) were chosen to include eight different fricative or affricate consonants that are common in English and contain important acoustic components at relatively high frequencies.

Table 2. Settings of the frequency-lowering schemes in the two hearing aids.

Widex LFT		**Phonak NLFC**	
Source octave	**Target octave**	**Cut-off frequency**	**Compression ratio**
2.5–5.0 kHz	1.25–2.5 kHz	2.3 kHz	1.7:1

The audio signals recorded from the HAs were sampled at 44.1 kHz with 16-bit resolution. The spectra shown in Figures00201 and 3 were obtained using a 512-point Fast Fourier Transform(FFT) preceded by a Blackman-Harris windowing function. The spectrograms shown in Figure 2 were obtained using a 256-point FFT after the original signals had been down-sampled to 16 kHz.

SUPPORTING INFORMATION

Audio S1 Sound recording from the Widex hearing aid (HA) with the Linear Frequency Transposition (LFT) function disabled. The input signals were four monosyllabic words (thatch, fish, says, verge), a sequence of notes played on a flute, and a swept sinusoid (0.1–10 kHz). (WAV) Audio S2 As for Audio S1, but with LFT enabled. (WAV) Audio S3 As for Audio S1, but for the Phonak HA with the Nonlinear Frequency Compression (NLFC) function disabled. (WAV) Audio S4 As for Audio S1, but with NLFC enabled. (WAV)

ACKNOWLEDGMENTS

The signals analyzed in this study were recorded with the assistance of Martin Rahn (Phonak AG). The Bionics Institute acknowledges the support it receives from the Victorian Government through its Operational Infrastructure Support Program. The constructive comments of a reviewer about an earlier version of this manuscript are appreciated.

AUTHOR CONTRIBUTIONS

Conceived and designed the experiments: HJM. Performed the experiments: HJM. Analyzed the data: HJM. Contributed reagents/materials/analysis tools: HJM. Wrote the paper: HJM.

REFERENCES

1. Glista D, Scollie S, Bagatto M, Seewald R, Parsa V, et al. (2009) Evaluation of nonlinear frequency compression: Clinical outcomes. International Journal of Audiology 48: 632–644.
2. Kuk F, Keenan D, Korhonen P, Lau CC (2009) Efficacy of linear frequency transposition on consonant identification in quiet and in noise. Journal of the American Academy of Audiology 20: 465–479.
3. Moore BCJ (1996) Perceptual consequences of cochlear hearing loss and their implications for the design of hearing aids. Ear and Hearing 17: 133–161.
4. Simpson A, Hersbach AA, McDermott HJ (2005) Improvements in speech perception with an experimental nonlinear frequency compression hearing device. International Journal of Audiology 44: 281–292.
5. Plack CJ, Oxenham AJ (2005) The psychophysics of pitch. In: Fay RR, Popper AN, eds. Springer Handbook of Auditory Research: Pitch. New York: Springer. pp 7–55.
6. Hillenbrand J, Getty LA, Clark MJ, Wheeler K (1995) Acoustic characteristics of American English vowels. The Journal of the Acoustical Society of America 97: 3099–3111.
7. Byrne D, Dillon H, Ching T, Katsch R, Keidser G (2001) NAL-NL1 procedure for fitting nonlinear hearing aids: characteristics and comparisons with other procedures. Journal of the American Academy of Audiology 12: 37–51.
8. Byrne D, Dillon H, Tran K, Arlinger S, Wilbraham K, et al. (1994) An international comparison of long-term average speech spectra. Journal of the Acoustical Society of America 96: 2108–2120.

SIEMENS

Now available as power instruments.

www.bestsound-technology.ca

A world's first: The new Insio binax hearing instruments.

The next generation of BestSound Technology featuring the world's first binaural CICs with enhanced natural directional hearing.

binax™, featuring e2e wireless™ 3.0, is now available in the complete Insio binax™ custom hearing aid portfolio.

binax processes the audio signals from both hearing aids and delivers an individually optimized audio signal back to each hearing instrument. This process actually enhances the natural directional hearing that the pinna provides to the front. We call this achievement binaural OneMic directionality. As a consequence, the wearer can enjoy previously unimaginable speech understanding, especially in challenging or demanding situations.

Insio binax' performance has been maximized also by adding power instruments and, at the same time, its dimensions have been minimized. The new, ergonomic construction allows Insio binax custom hearing aids to sit even more inconspicuously in the ear canal for a higher level of discretion. And they are also more customizable than ever before – increasing the fitting rate.

Insio binax

Furthermore, Insio binax offers wearers maximum convenience. Together with the new easyTek™ and easyTek App™, Insio binax models can be connected to Bluetooth-enabled devices such as mobile phones, mp3 players, TVs etc.

The new Insio binax hearing aids: Taking binaural hearing to a new level.

BestSound™ Technology

Life sounds brilliant.

Yuri Sokolov

In Conversation with Ted Venema

Yuri Sokolov, MS (Hon), PhD, MBA is a researcher, educator, and entrepreneur with over 30 years of business experience. From 1977-1991, he conducted research in Auditory Physiology and Audiology, authored and co-authored over 150 publications in peer-reviewed journals and 7 patents. He received a PhD in Human Physiology in 1986. He also earned a Master of Business Administration degree in 2002.

A pioneer of the market economy in Ukraine, in 1991, he founded one of the first private clinics, Hearing Rehabilitation Centre AURORA, which has grown into the country's leading hearing healthcare provider to patients of all ages. He continues to chair its Board, navigating AURORA through numerous economic storms, and sharing his knowledge with hundreds of hearing health care professionals.

In 1996-1999, he conducted research at the University of Toronto's Institute of Biomaterials and Biomedical Engineering, and in 1999 he founded Vivosonic Inc. in Toronto, ON to commercialize this research. He led Vivosonic to develop and bring to global markets innovative technologies and products. The world's first wireless system for auditory diagnostics, Integrity™ enables non-sedated ABR in infants and young children. Vivosonic has received a number of US patents, Frost & Sullivan's 2007 Technology Innovation Award, and has developed worldwide distribution.
In 2009, he founded Falcon Business Development Ltd. to provide management-consulting services in such businesses areas as strategy, customer-centric organizational design, finance, operations, marketing, and sales. He also shares his experience with entrepreneurs in Canada and internationally.

Public Profile on LinkedIn: http://www.linkedin.com/pub/yuri-sokolov/22/635/801

Yuri, I've known you for the past 15 years or so by now. Your wife Olena took the HIS program at George Brown College, and I never knew until later that she had been an ENT doctor in Ukraine and had a PhD in otolaryngology! No wonder she did so well in the program.

Some time later on, I did collaborative work with you, on the Auditory Steady State Response (ASSR), as you were developing that at Vivosonic. I remember you even had me present to your staff and to a group of audiologists in Toronto, on the Auditory Brainstem Response (ABR), Otoacoustic emissions (OAEs) and other auditory evoked responses. I've always been fascinated with Electrophysiology and my own doctoral dissertation was on assessing low-frequency thresholds with tone bursts. Enough about me however; this is all about you.

You were the one who started Vivosonic, right? Electrophysiology equipment by Vivosonic is selling all over North America and around the globe by now. I think Canadians would do well to know who you are, and what you have lent to our field. The purpose of this interview is the telling of your story, how it all began, how you started Vivosonic, and what you are up to these days. So without further ado, let's get started here.

Ted Venema (TV): I seem to remember you telling me once that you were from Vladivostok. That's on the completely eastern edge of Russia, or the Russian Federation, right? Olena, is she from Ukraine? How did you two meet?

Yuri Sokolov (YS): First of all, I would like to thank you, Ted, for your invitation to an interview and the opportunity to share our experience. It is always a great pleasure to see you. Yes, I was born at an air force base where my father served as an officer. Olena was born in Kyiv, Ukraine. We first met in 1978 at the Laboratory of Clinical Audiology and Vestibulology, Kyiv Research Institute of Otolaryngology, where we both worked. We did a lot of research together on hearing aid fitting rationales, acoustic reflex, ABR, speech perception, and other Audiology areas.

TV: How did you make up your minds to come to Canada, and when did you first arrive?

YS: Back in 1995, a Danish friend of mine, Steen Rasmussen, a pioneer of real-ear measurements (REM), called me. He told that his friends, Prof. Poul Madsen and Prof. Hans Kunov at the University of Toronto's Institute of Biomaterials and Biomedical Engineering (IBBME), were working on a novel combination audiometer and looking for people with the knowledge of both audiology and business to complete and launch it. He said my wife and I would be prefect candidates, as by the time we had already started our first company, Hearing Rehabilitation centre AURORA, which is still up and running. We called Poul and Hans, talked over the phone, and they sent us an invitation to join their Biomedical Acoustics Group at the IBBME. The names of Poul Madsen, the founder of Madsen Electronics (now GN Otometrics), and Hans Kunov were fascinating, and the opportunity to work at one of the world-renowned universities was too exciting to resist the invitation. We gratefully took it and joined the group in the summer of 1996. Shortly afterwards, we were granted the first Gerald Heffernan / Co-Steel Innovation Grant, which was jointly funded by the National Research Council, the U of T, and Dr. Gerald Heffernan himself. Just a bit later, Xinde Li, a very bright engineer, joined our group.

TV: I understand you have a PhD as well as an MBA. Tell me about those two degrees. Why did you want to pursue an MBA after your PhD?

YS: I received my PhD in Human Physiology from the Institute of Physiology of the National Academy of Sciences of Ukraine in 1986. The thesis topic was on binaural speech perception in various acoustic environments. The IBBME project was a combination of technology development and commercialization study. We conducted the commercialization study with the advice from the University of Toronto's Rotman School of Management professors Basil Kalymon and Harvey Kolodny, who were on the Steering Committee of the project. From the study, I realized that I would need a lot of business knowledge and managerial skills to bring the technology to the market, decided to enroll in the part-time MBA Program at the Rotman School, and received an MBA in 2002.

TV: How did Vivosonic get started? Were you always interested in electrophysiology?

YS: Yes, electrophysiology interested me greatly, especially its practical, clinical applications. Interestingly enough, I thought of the cochlea's ability to emit sounds, similar to what a microphone can do, back in 1977 and first recorded them in early 1990's, not much later than Dr. David Kemp published his famous work on OAEs. As to Vivosonic, we started it out of necessity. Initially, we tried to license the technology developed at the U of T, which was based on so-called Kalman Filtering and enabled real-time recording of OAEs and ASSR. However, the companies we approached considered it too scientific, too mathematical, requiring too much work to implement in their devices. Then, my colleagues told me: "Yuri, you started one company, why don't you start another one?" And so we did, started Vivosonic in July of 1999 – with the support from Dr. Gerald Heffernan, professors Basil Kalymon and Harvey Kolodny, who formed the Board of Directors, as well as Sunnybrook Working Ventures Medical Breakthrough Fund (SWVMBF) that was our first institutional investor. Then Bill Lambert and later on Paul Bernards joined the board.

TV: How did you come up with the name "Vivosonic?

YS: I thought of a name that would reflect both medical/biological and technical sides and came up with this name by combining the "Vivo" meaning "live" in Latin, and "Sonic" reflecting on the acoustic nature of the products we were developing.

TV: The ABR systems from Vivosonic are known throughout the industry as being famously quiet. How did that become a major focus for you and Vivosonic?

YS: It came from a thorough market research. Back in 2002, we were working on adding an ABR/ASSR product to our first OAE device, the VivoScan. We conducted an extensive survey of audiologists and ENT doctors in Canada, the US, and internationally asking about electrophysiological tests they needed and the challenges they may be facing when doing electrophysiological testing. The survey findings, published in *The Hearing Review*, overwhelmingly indicated NOISE as the one single biggest challenge. We also consulted many prominent researchers and clinicians, such as Jay Hall, David Stapells, Chuck Berlin, Linda Hood, Yvonne Sininger, and many others. Then we started thinking how we could overcome the problem. First, we thoroughly researched the noise sources and nature in recording auditory evoked potentials. We found that it was coming from three major sources: muscular and ocular artifacts, conducted electrical noise from the power line, and electromagnetic interferences. Given the ABR signals, and even more so ASSR, are very small (in the microvolts and nanovolts respectively), they are so deeply buried in noise that are very difficult to find – as the proverbial needle in a haystack. Then we developed a solution to cope with each kind of noise: The power-line noise was eliminated by employing wireless (Bluetooth) technology, electromagnetic interferences were dramatically reduced by placing the EEG pre-amplifier right on the ground electrode, and physiological artifacts were effectively removed by Kalman Filtering. Kalman filtering is unique in that it does not use a steady or static value whereby to accept or reject an ABR sweep. Instead, it adapts over time to predict or estimate the expected noise that will arrive. The noise estimates are constantly updated over time, with more weight given to estimates that were most accurate. It is therefore, very well suited for repetitive measures made over time, such as the ABR. It was very challenging, but also exciting. Our fantastic engineering team included great minds – Xinde Li, George Long, the late Isaac Kurtz, Aaron Steinman, PhD, John Temprile, Roger Zhang, and many others. The result was very rewarding – seeing those kids tested in their natural state, with no anesthesia or sedation and associated risks.

TV: It's normally thought that you have to be lying down in a quiet state in order to record a decent ABR. Yet, I've seen pictures of babies playing around with electrodes on their heads, and all the while, their ABR is being measured by a Vivosonic system. What's up with that?

YS: Yes, that is the greatest benefit of the technology. As you know well, among other reasons many infants were lost to follow-up after newborn screening due to the need to sedate or anesthetize them in the OR, which is difficult, unavailable in many places, and some children cannot be sedated at all. Our technology enabled non-sedated, awake ABRs, and thus, literally changed the lives of many children.

TV: These "awake ABRs" utilize some unique capabilities found only with Vivosonic. I am told they include the patented Amplitrode®, SOAP™ Adaptive Processing, and VivoLink™ Wireless Recording Technology. Basically in a nutshell, tell us about each of these.

YS: The Amplitrode® is the combination of electrodes and EEG-preamplifier we spoke about earlier. It largely eliminates electromagnetic interferences. SOAP™ stands for Spectrum-optimized Adaptive Processing that optimizes and dramatically speeds up averaging to reduce physiological and other residual noises in electrophysiological recording. The VivoLink™ utilizes wireless Bluetooth communication between the EEG amplifier and recording computer, thus, eliminating conducted power-line interferences. The combination of the three makes recording ABR and ASSR possible in awake patients, children and adults alike.

TV: It's quite an experience with Canadian medical technology commercialization. What would you recommend to fellow innovators, at universities and elsewhere, to bring their inventions to the market? Could you help them?

YS: It's a great question, Ted. I think, while technical and regulatory hurdles are also important, the most challenging is market adoption of a new product. The key is addressing a real medical need/problem and the more painful it is and the stronger solution, the faster the adoption. Then, among many other things, the solution must be easy to use, fit into regulations and clinical guidelines, protocols, supported by clinical studies, properly reimbursed, reliable. Saving health care costs has become also very important. From a business perspective, crucially important is speed to market and revenue – the faster, the better. I find it also very important for scientists, clinicians and engineers developing new technologies to collaborate with a business partner at the early stages. This helps tremendously to avoid costly mistakes, reduce investment, and accelerate the speed to market. Of course, Ted, I'll be happy to help innovators commercializing their technology – it's great for the patients, our economy, and society at large.

TV: You are no longer involved with Vivosonic, is that right? What happened there? Did you simply decide you wanted a new venture?

YS: Yes, we left the company in December of 2009. Olena and I felt that after 10 years, it was time to move on, and also my late mother was very sick at the time, and I wanted to dedicate as much of my time to her as possible. It was also safe to leave the company at that time, as it had fully developed and launched the Integrity™ worldwide, and had been doubling revenues annually for five years in a row. There was an excellent staff, management team, and a very strong board in place. We were very thankful to the board and colleagues for their great support of our transition. Yet, we could not stay still and within a month founded Falcon Business Development, a management consulting company.

TV: So, what are you doing these days?

YS: We share our experience with entrepreneurs, business owners and leaders of companies, predominantly small and medium-sized, in the healthcare, life sciences, medical devices and other fields. Particularly, we help commercialize their novel products, take them to the global markets, as well as help established companies expand, raise funding, grow revenues, export, reduce costs, and grow value. We are also supporting our first company, Hearing Rehabilitation Centre AURORA in Ukraine. We re-designed it into a truly patient-centric organization, re-branded, and help navigating the profound economic storm the country is going through. We give seminars and workshops to colleagues and clinicians across the country. We help finding sponsors for hearing aids – mostly for children and seniors, as well as the soldiers returning from the military actions in the east of Ukraine.

TV: It looks like your experience spans all facets of our industry – from clinical research to healthcare business, from medical device development to global sales. What do you think of the hearing healthcare industry, competitive strategies, now that such retail giants as Walmart, Costco, and Sears entered the field?

YS: Well, Ted, I think there is no magic bullet to compete in this market, but here are some thoughts. In the main stream, I think those large retailers will have to compete not only on price, but also through uniform standards of care across their clinics, basic-to-advanced mid-range instruments, operational efficiencies, capitalizing on economies of scale. Independent private clinics, rather than competing on price, can differentiate themselves by outstanding quality of services and products – in-depth diagnostics, including electrophysiology, individualized approach, and perfectly fitted top-notch premium hearing instruments from the best brands, providing exceptional service, personalization and flexibility, and by achieving the best possible patient outcomes. This strategy, for example, works and helps us navigate our company through the severe economic storm in Ukraine. There are also some niche areas where private clinics may succeed. For example, in rural areas with no giants and chains, or in areas with dense ethnic populations, capitalizing on cultural knowledge and language skills. Still, there is room right in the middle – for large chains of clinics, combining superior standards of care and wide range of instruments with economies of scale. Examples may be European chains KIND, GEERS, Amplifon, Neuroth, and ListenUp Canada.

TV: Yuri, you are involved in the hearing industry, but unlike its many players, you are not a hearing aid manufacturer. You have developed an incredible contribution to electrophysiology, and it's great to know you did it here in Canada. There's not many of you around, and I guess it's safe to say you are one of a kind. Thank you and it is always a pleasure speaking with you! –

YS: Ted, it is always a great pleasure to see you, thank you so much for inviting!

Remembering Raymond Carhart

The Academy Founder Reminisces on the Significance of Raymond Carhart's Accomplishments in the Field of Audiology

By James Jerger

At the end of a long career in audiology, I often think about the many events and the individuals who have made the journey so interesting and worthwhile. Chief among them are memories of Raymond Carhart and the years I spent under his tutelage at Northwestern University.

Carhart and I overlapped paths at Northwestern for 11 years, from 1950 to 1961 (he as mentor, I as graduate student—then faculty colleague). During this period, we had many frank discussions about the fledgling profession he worked so assiduously to nourish and grow. Here are some of my thoughts, based on my memories and reflections, on the significance of Carhart's accomplishments, complemented, I am pleased to acknowledge, by recent phone conversations and subsequent correspondence with Ray's eldest son, Richard Carhart, professor emeritus of physics at the University of Illinois at Chicago.

Raymond Theodore Carhart was born on March 28, 1912, in Mexico City. His father, Raymond Albert Carhart, was a Methodist missionary whose own father, Albert Elijah Carhart, was a temperance preacher. The latter penned a tome with the unique title, *How Booze Was Beaten in a Midwestern State: A Partial Biography*. Foreshadowing a life of bold accomplishment, young Raymond, at the age of 18, hitched a ride on a chicken truck to reach the United States and enroll at Dakota Wesleyan University (founded in 1885 by the Methodist church) in Mitchell, South Dakota. Then, with bachelor's degree in hand, he moved to Evanston, Illinois, for graduate study in the School of Speech (now the School of Communication) of Northwestern University, where he remained for the rest of his life. In 1935, Carhart married Mary Ellen Westfall. Their union produced three sons, Richard, Robert, and Raymond, as well as five grandchildren.

Carhart earned a master's degree in 1934 and a PhD in 1936. His doctoral work, titled "Infra-Glottal Resonances and a Cushion Pipe," included the construction of an artificial larynx. With his degree completed, Carhart joined the Northwestern faculty as a trained speech scientist. Fate, however, was to intervene.

Cordia C. Bunch, perhaps the world's first clinical and research audiologist, completed his doctoral degree in psychology under Carl Seashore, the renowned expert in the psychology of music, at the University of Iowa. Bunch's graduate work, during and after World War I, included the construction of an instrument for testing the threshold of human hearing across the audible frequency range in 1919. He used the device to test several patients in the office of a local otolaryngologist, Dr. L.W. Dean. When Dean moved from Iowa to Washington University in St. Louis, he invited Bunch to join him and to test the hearing of all his otological patients. Bunch accepted the post, moved to St. Louis, and spent the next two decades acquiring pure-tone audiograms on literally thousands of individuals who suffered from a variety of hearing disorders.

Reprinted with permission from Audiology Today 2015 Nov/Dec:32–37.

Bunch's book, ***Clinical Audiometry***, published in 1943, was a systematic attempt to relate the shape and degree of audiometric loss to the nature and site of the hearing disorder. After Dean retired, Bunch accepted an invitation from Northwestern's School of Speech to come to Evanston as research professor in education of the deaf, and to teach courses in hearing testing and hearing disorders. Here, he met and shared his audiometric knowledge with Carhart. When Bunch suddenly and unexpectedly died in 1942 at the age of 57, Bunch's courses were assigned to Carhart, necessitating a rapid shift of emphasis from speech production to speech perception. Years later,Carhart told me that no single person had influenced his career more than C.C. Bunch.

From 1944–1946, Carhart served in the wartime U.S. Army as captain in the Medical Corps. He was assigned to Deshon General Hospital, in Butler, Pennsylvania, and ordered to set up a system for aural rehabilitation and the dispensing of hearing aids to men and women who had sustained hearing loss as a result of their wartime service. Building on this experience, he returned to the Northwestern faculty in 1947 and established the country's first audiology graduate training program, leading to the doctoral degree. For the next 28 years, he mentored a long list of individuals who have had a significant impact on the profession, beginning with his first graduate student, the late John Keys, who organized the audiology program at the University of Oklahoma, and ending with his last graduate student, Mead Killion, president of Etymotic Research Inc. Throughout his career, Carhart served as valued consultant to many government agencies and programs, especially the Veterans Administration and the National Institutes of Health, until his death in 1975 at the age of 63.

This is the bare-bones outline of Carhart's career, but it was only the beginning of my search to understand the deeper implications of the life of this remarkable audiological pioneer.

A TOUGH ASSIGNMENT

Upon his arrival at Deshon Hospital in 1944, Carhart was directed to develop a meaningful system for dispensing hearing aids to U.S. Army personnel returning from the battlefields of World War II. Looking back from our present vantage point, it is difficult to grasp the difficulty of the task facing him. Nothing like this had ever been attempted on such a scale. How do you decide what is the appropriate hearing aid for a particular person with a particular loss? Previous approaches to the problem, in the civilian population, had been limited to matching the gain of the aid to the degree of audiometric loss. This usually fell short of the ideal solution for a given individual, but what else could you do? Well, thought Carhart, quite a bit. The U.S. Army had chosen the right man for the task. In his 1943 book, Bunch said that the information provided by the pure-tone audiogram alone was not enough to guarantee a satisfactory matching of the hearing aid to the listener. Bunch was convinced that it was equally important to understand how well the hearing aid helped the user to understand speech; he regretted that there was no standardized tool available for this purpose in the 1930s.

But Carhart, a previous student of both speech production and speech perception, had been following closely the wartime research from the Defense Research Laboratory (DRL) at Harvard University, headed by Hallowell Davis. One body of this research focused on the problem of quantifying how well a pilot in an aircraft could understand the speech messages transmitted through radio from an operator on the ground and vice-versa. In those days, radio communication between aircraft and ground was constantly plagued by static and competing engine noise. How well, the Harvard researchers asked, was such a degraded signal-in-noise understood by persons at either end of the communication channel? For this purpose, they had developed, among several other auditory tests, two types of word lists to quantify speech understanding—two-syllable words with equal stress on each syllable (the spondee lists), and one-syllable words in which the various phonemic elements of the English language were distributed in equivalent proportion to their orthographic occurrence in the written language (the phonemically balanced, or PB lists).

Carhart, the consummate scientist, realized that to accomplish his mission, he needed to be able to measure, not only whether the hearing aid made speech loud enough, but how well it also made speech understand-able. He realized that much of the laborious research making this possible had already been done for him at the Harvard Psychoacoustic Laboratory for the ground-to-aircraft problem. But Carhart visualized a different communication system—one in which the tester, presenting words, took the place of the ground operator. The hearing aid was substituted for the radio system, and the hearing-impaired person, isolated from the tester by means of a sound booth, took the place of the pilot in the aircraft. Thanks to the DRL word lists, one could assess (from the standpoint of speech perception) both the adequacy of the degree of amplification and the extent of supra-threshold understandability.

Carhart reasoned that the spondee lists could be used to measure the threshold of hearing for speech through the hearing aid by systematically varying the intensities at which the spondee words were presented, while the PB words could be used to measure how well the user could understand speech presented well above the intensity of his or her speech threshold. This "threshold for speech" he called the "speech reception threshold" or SRT, an acronym still widely used in the profession. The percent correct score for a 50-word PB word list presented at a level well above the SRT, came to be called the "PB Max." Both SRT and PB Max remain in the audiological lexicon more than 75 years after their invention. With these tools in hand, Carhart developed the concept of what came to be called the hearing aid evaluation (HAE): a systematic procedure for selecting a best-performing hearing aid for each candidate. The exact procedure has been frequently criticized over the years, especially for its dependence on small differences in percent-correct scores, but the fundamental idea that, for most users, the best hearing aid for any individual is the one that renders ongoing speech most understandable remains an article of audiological faith. In effect, Carhart invented, out of sheer necessity, what we now call speech audiometry. And speech audiometry has grown, over the years, to become one of the unshakable pillars of the profession.

Carhart was also one of the early leaders in calling for systematic auditory training to complement hearing aid use. He incorporated such a procedure in the Deshon program, and, in later years, regretted that it had not enjoyed a subsequent robust acceptance by the profession. He would have been pleased to see how cochlear implants revived, in the 1980s and 1990s, a strong interest in and research on auditory training.

PARTNERSHIPS

In the late 1940s and early 1950s, the concept of audiology as an independent profession was still a very new idea. There were, among the younger members, pervasive fears that it would be swallowed up by an older, well established profession—the most likely candidate was otology. This had been the medical specialty responsible for persons with hearing problems for hundreds of years. Would otologists allow a non-medical specialty to invade their territory? Optometry, a good deal older than audiology, was still locked in conflict with ophthalmology over rights and borders.

Shortly after Carhart had returned from his U.S. Army service, he approached the problem locally by forming an alliance with Dr. George Shambaugh, chairman of the otolaryngology program at the Northwestern University Medical School. In order to convince Shambaugh that audiologists could complement otolaryngologists in the overall rehabilitation of hearing-impaired patients, he organized a weekly clinic in which four to six patients from Shambaugh's private practice were jointly evaluated and counseled by a team consisting of Shambaugh, with one to two of his residents and Carhart with two to three of his graduate students. The venue was a suite of offices in the Northwestern University Medical School on the Chicago campus every Wednesday morning. Many of my most interesting conversations with Carhart took place on those weekly drives from the Evanston campus to the Chicago campus.

Each patient was examined otologically by Shambaugh and his residents, and audiologically by Carhart's team of graduate students. After all patients had been examined by both specialties, each patient was jointly counseled. Shambaugh explained the nature of the medical findings and the possible options open in that direction. Then Carhart summarized the audiological findings and explained the non-medical treatment options. Throughout this joint counseling session, there was a small audience consisting of the residents, the graduate students, and other audiology students from the Evanston campus. I cannot tell you how impressive this was to a young graduate student like me, and to so many of the graduate students who followed me at Northwestern. These clinics, and the camaraderie and mutual respect they engendered among all participants, was, I think, important to the subsequent sense of partnership rather than hierarchy that developed between audiology and otology over the next decades. Enlisting in the partner-ship

George Shambaugh, an internationally prominent otolaryngologist of the period, was one of the many maneuvers Carhart successfully made to advance the fledgling profession. I think he knew that Shambaugh would spread the word among his colleagues that Carhart and his cohorts had a good deal to offer hearing-impaired people. I think he reasoned, also, that the many audiology students who were exposed to these clinics would carry the message to their new posts and widen the partner-ship across the nation.

Hugh S. Knowles, the developer of Chicago-based Knowles Electronics, was an internationally prominent acoustical engineer. He is perhaps best known to audiologists for his development of increasingly subminiature microphones and receivers so common in contemporary hearing aids, and for the development of the Knowles Electronic Manikin for Acoustical Research (KEMAR). He was also a close friend of Raymond Carhart. They served together for several years on the Veterans Advisory Committee on Hearing Aid Performance and greatly respected each other's commitment to improve resources for hearing-impaired persons.

I first met Hugh Knowles in 1953. I was setting up the apparatus for my dissertation study and encountered a problem with the electronic circuitry. I reported the problem to Carhart, who promptly picked up his phone and made a call. A few hours later, Hugh Knowles appeared at the old Speech Annex Building on the Evanston cam-pus, ready to assist me. It was many years later before I realized the incredible electronic fire-power that Carhart had brought to bear on my minor problem. But that was typical of Carhart's concern for his students. I suspect that, like the partnership he forged with otolaryngology, Carhart viewed his friendship with Knowles as an opportunity to nourish a similar partnership with the hearing aid industry, recognizing its ultimate importance to the development of audiology.

IN THE DEPARTMENT OF NO ONE IS PERFECT

In July 1952, the *Scientific American* magazine published an article by Morgan Sparks explaining the newly developed transistor, its underlying principles, and how it was impacting the world of electronics. In those days, hearing aids were necessarily much larger and heavier than they are today, because amplification was achieved by vacuum tube circuits powered by a 6 volt "A" battery and a 20-45 volt "B" battery. I showed the article to Carhart, suggesting that transistors might have a profound effect on hearing aids since they required only 1.5 volts from a single battery to produce amplification. After reading the article, he grimaced and said, "No, I doubt it."

EPILOGUE

It has been 65 years since I entered the graduate program in audiology at Northwestern. Times have changed, audiology has changed, the world has changed, but Raymond Carhart's incomparable contributions to our profession will never change.

ACKNOWLEDGMENTS

The assistance of Richard Carhart, Sumitraji Dhar, Mead Killion, and Laura Wilbur in the preparation of this article is gratefully acknowledged.

ABOUT THE AUTHOR

James Jerger, PhD, was for many years a professor of audiology at the Baylor College of Medicine, Houston, Texas, and, later, distinguished scholar-in-residence at the University of Texas at Dallas. He and his wonderful wife of 50 years, Susan, have retired to Lake Oswego, Oregon, where they presently pursue a life of leisure. Dr. Jerger can be reached at jjerger@utdallas.edu.

Reprinted with permission from Audiology Today 2015 Nov/Dec:32–37.

Speech Guard: Providing Audibility and Cues for Auditory Processing

By Thomas Behrens and Kamilla Angelo
Centre for Applied Audiology Research, Oticon A/S

About the Authors
Thomas Behrens is heading the Centre for Applied Audiology Research at Oticon. He is responsible for clinical research, professional communication and definition of the audiology in new products for Oticon. To ensure a broad evidence base, the Centre is typically involved in 12-15 research projects globally on an ongoing basis. Prior to this Thomas spent 10 years as researcher and project manager at the Oticon Research Centre, Eriksholm. Thomas areas of expertise in particular are spatial hearing, signal processing for hearing aids, technical audiology, outcome measures and methods for laboratory as well as field trials. Thomas has a master's degree in Applied Signal Processing and a bachelor degree in Business Administration.

Kamilla Angelo is the senior researcher at Oticon A/S responsible for research activities in Pediatrics & Power Program. Dr. Angelo has a background in the field of sensory neuroscience, from working at University College London, UK and from experimental animal models in her laboratory at the University of Copenhagen. With an electrophysiological approach, Dr. Angelo has studied how sensory stimulation influences the activity of neuronal network in the brain. The past four year, Dr. Angelo has been with Oticon at the headquarters in Denmark. As part of Centre of Applied Audiology Research, she conducts research and clinical studies with external partners on hearing aid technology to the pediatric population. Dr. Angelo is a member of the Eriksholm Scientific Advisory Board.

Normal hearing and hearing impaired listeners use auditory cues of the speech signal to recognize and focus on a speaker of interest in a complex multi-talker backgroun.[1] Given adequate spatial information, the brain efficiently suppress unwanted sounds and encodes desired sounds. It is well know that hearing impairment comes with a reduced perceptual resolution, which is likely to make object formation less precise.[2] In other word, the damage to the peripheral auditory system makes it harder to extract full information of the speech signal. With experience, however, some listeners learn to compensate for a degraded sensor input by "filling in the gap" and by resolving ambiguity with top-down strategies.[3] To give hearing-impaired individuals the best possible conditions for attending to and understanding speech the amplification they use should not restrict the availability of auditory cues.

Amplitude compression in hearing aids automatically adjusts gain and compression based on the intensity of the input signal. The primary purpose of compression is to provide audibility whilst ensuring comfort. Yet, for users to extract the most information from their aided hearing in complex environments the natural intensity contrasts of the original sound should, despite compression, be preserved to the highest degree possible in the output signal of the hearing aid.

Figure 1 provides an example of a sentence spoken in background noise. The temporal information in the speech signal has been partitioned into to features based the fluctuation rates: the slow envelope (pink) and the fine structure (green). The speech envelope, for instance, is used for segregating sound sources, whereas the fine structure is used for example for detailed pitch perception, which helps the listener identify one speaker from another. In simple situations, much of this information is redundant, but in

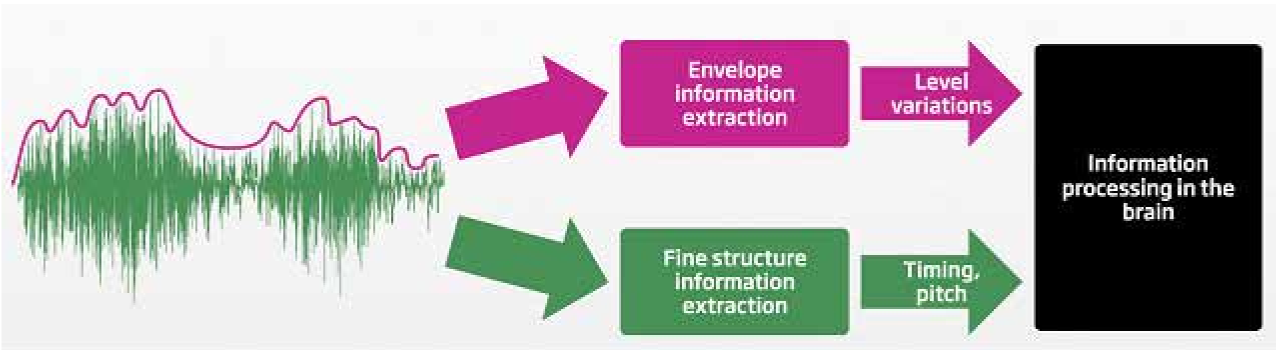


Figure 1: Two types of information present in sound that is used for auditory processing, for instance for identifying the pitch of a speaker and for segregating one sound source from another. Envelope is low frequency and high intensity (pink) and fine structure is high frequency and low intensity (green).

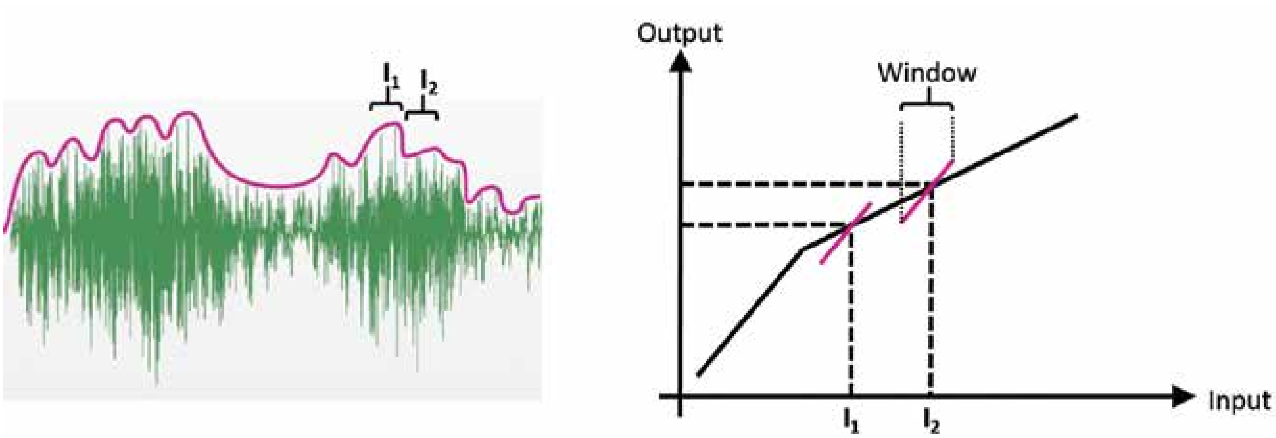


Figure 2: Input-Output curve for Speech Guard illustrating floating linear gain.

more complex listening situations both perception of envelope and fine-structure cues make a difference as how well a listener performs. In the design of compression systems in hearing devices it therefore becomes critical that compression strategies are not only tested under simple listening conditions. Rather, it is necessary to test under complex conditions where the cognitive load of the listener is high and where the brain needs as many salient spatial cues as possible to be able to extract the desired signal.

ADAPTING GAIN TO CHANGES IN THE ENVIRONMENT

Generally, amplitude compression is understood as a technology that reduces the dynamics of sound, such that it is possible to amplify it into the narrow listening range of a person with a hearing loss. However, given the context described above, compression should only be used "globally" to make soft situations more audible and to prevent discomfort in loud situations. "Locally" amplification should be steady (i.e., linear), so the auditory system can benefit from all the amplitude modulations.

Based on this, Speech Guard was originally developed with floating linear gain in mind. Figure 2 gives an example of how floating linear gain is operating. In the first interval, I_1, the level as indicated by the envelope is quite stable and within the window. As a consequence, amplification can be linear for that interval. Right after, the gain suddenly drops and the gain is rapidly adjusted downwards and then remains within the new window during interval I_2. In this way the gain is kept "locally" stable during the input intervals, whilst "globally" adapting according to the prescribed input-output relation.

Speech Guard was introduced in Oticon Agil in 2010 as a combined amplification and compression system that balances audibility, prevents levels from becoming uncomfortable, and maintains amplitude-based information in the input (envelope).

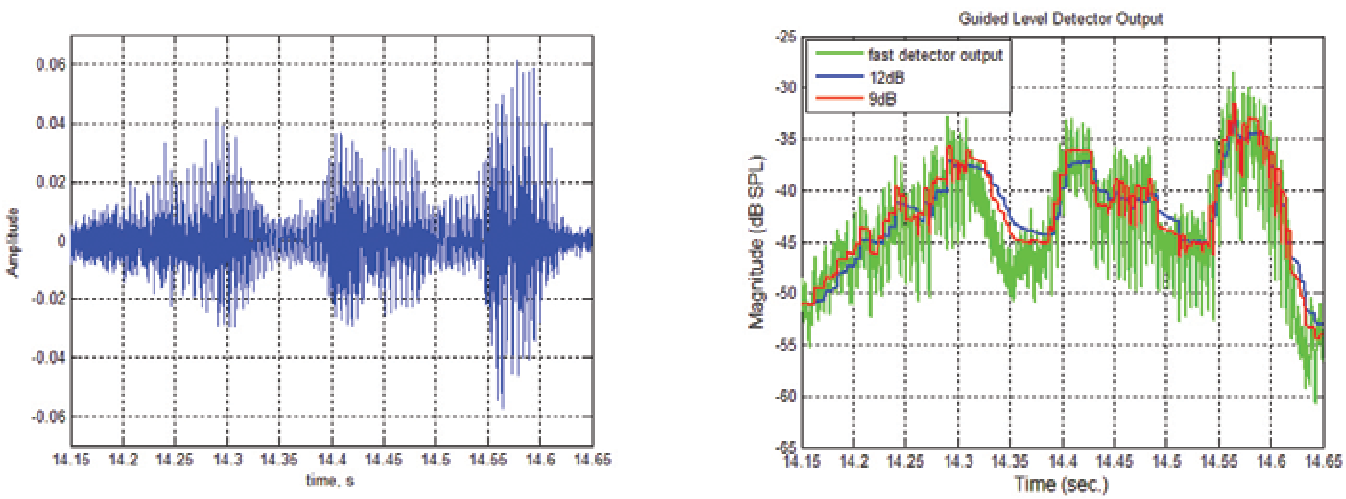


Figure 3. Input speech in noise waveform (left) and resulting level detected (right) in the hearing aid with fixed fast (green), and two versions of Speech Guard with 12 dB (blue) and 9 dB (red) linear window.

To achieve this, Speech Guard is an adaptive system, which continuously compares the current input to the most recent input to determine the status of the situation. Which means:

- If the acoustical environment is relatively stable priority is given to maintaining envelope cues by keeping the gain stable and consequently, the compressor operates with long time constants (in the order of seconds).
- If the acoustical environment changes significantly, it is necessary to adjust gain quickly to ensure audibility or to prevent discomfort. Thus, the compressor operates with short time constants (~ 5 ms and ~ 50 ms for attack and release the time constants, respectively).
- As soon as the situation is stable, the system automatically keeps the gain stable again to prioritize the auditory cues.

Maintaining amplitude-based information in the input is crucial for the listeners to be able to perceive qualities such as voice pitch, to detect onsets and offsets of words and phonemes, to localize with moderate to high levels of reverberation, and to be able to segregate competing voices from each other. To address the latter point, Speech Guard compression in the Alta, Alta 2 and Sensei Pro hearing aids was optimized and is named Speech Guard E. Speech Guard E has an increased range within which it only changes gain slowly. We typically refer to this as the linear window (see Figure 2). Specifically, the linear window was increased from 9 dB to 12 dB. This range better aligns with the parts of the speech signal contributing to speech recognition in the hearing impaired listener.[4] Figure 3 shows how Speech Guard differs from fast compression and how the linear window has been widened to maintain gain stable for steady parts of a speech signal in noise.

The curves in Figure 3 illustrate that Speech Guard gives more steady gain compared to fast compression. Consequently, fast compression will provide changes to the output, also when the level of the input is stable. Both versions of Speech Guard provides steady level estimates, but sometimes they change abruptly reflecting the abrupt changes in the input. Note that the 9 dB curve of Speech Guard (red) fluctuates more than the 12 dB curve of Speech Guard E. With the more steady gain of Speech Guard E the more amplitude modulations of the input signal will be preserved in the output signal.

HOW SPEECH GUARD DIFFERS FROM OTHER COMPRESSION SCHEMES

A number of compression schemes exist in commercially available hearing technology. Some are simple fixed time constant compressors. Others switch between fast and slow compression using simple comparisons, for instance by always selecting the maximum. This has the effect of providing a fast response to sudden loud sounds and avoid promoting background noise. When it comes to advanced adaptive compression ADRO is the only alternative to Speech Guard. ADRO was originally designed for cochlear implants and therefore must be able to keep the dynamics of the output within a very limited range to be able to adapt to the narrow dynamic range of cochlear implant electrodes.

One of the major differences of ADRO and Speech Guard is the use of statistical versus deterministic rules for setting the gain in the device. ADRO continuously keeps track of the amplitude percentiles of speech and ensures that the output does not exceed a comfort threshold more than 10% of the time and does not go below the audibility threshold more than 30% of the time.[5] This means that ADRO determines the amplification based on the statistics of the current situation as well as the recent past.

In contrast, Speech Guard continuously compares the output to the input to determine if the gain setting is appropriate, given the chosen rationale. As long as the input is only changing within the linear window amplification is slowly adapted to increase for lower input and decrease for higher input. If the input is rapidly increasing or decreasing, it will fall outside the linear window and Speech Guard quickly adapt to a new amplification level inside the linear window (as also indicated in Figure 2). This means that Speech Guard only operates based on the current situation and only uses deterministic rules for calculating amplification.

BENEFITS OF MAINTAINING AUDITORY PROCESSING CUES

That maintaining the intensity contrasts of the speech signal is important has been demonstrated by Stone et al.[4] Stone and colleagues tested if listeners with hearing impairment make use of high-rate envelope speech cues in a situation with competing talkers. The rapid modulation gives people with moderate to severe hearing loss access to the pitch of the talker, which significantly contributes to speech understanding, both at low frequencies and at high frequencies. Overall performance increased by approx. 20% when the rapid modulations were present.

Speech Guard E was also compared to fast and slow compression by Pitman et al.[6] To quantify the potential benefits of maintaining envelope cues, the researchers varied the level of complexity of the listening situation. Participants in the study were listening to words and environmental sounds embedded in a background of playground noise. To vary the difficulty of segregating words from the environmental sounds, the stimuli were presented temporally separate or with 50% overlap. The participants performed a self-paced secondary task to increase the cognitive load during the auditory test. This visual task consisted of decoding the logic in a series of shapes (triangles, squares, circles and stars) and from that predict the next symbol. In the simple situation with temporally separate words and sounds fast, slow and Speech Guard E lead to similar performance. In contrast, with temporal overlap of the extraneous sounds and words performance increased significantly (15–20% with Speech Guard E compared to fast and slow compression. This likely reflects that in challenging situations, preserving cues for auditory processing, can help listeners segregate and thereby better focus on the target words while suppressing the interfering background sounds.

To validate that the above findings also applies to hearing aids the impact of Speech Guard E compression on the speech recognition performance of school-age children with hearing loss was investigated. We compared the speech recognition scores from listening with the Oticon Sensei Pro hearing aids to those obtained under conditions with fast wide dynamic range compression (WDRC) (the Safari900 hearing aid) as well as linear amplification (the Sensei instrument with fixed gain) in an ecologically valid listening test. It was found that Speech Guard in the Sensei Pro device gives, on average, a significant speech in noise and reverb advantage of approx. 6.6% above the Safari900 instrument and above linear amplification (Figure 4). We hypothesize that the Speech Guard E compression scheme, with its combination of both linear and non-linear wide dynamic range compression characteristics, strikes a better balance between providing audibility while ensuring minimal alterations of speech cues. The following presents a condensed version of the study. For access to the full description of the study, please see Angelo & Behrens, 2014.[7]

METHOD

PARTICIPANTS AND HEARING AIDS

22 children aged 8 years 1 month to 14 years 11 months (average 11 years, 2 months) with moderate to severe sensorineural hearing loss participated in the study. The children were bilaterally fitted with BTE Oticon hearing aids: the Pediatric Sensei Pro instrument and the Safari900

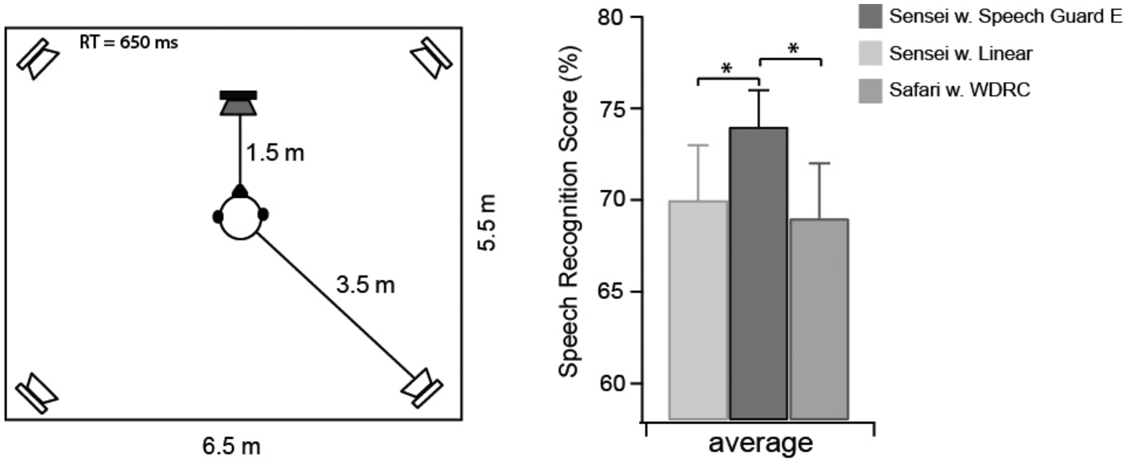


Figure 4: (Left) Listening test setup. Speech recognition performance of the participants was tested at an SNR of + 3 dB in reverberation of 650 ms. Target speech was presented at 0 degrees at 65 dBA SPL. Uncorrelated samples of single talker modulated HINT-C noise were presented at 62 dBA SPL from all four corners of the room. (Right) Effect of Sensei, Safari and linear amplification on speech recognition performance. The mean scores ± error of mean (S.E.M) of all participants are shown as the average across all trials.

instrument. Prior to each child's arrival, the hearing instruments were programmed to DSL v5 targets using age-appropriate RECDs. Individually measured RECDs were obtained when the child was in the lab and necessary changes were made to the programming in the Audioscan Verifit test box. Verification of hearing aids was done using the test box running the speech passage at 55 and 65 dB SPL. MPO was verified using an 85 dB SPL swept tone stimulus. Targets were matched to within 4–5 dB RMS across frequencies from 0.25 to 8 kHz. During lab testing, the settings of the Sensei devices was the following: Adaptation manager always at level 3, binaural broadband OFF, noise reduction disabled and the acoustic setting was Omni. During the acclimatization period, these features were again enabled. Because all advanced features were disabled in the instruments during the lab test, the major factor differentiating Sensei from Safari was the amplitude compression scheme of the instruments. Linear floating gain (i.e., Speech Guard E) is Sensei's compression system whereas Safari used traditional WDRC.

PROCEDURES

The study was done at the Vanderbilt Clinical Trial Center. A blinded repeated measures cross-over design was conducted. Speech recognition behavioral testing using the hearing in noise sentence material, specially developed for use with children, (Hint-C) was completed three times: 1) Pre-acclimatization; 2) After 1st Acclimatization Period with hearing aid set 1; 3) After 2nd Acclimatization Period with hearing aid set 2. The subjects were required to repeat sentences in the presence of both noise and moderate reverberation (RT_{60} = 650 ms). The competing masker noise was the HINT-C noise with single-talker modulation, presented uncorrelated from speakers at 45, 135, 225 and 315 degrees (Figure 4, left). Speech was presented at 0 degrees at a level of 65 dB SPL. All levels were calibrated with A weighting at a fixed intensity of +3 SNR for all conditions. During each visit, three conditions were run in a counterbalanced design: (1) the Sensei instrument with linear floating gain (Speech Guard E); (2) the Safari900 instrument with fast WDRC; (3) A condition where the Sensei instrument was set up to provide linear amplification (i.e. fixed gain according to the gain setting at moderate speech, 65 dB SPL). The "linear" hearing instrument had the same gain-frequency response for all input levels and did not hold the distortion of a compressed signal. This provides the optimal linear reference for maintaining cues for auditory processing.

RESULTS

A total of 21 children completed both trial periods (2–3 weeks each) and all three lab testing sessions. On average, children exhibited mild sloping to moderate hearing loss. Word recognition performance was analyzed using a repeated measures Analysis of Variance (ANOVA) with gain processing (Speech Guard E [Sensei], Syllabic compression [Safari] and linear [Sensei, fixed gain]) and trial (pre-trial, post-trial 1, post-trial 2) as the "within subjects" variables. Any significant main effects and/or interactions were explored ***post hoc*** using linear contrast with Bonferonni correction. Mean word recognition performance for all participants is shown as an average across all lab tests (Figure 4, right). For result at each visit go to Angelo & Behrens 2014.[7] The ANOVA results revealed a significant main effect of gain processing ($F_{2,40} = 8.450$, $p < 0.001$, partial $\eta^2 = 0.297$). Follow-up analysis of the main effect of gain processing revealed significantly better performance for listeners when they were fitted with Sensei than when they were fitted with Safari ($p < 0.009$) or linear processing ($p < 0.011$) (Figure 4). There was no significant difference between performance for individuals when fitted with Safari versus Linear processing. Overall, these results show that on average children demonstrated significantly better word recognition when fitted with Sensei instruments using Speech Guard E compression than the other two types of gain processing. While there was an average benefit of approximately 6 percentage points for Speech Guard E compression over syllabic fast compression processing, considerable variability existed within the data. Specifically, the magnitude of the average advantage ranged from –6 to +16 percentage points. To examine individual differences, the average performance advantage for Sensei over Safari in individual listeners was compared to their receptive vocabulary ability as measured by the Peabody Picture Vocabulary Test (PPVT). The average age equivalent receptive vocabulary was plotted against Sensei's advantage over Safari and partial correlation analysis controlling for participant age revealed a significant negative correlation ($r = -0.470$, $p < 0.036$). These findings suggest that those children with the poorest receptive vocabulary are the most likely to perform better when fitted with Sensei than when fitted with Safari, even when controlling for age.

DISCUSSION

The major finding of this study is the significant increase in word recognition performance that children attain, on average, when wearing Sensei Pro over Safari or linear amplification. Overall, the results demonstrate that school-age children are sensitive to the specific compression characteristics implemented in a hearing instrument. Even when presenting speech and noise at moderate levels only, the way in which gain can be varied to best accommodate the amplification of sound seems to have an impact on the speech recognition in children. So far, very few studies have addressed the effect of WDRC and linear amplification on children.[8]

To our knowledge, this is the first pediatric study that explores the potential benefits of an amplification compression algorithm that deploys adaptive reaction speed, such as the Oticon Speech Guard. We show here that in a realistic acoustic situation, the participants benefited from listening to sound where the intensity modulations of the original speech signal were preserved to a higher degree than can be provided by traditional fast WDRC. Our results indicate that this combination of fast and slow compression seems to be an optimal system, designed to maximize preservation of important auditory information while maintaining audibility of the usable information. The two other hearing instruments with alternative compression strategies that were also tested did not reach the same level of speech recognition as the Sensei instrument. We hypothesize that Safari with syllabic WDRC provided the listener with sufficient audibility. However, the frequent gain adjustments inherent to fast-acting compressors (see Figure 3) are more likely to have introduced temporal distortions in the speech envelope, thus removing the intensity contrasts in the signal and reducing the amount of salient speech cues available to the child. Conversely, with Sensei set in a linear condition (where gain was fixed at all input levels according to gain settings at 65 dB), contrasts in input signals would have been faithfully reflected in output. Nevertheless, moderate speech has a crest factor of approx. 12 dB and this is likely to have caused some loss of audibility for the weaker components of the target signal, ultimately compromising word recognition.

A strong finding of the study is that children with low vocabulary score have the largest benefit of Speech Guard over fast compression. Much research on fast compression[9] finds that the people with the best cognitive abilities have the largest benefit from the technology. This likely is an indication that good cognitive abilities are needed to overcome the processing artefacts of fast compression. That we do not observe such an association for Speech Guard is promising for those who are most in need of a compression system to provide them with a clear representation of the auditory world.

SUMMARY

Speech Guard is an adaptive compression system that provides audibility, comfort and cues for auditory processing to facilitate segregation of speech in noise. In steady or slowly changing situations, Speech Guard keeps amplification almost linear by slowly adapting to the changes in the input. In rapidly changing situations, quick adaptation takes place to ensure audibility and comfort thereafter the compressor transitions back into the slow adaptation. The described studies suggest that adaptive compression does a better job at preserving amplitude modulation than conventional compression. Especially in complex environments, where segregating desired speech from competition this provides the impaired auditory system with what it needs to extract enough information to encode relevant speech and suppress disturbing unwanted sounds. This evidence has been gathered with the help of listeners ranging from the ages of 7 to 78 years of age and with a wide range of prior amplification experience. It shows that it is possible for hearing aids to amplify sound in a way that is not too cognitively demanding, so the users have residual capacity to make sense of sound even in very complex listening environments.

REFERENCES

1. Neher TT, Behrens S. Carlile C, et al. Benefit from spatial separation of multiple talkers in bilateral hearing-aid users: Effects of hearing loss, age, and cognition. Internat J Audiol 2009;48:758–74.
2. Shinn-Cunningham BG and Best V. Selective attention in normal and impaired hearing. Trends Amplif 2008;12:283–99.
3. Pichora-Fuller KM. 2009. Use of supportive context by younger and older adult listeners: Balancing bottom-up and topdown information processing. Int J Audiol 2009;47:S72–82.
4. Stone MA, Anton K, and Moore BC. Use of high-rate envelope speech cues and their perceptually relevant dynamic range for the hearing impaired. J Acoust Soc Amer 2012;132:1141–51.
5. Venema T. Adaptive Dynamic Range Optimization (ADRO): An alternative strategy to WDRC. Can Hear Rep 2015;10(4):16–22.
6. Pittman AL, Pederson AJ, Rash MA. Effects of fast, slow, and adaptive amplitude compression on children's and adults' perception of meaningful acoustic information. J Am Acad Audiol 2014;25:834–47.
7. Angelo K, and Behrens T. 2014. Sensei Pro -Using Speech Guard E to improve speech recognition. Oticon White Paper –Paediatric Knowledge Brief. http://www.oticon.com/~asset/cache.ashx?id=34046&type=14&format=web.
8. McCreery RW, Venediktov RA, Coleman JJ, and Leech HM. An evidence-based systematic review of amplitude compression in hearing aids for school-age children with hearing loss. Am J Audiol 2012; 1:269–94.
9. Souza PE, and Sirow L. Relating working memory to compression parameters in clinically fit hearing AIDS. Am J Audiol2014;23:394–401.

STARKEY **HEARING** ALLIANCE

YOUR SUCCESS IS OUR GOAL

As an **INDEPENDENT HEARING** professional, you can take advantage of our complete array of proven Partner Resources designed to help you grow every facet of your business.

MARKETING

- Retail Marketing Experts
- Appointment Booking
- Practice Websites
- Social Media Strategy
- And More

FINANCE

- Wealth Building
- Financial Services
- Financial Consulting
- Competitive Pricing
- And More

SALE & OPERATIONS

- Experienced Business Planners and Consultants
- Professional Training Opportunities
- And More

HUMAN RESOURCES

- Personal Training Programs
- Exclusive Training Opportunities
- Human Resources Services
- And More

WANT TO LEARN MORE?

Contact your Starkey representative at

1-888-230-5233

www.starkeycanada.ca

Starkey®

Hearing Is Our Concern™