/CN/Chapter 6

/CT/Evidence-Based Practice and Emerging New Technologies

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/H1/ Introduction

A challenge for engineers working in the hearing aid industry has always been to work within the constraints of size and power. When hearing aids were analogue, it would take considerable time to bring an idea from concept to commercial product, as a completely new design of the electronic layout was usually required. In the digital era, however, the journey from concept to commercial product may require no more than a change to the code in the central processing unit (CPU) of the instrument. As a result, new hearing aid processing strategies and features are often introduced every 6 months, typically launched at the large annual meetings organised by the American Academy of Audiology (AAA) in the United States of America in April, and the Union of Hearing Aid Acousticians (EUHA) in Germany in October. The new features are, however, not always directly aimed at improving the hearing performance of hearing-impaired people, but sometimes have the goal of making the instruments more convenient to wear (e.g. synchronisation of volume control and program changes across ears, Powers & Burton, 2005), or easing the clinician's job (e.g., data logging; Fabry, 2005).

The issue for Evidence-Based Practice (EBP), however, is that the introduction of digital technology has not proportionally reduced the time it takes to evaluate a new product in a well-designed study, and to have the findings reach the public domain through a peer-reviewed process. It may come as a surprise that it easily can take up to 30 months for study outcomes to appear in a peer-reviewed journal. The steps involved in designing and executing a comprehensive study that meets the criteria for a high-level and statistically well powered study are listed in Table 6–1. In my experience, the preparations for a project (steps 1-4) can take 6 months or more to complete. Depending on the scope of the study and the number of participants, data collection can take up to 9 months, especially when the study is based on a field test conducted in a crossover design with about 30 participants. It can take 3 months to complete the study and produce a manuscript (steps 6 to 7), and may take longer depending on the complexity of the data set. The review process in step 8 can be expected to take up to 6 months, allowing for one revision, but can take much longer if substantial changes to the manuscript are requested, resulting in several iterations of the review process. Finally, a 6-month wait from the time a manuscript is accepted until it appears in the journal is not uncommon. Unfortunately, such time frames mean that, in the highly competitive environment in which hearing aid manufacturers operate, it is not viable to wait for evidence based data supporting a new technology to be available in peer-reviewed publications before introducing it on the market. This means that some technologies are commercially available long before the evidence for their effectiveness is published. What should clinicians do in such situations when they want to apply EBP to their clinical work? This chapter provides some ideas for where to obtain relevant information about new hearing aid technology in the intervening time, and how to process it. Examples of emerging technologies for which peer-reviewed data are currently very limited, or nonexistent, are introduced. The available data related to these new inventions are discussed, and recommendations are made for which further evidence is needed.

Insert Table 6-1 here

/H1/ When Peer-Reviewed Evidence Is Not Available

This section briefly introduces three procedures the clinician can follow when peer-reviewed evidence is not readily available. Each procedure is presented and discussed in more details in the following sections.

New features in hearing aids are rarely introduced without some supporting data, or at least a description of the motivation for the implementation and the processing strategy. Sporadic data collected either in-house or by an independent institution using a prototype of the final product, or a computer simulation of the invention, may be available at the time a new technology is launched, or shortly thereafter. Such data are often based on simple analytical studies conducted in controlled laboratory environments or short field tests based on a small sample size, and are presented through non-peer-reviewed channels such as white papers produced by the manufacturer or trade journals. The fact that these publications have not been refereed should not exclude them from the review of research evidence (e.g., Cox, 2005), because the decision to bypass the peer review process could be one based on time. Nevertheless, non-peer-reviewed presentations should be appraised with extra care and with special attention to the design and statistical power. These days, online presentations are quickly becoming a popular means of early distribution of information by manufacturers.

As long as mainly non-peer-reviewed publications are available, it is recommended to also look at the rationale and theory behind the new feature and to review the literature on the basic research that has led to the particular signal processing strategy. Although this information does not directly prove that the new strategy is effective for hearing aid wearers, the evidence can be used to obtain a better understanding of what, specifically, the new strategy is expected to achieve, and in which situations it may be beneficial. This knowledge can be used to develop an informed opinion about how likely it is that the new strategy will have a significant impact on a hearing aid user's performance in specific listening environments. An example is outlined in the detailed section below.

A further, and not unreasonable, approach would be for the clinician to obtain some personal experience with the effectiveness of the feature in question from test box measurements and from listening to the hearing aid processing in targeted listening situations. This is evidence, albeit at a low level (Level 6) in terms of the EBP hierarchy in Table 1-5 in Chapter 1, but it may well be the only evidence available.

/H2/ White Papers and Trade Journal Articles

White papers and trade journal articles would not turn up using the database search engines referred to in Chapter 1. The best place to look for the manufacturers' white papers, or any information about a recent release, is on the manufacturer's global Web site, typically addressed www.companyname.com. Most sites will have a section for professionals that is accessed from the main home page. In this section you may find access to a "library," "publications," or "presentations" where available information is found. "Downloads" may be found under the specific products. Note that not all manufacturers engage in publication of white papers, but if they do, these are also most likely available directly from the local supplier. Searching for information on the manufacturers' Web sites provides the best result if the proprietary name of the technology and/or the product in which it is available is known.

The two most common trade journals used to disseminate audiology and hearing aid-related research are *The Hearing Journal* (http://journals.lww.com/thehearingjournal/pages/default.aspx) and *The Hearing Review* (http://www.hearingreview.com). Both provide a search engine and access to their most recent and archived articles free of charge. *Audiology Online* (www.audiologyonline.com) is another website that is increasingly used by manufacturers to distribute information about new products and features in a timely manner, especially through the live and recorded courses hosted by the website. These courses can be accessed through registration free of charge. Fees only apply if you wish to accrue continuing education points for a membership in one of the American or Canadian audiological societies.

Statistical strength and the type of evidence are still the foremost characteristics to examine when scrutinising non-peer-reviewed publications and presentations, but it is unusual to find sufficient information about these characteristics in such publications. The recommendation here is to disregard data sets based on fewer than ten participants (unless the data concern a special population from which recruitment of larger numbers is clearly very difficult), and data presented without any statistical analyses to support the interpretation of the results. Otherwise, the larger the sample size and the lower the significance level is below 0.05, the more confidence one can have in the findings. This is because increasing the number of observations will reduce the variation (i.e., standard deviation) in data and hence a greater effect size is expected. Furthermore, the lower the significance level is, the greater is the chance that the same significant result will be obtained in a repeated study. Consider two studies, A and B, which both show a statistically significant benefit from a directional microphone of 3.5 dB, and a similar spread in data. Because the result of study A is obtained on 25 participants and reaching a significance level of 0.00006 whereas in study B the result is based on 12 listeners and a significance level of 0.02, study A will on its own present a more conclusive outcome. Of course, reviewed together, the two studies would provide stronger evidence for the benefit of directional microphones. Chapters 1 and 2 also provide details of study characteristics to examine when evaluating evidence.

Another factor to look for is how closely the evaluation of the technology under investigation mimics real-life usage. The early studies found in non-peer-reviewed publications are often limited to laboratory experiments. Laboratory tests offer the controllability and repeatability that are essential to quantitative scientific research, and although such data can be informative, the limitations introduced by evaluating the feature in an environment without the dynamic changes, additional cues, and distracters present in most real-world listening conditions should not be ignored. Furthermore, early studies published in either peer-reviewed or non-peer-reviewed journals may be conducted using a computer simulation (offline or real-time processing) of the new signal processing strategy. For example, to investigate the effect of extending the hearing aid bandwidth on speech intelligibility, Moore, Füllgrabe, and Stone (2010) used stimuli that had been processed off-line through a computer-simulated hearing aid. At the test appointment the pre-processed stimuli were presented to study participants through headphones. To evaluate the effect of a new transient noise reduction algorithm, Keidser, O'Brien, Latzel, and Convery (2007) presented stimuli that were picked up by the microphones on a pair of hearing aids on the participant's ear, and directed to a computer for real-time processing before being sent back to the earphone receivers in the hearing instruments. Apart from the microphones and receivers, the hearing instruments were empty shells. In the case of such simulations, it is important to consider whether the invention has been evaluated in isolation or in interaction with common hearing aid signal processing strategies such as wide dynamic range compression (WDRC), feedback cancellation, and noise reduction (NR). If the simulation was presented through headphones or a loudspeaker, the results may be different from those if the same strategy was presented through a hearing instrument due to the possible effect of the microphone placement on the hearing aid and the coupling to the ear.

/H2/ Scrutinizing Basic Research

A prerequisite for following this path is that sufficient information about the feature of interest and its implementation is disclosed by the manufacturer. This information may be found in product brochures or in any publication presenting evaluation data. These publications may also provide preliminary references for the basic research that forms the rationale behind the new strategy. The steps for scrutinizing basic research follow the first three steps described in Chapter 1 for evaluating the evidence directly related to the technology or intervention in question. That is, a specific question is defined, the evidence to answer the question is searched, and the available evidence is evaluated. However, instead of relating the evidence directly to the client, in this case the evidence is related to the claims of the new strategy. The knowledge is used to make an informed decision about how likely the strategy is to be effective for its stated purpose.

For example, recently, a new signal processing strategy that limits directionality to the high frequencies was introduced with the claim that it would help hearing aid users localise sounds better. The strategy is explained in detail and reviewed in a later section of this chapter. To support the claim manufacturers described how spectral cues used to localise sounds get distorted when the ear is occluded with a hearing aid, and presented the results of objective measurements that demonstrated how the new strategy, relative to conventional signal processing , produced a directivity pattern that better resembled that available to the open ear. However, the objective measurements do not guarantee the efficiency of the signal processing strategy. If no evaluation data on hearing-impaired listeners were available to review, the clinician could ask to what extent distortion of spectral cues caused by occluding the ear affect localization performance and how. Such data may provide an insight into how important the new strategy may be to hearing aid users. A literature search would reveal that localization performance by normal- hearing listeners is dramatically reduced when the external ear is occluded in some form to distort spectral cues produced by the pinna (e.g., Gardner & Gardner, 1973; Hofman, Van Riswick, & Opstal, 1998; Musicant & Butler, 1984;). It also has been found that hearing aid users with milder sensorineural hearing loss generally localize sound better unaided than aided with conventional devices (Byrne & Noble, 1998; Van den Bogaert, Klasen, Moonen, Van Deun, & Wouters, 2006). This would suggest that the new signal processing has potential to be beneficial to hearing aid users. However, it is worth noting that Hofman et al. (1998) found that while there was a dramatic reduction in localisation performance immediately after normal-hearing listeners had their ears occluded with a mould, over time while wearing the mould, their ability to localize sounds improved. The finding that the brain can recalibrate to different cues, which is supported by Zahorik, Bangayan, Sundareswaran, Wang, and Tam (2006), could suggest that long-term hearing aid users may not benefit from the new strategy immediately after fitting, or may require some time to show the optimum benefit. Such knowledge is valuable in guiding the expectations of clients.

/H2/ Gather Personal Experience

To explore modern hearing aid features, a hearing aid analyser offering a variety of noise and speech, or speechlike, stimuli and a chamber that allows the hearing aid to be positioned such that the microphone can be reliably aligned with the loudspeaker in any azimuth is required for test box measurements. A hearing aid listening tube or instant fit tips and a field setup with at least two loudspeakers are needed for obtaining a personal listening experience. Test box measurements are best used to verify the effect of a feature and to quantify the resulting response change. For example, with noise presented from 180 degrees, is gain reduced when switching from omni-directional to directional microphone mode? Or, how much is gain reduced when the instrument is exposed to various noise stimuli and the noise reduction strength is set to minimum, medium, or maximum relative to the off position?

Listening experiences provide a better idea of the real life effectiveness of the feature. Ideally, the function of the technology is trialed in the real-world listening situations it is designed to target, but valuable experiences can also be gathered from listening to different simulated environments in the clinic. For this purpose, it is desirable to be able to present speech and noise through different loudspeakers that surround you and allow you to position yourself in different directions relative to the speech and noise. CDs containing different speech and noise stimuli are widely available. It should be noted that test box measurements and unilateral listening tests (e.g. through a listening tube) are sufficient to evaluate most current technologies. However, proper evaluation of newer signal processing strategies, which utilise information about the sound arriving at the microphone on each of two instruments, require a listening test in which the paired devices are worn bilaterally.

/H1/ Emerging New Technologies

New strategies in hearing aids seem to be focused on three areas in particular: (1) improvement of the signal-to-noise ratio (SNR), (2) restoration of binaural hearing, and (3) personalization of the hearing aid processing. It is well established that hearing-impaired people require a better SNR than normal-hearing people for understanding speech in noise (e.g., Plomp, 1978), and that hearing aids distort the binaural cues that are used for localisation of sounds and to ease conversation when with a group of people or in competing noise (e.g., Byrne & Noble, 1998). The former problem has been a research focus for a long time and has resulted in the introduction of increasingly sophisticated directional microphone modes and noise reduction algorithms (e.g,. Chung, 2004), whereas attention to the latter problem is more recent and has been sparked by the possibility of binaurally linking the signal processing in bilaterally fitted hearing aids. There is no doubt that rapid advancement in the area of wireless sound transmission, enabling increased data transmission and faster communication between two hearing aids, opens up new possibilities to both improving the SNR and restoring binaural cues. The motivation to personalize the hearing aid processing arises from the general acceptance of the fact that hearing aid wearers have different needs and different preferences, and that younger hearing aid candidates especially are likely to enjoy taking some control of their instruments. In a world where technology and experimentation are such an integral part of our life, it seems natural to start handing over some of the decision-making processes and controls to the hearing aid wearer.

Table 6–2 lists some of the more recently established hearing aid features that address one of the three areas outlined above and for which there currently is no strong evidence to support the feature as implemented in the final product. By “established,” I mean that the feature is included in hearing instruments by several manufacturers and is thus considered widely available, although the specific implementation of the feature can vary greatly among manufacturers. The following sections provide more background about each of these features, discuss the limited information currently available about the effectiveness of the features, and identify the research needed to support recommendation of these features in hearing rehabilitation. Suggestions for test box measurements or personal listening tests are not made as these would depend on the equipment and real-life opportunities available.

Insert Table 6-2 here

/H1/ New Strategies for Improving the SNR and the Evidence

One of the most recent contributions to this area is the introduction of a feature that enables the hearing aid wearer to direct the sensitivity of the directional microphones to the front, to either side of the head, or to the rear. This means hearing aid wearers do not need to turn their heads to face a speaker situated next to or behind them to obtain an optimum SNR. This may be an advantage in situations where the hearing aid wearer needs to keep looking to the front while conversing with someone situated next to them (e.g., when driving a vehicle). Currently, there are no peer reviewed studies published that have evaluated this feature as implemented in a hearing instrument.

A search in the public domain revealed three evaluation studies of the final product described in publications produced by the manufacturers.[1] One of these has been further published in a trade journal (Nyffeler & Dechant, 2009). In terms of the speech reception threshold (SRT) required for 50% speech recognition in noise, all three studies used a crossover design and at least 20 participants to demonstrate a significant directional benefit with the selective directional focus feature relative to omni-directional and conventional fixed directional microphones when speech was presented from 180 degrees and noise was presented from other directions (Fig. 6–1). None of these publications would be regarded as achieving a high level of

[1] References to manufacturers' own publications are not given in this chapter, because these quickly become outdated and later disappear from the public domain.

evidence (see Chapter 1), mainly due to insufficient statistical interpretation of data; that is, the papers lack clear information about such parameters as the statistical analysis used, variation in data, degrees of freedom, or significance level. However, it should be noted that the average directional benefit measured when speech is presented at 180 degrees azimuth in all three studies is comparative to the directional benefit measured with conventional fixed directional microphones over omni-directional microphones when speech is presented at 0 degrees azimuth (e.g., Chung, 2004). In the condition where the sensitivity of the microphone is directed to the front, the real-world effectiveness of directional microphones is supported by evidence based research (e.g., Bentler, 2005), and therefore, it seems justifiable to expect this new feature to be effective in achieving better speech understanding when the speaker is not situated directly in front of the hearing aid wearer.

Insert Figure 6-1 here

In one of the products evaluated above (Product 1 in Fig. 6–1), the directional focus to either side of the head is achieved by transferring the microphone signal from the chosen focus ear (e.g. right) to the opposite instrument (e.g., left), where the original microphone signal is attenuated by 20 dB (Biggins, 2009). This approach, which significantly increases the SNR presented to the hearing aid wearer, is supported by the findings by Richards, Moore, and Launer (2006). In that study, speech performance of normal-hearing and hearing-impaired listeners was measured when a simulated conversation in a car was presented in various binaural listening configurations under headphones using linear amplification. The study demonstrated that both groups performed best when speech and noise arriving at the ear closer to the talker were presented to both ears. The low number of study participants (<10 in each group) and the simplified representation of the real situation are factors that reduce the impact of this study. In fact, according to Moore (2007), subsequent work has revealed that due to reflections from the windshield, speech in a car is not received mainly in the ear closer to the talker, but also in the ear away from the talker. Some more fundamental questions that have not been answered through the speech testing presented in these publications are: to what extent does the hearing aid wearer find it acceptable to have the audio input intensely focused away from a real-world visual scene? What proportion of hearing aid wearers find the feature useful? Of those who do find it useful, what characteristics do they have, and how often do they use the feature? These questions are important to consider when a clinician discusses amplification options with individual clients—the fourth step in the EBP process, as described in Chapter 3.

/H1/ New Strategies for Restoring Binaural Cues and the Evidence

The cues important for binaural hearing, which is necessary for localising sounds and for being able to focus on a target sound among a myriad of sound sources, are well established in normal-hearing listeners. They include the interaural time (ITD) and level (ILD) differences resulting from the head shadowing effect, and monaural high-frequency spectral differences created by the pinna (e.g., Blauert, 1999). That is, sounds that are not presented directly in front of or behind the listener will arrive sooner and louder to the ear closer to the sound source than the ear further away. Similarly, a sound will have a slightly different spectral shape when arriving at the eardrum depending on the direction from which it enters the ear canal. ITD and ILD cues are used primarily for left/right discrimination and require access to information arriving at each side of the head. In contrast, the monaural spectral cue, which in particular is used for front/back and up/down discrimination, requires access to high-frequency information and an unobstructed ear. Originating directly from the theory, different features currently address different aspects of these requirements and three of these (high frequency directionality, binaural gain and extended high frequency bandwidth) are discussed in this section.

/H2/ High-Frequency Directionality

Natural pinna cues are often not available to hearing instrument wearers, either because of the microphone location on the instruments, which means that sounds are picked up for amplification before they are filtered by the pinna, or because the external ear is occluded with an earmould or the instrument itself. Limiting the microphone directionality to the high frequencies, however, alters the spectral shape of sound as a function of arrival azimuth in a way similar to the unobstructed pinna (Fig. 6–2). This signal processing strategy may therefore enhance front/back and up/down localisation. To date, only one study evaluating the effect of high-frequency directionality has been published in a peer reviewed journal (Keidser, O'Brien, Hain, McLelland, & Yeend, 2009). In this study, localisation performance with high-frequency directionality was compared to the performance with omni-directional and full-band directional microphones in both the laboratory and the field. Data were collected from a relatively small sample (21 participants) in a randomized crossover design. Although objective data collected in the laboratory showed a small but significant ($p = 0.004$) improvement in front/back localisation with high-frequency directionality compared to the other conditions, subjective reports from the field test revealed no effect from this feature. The difference in outcomes again shows the importance of considering both intervention efficacy (i.e., laboratory results obtained in ideal conditions with a specified population) and effectiveness (i.e., real-world results obtained in average conditions with an average population), as discussed in Chapter 1.

Insert Figure 6-2 here

Moving to non-peer-reviewed publications, one study published in a trade journal, which evaluated a different implementation of the feature, confirms the findings above (O'Brien, Yeend, Hartley, Keidser, & Nyffeler, 2010). A more recent trade journal article (Groth & Laureyns, 2011) and a publication from one manufacturer both report studies in which front/back discrimination measured in the laboratory was significantly improved with different implementations of high-frequency directionality. A crossover design was used in all studies with participation of 23, 20, and 14 hearing-impaired listeners, respectively. In O'Brien et al. (2010) all results are interpreted using appropriate statistics with all input and output parameters, including degree of freedom and significant levels, clearly presented. Consequently, the current trend is that limiting directionality to the high frequencies can improve front/back discrimination in a controlled environment, but the effect does not seem to be significant enough to have an impact on hearing aid wearers in real life. As individual variations in performance were noted by Keidser et al. (2009), it would be interesting to conduct research into why this feature is more helpful to some than others and why the feature seems to be less effective in real life than in the laboratory. The effect of high-frequency directionality on localization in the vertical plane also needs to be investigated.

/H2/ Binaural Gain Control

Unsynchronized operation of such common features as WDRC, NR, and adaptive directional microphones in bilaterally fitted hearing instruments can cause distortions to the ILD cue used to resolve left/right localisation (Fig. 6–3). This is because any one of these features will, in each instrument, change gain adaptively in response to the overall level and SNR arriving only at the microphone of that instrument. For example, in the case of WDRC, less gain is applied to the ear closer to the sound source where the sound is louder, whereas more gain is applied to the ear further away from the sound source where the sound is softer, a response that will reduce the natural ILD. The greater the compression ratio in the instruments, the greater the resulting reduction of ILD. Synchronization of noise reduction and directional microphone mode has been possible for some time and is commercially available in several products. More recently, manufacturers have utilized wireless communication between two instruments to coordinate gain changes applied by WDRC in the two instruments to ensure that the typical ILD caused by the head shadow effect is maintained (e.g.,Wilson, Lindley, & Schum, 2007).

Insert Figure 6-3 here

There are currently no peer-reviewed studies that have evaluated the effect of this strategy on horizontal localisation performance. One non peer reviewed paper shows a small but significant ($p < 0.05$) improvement in the left/right discrimination of 30 bilaterally fitted hearing aid wearers when binaural communication between the two instruments was enabled (Sockalingam, Holmberg, Eneroth, & Shulte, 2009). One potential caveat in this study is that the stimulus used for localisation testing was a bird chirp, which is likely to have been a high-frequency weighted sound. It is well established that the ITD cue is dominant as long as the signal has audible components at frequencies below 1.5 kHz, whereas the ILD cue is more pronounced at higher frequencies (e.g., Wightman & Kistler, 1992). This means that as long as there is no distortion to ITD and the sound contains audible low-frequency information, localization performance by the hearing aid wearers should not be severely affected. This is partly supported by Keidser et al. (2006), who demonstrated that although the ILD cues were distorted by WDRC and NR, this did not have a significant effect on left/right localization performance measured on 12 hearing aid wearers using a broadband test stimulus. Consequently, it is possible that the benefit of binaural gain control is limited to high-frequency weighted sounds, which are experienced less often in everyday life. Further research in this area needs to systematically investigate the effect of the feature on everyday sounds with different frequency contents while also taking the effect of the ITD cue into consideration. The real-life benefit of binaural gain control also needs to be verified.

/H1/ Extended High-Frequency Band

Extending the bandwidth of the hearing aid response has been a goal in hearing aid design for a long time and gradual progress has been made over the years as technology made it possible. The parameters driving this development include the responses of the microphone and the receiver and signal processing power. Recent technologic advancement has allowed the hearing aid bandwidth to be extended to 10 kHz. One possible advantage of this feature is access to high-frequency speech components and hence improved speech understanding (e.g., Moore et al., 2010). Because pinna cues used for localisation are most prominent at frequencies above 5 kHz (Fig. 6–4), access to higher frequencies may also improve localization performance in hearing aid wearers, provided pinna cues are available, or somehow restored.

Insert Figure 6-4 here

Byrne and Noble (1998) report some unpublished data from a study in which participants were asked to localise sounds that were amplified linearly up to 10 kHz, using an experimental hearing aid, and delivered through a completely-in-the-canal style mould that left the pinna function intact. They found that a few hearing-impaired listeners with good high-frequency hearing and good unaided vertical localisation performance produced their optimum and near-perfect result when the bandwidth was 8 kHz. Beck and Sockalingam (2010) refer to two recent field tests with participation of 58 and 39 participants in which a device with extended bandwidth was evaluated against a conventional instrument. Both studies reported that the participants found the spatial perception in everyday life significantly improved with the new instrument ($p < 0.05$). These studies do not, however, suggest in which dimension localisation was improved. Additionally, the instruments used in both studies had other features activated that aim at improving the spatial perception, such as those outlined above. Therefore, there is currently very limited information available to lend direct support to extending high-frequency gain for hearing-impaired listeners in order to improve their spatial perception. In fact, Keidser et al. (2009) has speculated that due to long-term deprivation from high-frequency information, hearing-impaired listeners have learned to utilise spectral differences at mid frequencies (1 to 2 kHz) to determine whether sounds are coming from the front or the back. It is also possible that extending the frequency band is most effective when combined with other processing strategies aimed at improving spatial perception.

A current complication in evaluating instruments with extended high-frequency bands is the lack of knowledge regarding how much gain should be prescribed and the difficulty in verifying the hearing aid output across the higher frequencies (e.g., Kuk & Baekgaard, 2009). Further research in this area needs to first address the more fundamental question of how to prescribe and verify the extended amplification characteristics, and second, to demonstrate that gain at higher frequencies, either alone or combined with other processing strategies aimed at enhancing spatial perception, can improve localization in hearing aid wearers. In addressing the latter question, acclimatization should be considered. The issue of acclimatization is also discussed in Chapter 4.

/H1/ New Strategies for Personalizing the Processing and the Evidence

It is highly recommended that hearing aids are fitted using a prescriptive formula for setting the hearing aid parameters. It is also widely acknowledged that the prescribed setting is based on average data and that fine-tuning of the hearing aid parameters based on real-life experiences with the instruments is a crucial part of the fitting process (e.g., Dillon, 2001, pp. 302–320). Fine-tuning in a clinical setting can be complicated and cumbersome. This is partly due to difficulties in determining the particular needs of a client, and partly because the fine-tuned response cannot be immediately verified. Trainable instruments, which give the hearing aid wearer access to a set of controls that enable them to self-adjust selected hearing aid parameters to reach their preferred amplification settings, attempt to address this problem. These devices include a learning algorithm, which collects and stores information about the hearing aid parameter settings chosen by the hearing aid wearer in different listening environments and the acoustic characteristics of the environment at the time a new setting is selected. Over time, the instruments will, based on the acoustic input, use this information to automatically apply the wearer's preferred settings (Dillon et al., 2007). Several commercial products are trainable, of which most offer training of environment-specific overall gain. This is achieved by applying the average volume control setting selected by the wearer over time for each class of sound recognized by the instrument's environmental classifier (e.g., Groth, Nelson, Jespersen, & Christensen, 2008). For example, a wearer may choose to increase the volume relative to the baseline setting when listening to speech in quiet, but to reduce the volume when in noisy environments. If this is a consistent pattern, then over time the hearing aid will automatically set overall gain higher when the instruments' environmental classifier decides that the wearer is listening to speech in quiet, and set gain lower when it decides the environment consists only of noise. Thereafter, further training of the instrument would not be required.

The implementation described above is a very simple representation of trainability that utilises only a very crude classification of sounds based on their acoustic characteristics. If one considers that the acoustics of background noises, for example, vary across multiple dimensions, including intensity, spectral, and temporal characteristics (e.g., Keidser, 2009), it would seem reasonable to expect better outcomes from training algorithms that analyse a continuum of acoustic parameters. At least one manufacturer has recently introduced a training algorithm that enables independent training of input-dependent gain in four channels, which means that the static compression characteristics can be varied in each of four channels and hence the gain-frequency response shape can be changed at various input levels (Chalupper, Junius, & Powers 2009).

A literature search on trainable, or self-learning, hearing instruments, including non-peer-reviewed publications, revealed no studies that had aimed to evaluate the potential real-life benefit of training the device to adjust overall gain in specific environments. The search, however, produced four peer-reviewed articles on the trainable concept. One of these papers (Zakis, McDermott, & Dillon, 2007) describes the evaluation of a sophisticated training algorithm implemented in a programmable research device. This device provided slow-acting

WDRC and modulation depth-based NR in three channels. Using one control, the hearing aid wearer could alternate between three different gain adjustments: overall gain, response slope, and mid-frequency gain. The static compression parameters and the noise suppression strength in each hearing aid channel were trained using information about the intensity level and estimated SNR overall and in each of the three channels, and the preferred gain settings in each channel. After a training period of up to 4 weeks, 13 hearing-impaired people compared their trained response with the baseline response in a double-blind field trial. At the end of the trial, nine participants showed a significant preference for the trained response. Only one preferred the baseline response. The double-blind evaluation was achieved by having the two responses randomly assigned to each of two programs every time the device was switched on, thus totally excluding a biased effect on the preference. This well-designed study therefore suggests that hearing-impaired people can manage a complex control to train the device to produce amplification characteristics that, overall, provided a better listening experience across everyday listening environments compared to a prescribed response. The weaknesses of this study include the low number of test participants, and the fact that participants were not randomly selected from a typical clinical population, but were recruited from a group of volunteers who could manage a nonstandard device and who were willing to wear the body-worn style device in real life.

That hearing-impaired people can handle different and complex control configurations to reliably select their preferred responses in a variety of listening situations was confirmed by Dreschler, Keidser, Convery, and Dillon (2008), whereas three independent studies have shown that the baseline response from which training begins significantly affects the preferred gain setting (Dreschler et al., 2008; Keidser, Dillon, & Convery, 2008; Mueller, Hornsby, & Weber, 2008). All three studies used a randomised crossover design with participation of 21 to 24 hearing-impaired people. In two of the studies (Dreschler et al., 2008; Keidser et al., 2008), gain manipulations were collected in the laboratory using a numerical keypad as the controller and real-time linear processing of sounds that were presented in the free field to the listeners' unaided ears. Consequently, these studies used an overly simplistic representation of trainability. In Mueller et al. (2008), participants were fitted with a commercial product that enabled training of overall gain through the usage pattern of the volume control. Adjustments were made in the field starting from two different baseline conditions that shared the same gain-frequency response shape but differed in overall gain by 12 dB. Although this study suffers from a problem with floor and ceiling effects in the adjustment range from the two starting points, rendering some data invalid, the finding is in agreement with the two laboratory studies. All three studies demonstrated that lower gain levels generally were selected when starting from the baseline setting that provided less gain relative to a baseline setting that provided more gain (Fig. 6–5). Keidser et al. (2008) further showed that while some individuals were able to consistently select the same preferred response from two different baseline responses, most participants had no real preference when selecting from among a wide range of responses.

Insert Figure 6-5 here

Taken together, current publications on trainability would suggest that at least some hearing-impaired people can manage and benefit from trainability, but that the starting point must be appropriately prescribed to reduce the likelihood of wearers who are likely to make few changes to the amplification choosing a potentially harmful setting. It is likely that more complex training algorithms will be introduced in hearing instruments in the future. To build up better evidence-based knowledge about this feature, future studies need to investigate the benefit of trainable instruments in a large clinical population, and to determine the degree of algorithm complexity the average client can handle. From our current knowledge, it seems that some hearing-impaired people are more likely than others to benefit from trainability, in which case determining the profile of candidates for trainability would be a desirable goal. The importance of assessing the individual needs and abilities of people with hearing impairment who seek rehabilitation is discussed in Chapter 3.

/H1/ Summary

In the field of audiology, new technology is often available years before peer-reviewed evidence. In the meantime, clinicians should critically scrutinise the findings of available non-peer-reviewed publications for consistency, with a specific focus on sample size, statistical analyses, and the realism of the test paradigm used (implementation and listening situation). For further information, the literature on basic research supporting the new technology could be reviewed to obtain a better understanding of what the technology is likely to achieve. Finally, with the right equipment, experience with the operation and effectiveness of the technology can be obtained through test box measurements and personal listening sessions.

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/H1/ Figure legends

Figure 6–1. The average directional benefit measured when the directional beam was pointing at the rear of the listener relative to an omnidirectional and a fixed, forward-pointing directional mode. Data that stem from three different studies using two different products, different populations, and sample sizes (*N*) were obtained from the manufacturers' white papers.

Figure 6–2. The directional index (DI) measured as a function of frequency on the Knowles Electronics Manikin for Acoustic Research (KEMAR) when unaided, and when fitted with an omnidirectional, a full directional, and a high-frequency (from 1 kHz) directional microphone. The raw data for these measurements were kindly provided by Siemens Hearing Instruments.

Figure 6–3. The interaural level difference (left minus right ear) measured as a function of azimuth on KEMAR when fitted with two instruments providing linear amplification (no distortion to ILD), WDRC, NR mismatch across ears, and directional microphone mode mismatch across ears. Data are extracted from the Keidser et al. (2006) study.

Figure 6–4. The changes in spectral shape caused by the pinna shadowing effect as the sound (white noise) moves from the median plane in front of KEMAR (0 degrees azimuth), to the side (90 degrees azimuth), and to the back of KEMAR (180 degrees azimuth). The raw data for these measurements were kindly provided by Dr. Jorge Mejia from the National Acoustic Laboratories.

Figure 6–5. The baseline responses (*A and B, thin lines*) and the average preferred responses across 12 different listening environments and 24 participants when starting from each baseline response (*A and B, thick lines*). Data are from the study by Keidser et al. (2008).