# Non-Linear Signal Processing in Digital Hearing Aids*

Thomas Lunner[1], Johan Hellgren[1], Stig Arlinger[1] and Claus Elberling[2]

[1]*Department of Neuroscience, Division of Technical Audiology, Linköping University, Sweden;* [2]*Oticon Research Unit, Eriksholm, Snekkersten, Denmark*

Lunner T, Hellgren J, Arlinger S, Elberling C. Non-linear signal processing in digital hearing aids. *Scand Audiol* 1998;27 Suppl 49: 40–49.

Three different non-linear digital signal processing algorithms were developed: LinEar, DynEar and RangeEar. All three provided individual frequency shaping via a seven-band low-power filterbank and compression in two channels. RangeEar and DynEar used wide dynamic range syllabic compression in the low-frequency (LF) channel, while LinEar used compression limiting. In the high-frequency (HF) channel, RangeEar used a slow-acting automatic volume control, while DynEar and LinEar used compression limiting. Wearable digital signal processing-based experimental instruments were used to evaluate the fitting algorithms under real world conditions with experienced hearing aid users. Evaluation included laboratory testing of speech recognition in noise and questionnaires on sound quality ratings. Results did not indicate one general good-for-all algorithm, but different algorithms resulting in preference and performance depending on the hearing loss configuration. Preference for any of the new algorithms could be predicted based on auditory dynamic range measurements. It was hypothesized that the different preferences were affected by different susceptibility to masking of HF sounds by amplified LF sounds.

*Key words*: Compression limiting, digital signal processing (DSP), dynamic range compression, field test, filterbank, multichannel.

*Address for offprints*: *Thomas Lunner, Ph.D., Department of Neuroscience, Division of Technical Audiology, University Hospital, S-581 85 Linköping, Sweden (Tel. +46 13 22 28 57, fax. +46 13 12 51 42, e-mail: Thomas.Lunner@oto.liu.se)*

## Introduction

Cochlear hearing losses have highly non-linear, frequency dependent characteristics. Usually, sensitivity for weak sounds is impaired, whereas loud sounds are almost unaffected in loudness, resulting in a reduced auditory dynamic range for certain frequency regions, usually high frequencies (HFs). The hearing loss configuration, i.e. the amount of loss at different frequencies, is very individual. Digital signal processing (DSP) hearing aids will provide increased flexibility to compensate for the hearing loss. This motivates audiological research using DSP hearing aids to take advantage of the increased flexibility. Various types of amplification are possible in a hearing aid as follows.

### *Linear Amplification to Compensate for Loss of Audibility*

Linear amplification implies that the gain in a hearing aid is independent of sound level. The primary goal is compensating for the loss of audibility through frequency selective amplification. Because of the limited dynamic range associated with loudness recruitment, most hearing aids include some processing, such as peak clipping or compression limiting, to limit the maximum output levels.

Several methods have been suggested that prescribe a gain between one-third and one-half of the hearing loss (e.g. the National Acoustic Laboratories' [NAL] procedure, Byrne & Dillon, 1986; the Prescription of Gain and Output [POGO] procedure, McCandless & Lyregaard, 1983). However, an inherent problem with linear hearing aids is the need to adjust the gain in different listening situations/environments.

### *Compression to Compensate for Reduced Dynamic Range*

Steinberg & Gardner (1937) suggested that difficulties associated with decreased dynamic range could be reduced by the use of automatic gain control (AGC). AGC systems respond to the signal level, amplifying weak sounds more than stronger ones, resulting in the wide dynamic range of the input signal being compressed into a smaller dynamic range at the output. Therefore, AGC systems are also called ''compressors''. In practice, there are many ways of implementing AGC with different input–output characteristics, attack and recovery times and number of frequency channels. There

* The studies presented here have previously been published in *Ear and Hearing* (1997).

is still no clear consensus concerning the ''best'' method. AGC systems have been designed on the basis of different rationales (e.g. Moore, 1996):

*AVC Systems.* Automatic volume control systems are intended to adjust the gain automatically for different listening situations to relieve the user of the need to adjust the volume control. By making the recovery time of the AGC circuit greater than a few hundred milliseconds, such systems change their gain slowly with changes in sound level.

*Compression Limiting.* Compression limiting systems are often used to limit the maximum output of hearing aids as a means of preventing discomfort at high sound levels. The compression ratio is usually high, as well as the compression threshold. Compression limiters usually have a short attack time (< 5 msec) to respond rapidly to transient sounds. The recovery time is also fairly short (20 to 100 msec) to avoid problems with 'pumping'. Compared to peak clipping systems, which cause distortion, the negative effects of compression limiting are less noticeable (e.g. Hawkins & Naidoo, 1993).

*Syllabic Compression Systems.* Syllabic compression systems are active not only for loud sounds but over the whole dynamic range of input sounds. The system may attempt to make the hearing-impaired person's perception of loudness more like that of a normal listener and ensure that the weaker consonant sounds of speech will be audible without the more intense sounds (e.g. vowels) becoming uncomfortably loud. The term ''syllabic'' refers to the use of short-time constants (typically 20 to 100 msec), i.e. of the same order of magnitude as the duration of individual syllables.

*Multiband Compression Systems.* Several authors (e.g. Barfod, 1978; Mangold & Leijon, 1979; Plomp, 1994; Villchur, 1973) have proposed that compression should be applied separately in two or more frequency bands. This might be beneficial for the following reasons (Moore, 1996): (1) The degree of hearing loss often varies considerably with frequency. Therefore the amount of compression needed varies with frequency. (2) Weak high-frequency components in speech, which can be important for intelligibility, often follow after loud low-frequency components. Compression in two or more separate bands may ensure that these weak high-frequency components are always audible. (3) Multiband compression, as opposed to wideband compressors, can reduce the effects of upward spread of masking (e.g. van Dijkhuizen et al., 1991; Ono et al., 1983). By reducing the gain only in the band where the noise is present, the masking effect of the noise will be reduced without affecting the audibility of the parts of the speech spectrum remote from the frequency of the noise.

### Internal and External Variations of Speech

The amplitude variations of speech can be divided into *external* variations (for example, differences in voice level of or distance to a speaker) and *internal* variations (differences in the level of various phonemes of the speech signal itself) (Plomp, 1994). External and internal variations differ in their time-scales, being large (several hundred milliseconds) and small (less than a hundred milliseconds), respectively. Results collected by Van Dijkhuizen and referred to in Plomp (1994) indicate that compression of the internal variations does not improve speech intelligibility in noise and that the negative effects increase with increased compression ratio and the number of frequency channels. However, the detrimental effects of fast compression on speech intelligibility in noise are small as long as both the compression ratio and the number of channels are small.

### Aims of the Study

The overall aim of this study was to develop a new DSP algorithm with new fitting rationales that were flexible enough to suit a wide range of hearing losses. The new fitting rationales should then be evaluated in real life as well as in the laboratory using a wearable DSP-based platform.

## Experimental Wearable Hearing Aid

### Hardware

The binaural experimental device consisted of a wearable digital unit ($11 \times 7 \times 4.5$ cm) with two in-the-ear (ITE) modules (housing a microphone and a receiver) connected via flexible cables. The wearable unit had a switch to select either of two signal processing algorithms, A or B, and two user-operated volume controls. The test subject did not know which algorithm was represented by A and B. A data-logger function was included for objective recording of the total accumulated on-time for A and B settings and how the volume controls were used as a function of time. The ITE modules were individually manufactured from modified Oticon I24 hearing aids.

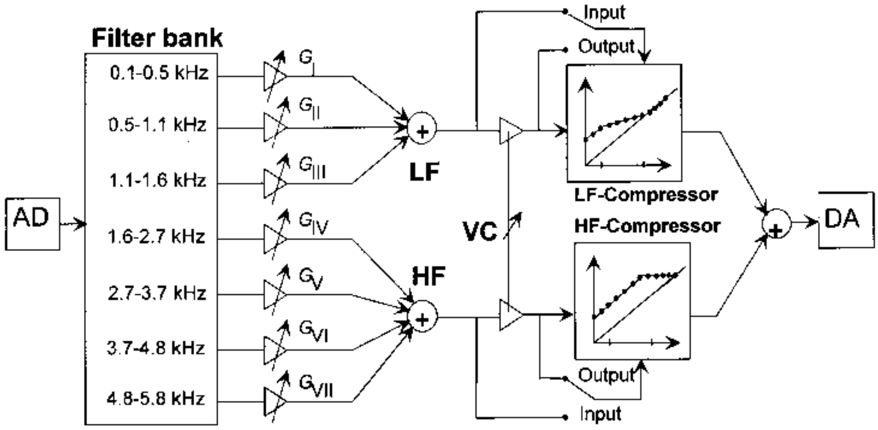


*Fig. 1.* General signal processing algorithm.

### *General Signal Processing Algorithm*

The digital signal processor processed the binaural input signal as two parallel hearing aid processors having a sampling rate of 17 kHz. The adjustable parameters of the general signal processing algorithm were used to form the DynEar or LinEar and RangeEar algorithms (details below). Figure 1 shows a block diagram of the algorithm: the AD-converter signal was sent to a seven-band filterbank (Lunner & Hellgren, 1991) with adjustable gains, $G_{\mathrm{I}}$ to $G_{\mathrm{VII}}$, used to individually shape the frequency response. A low-frequency (LF) channel was formed by summation of the three lowest filter bands and a high-frequency (HF) channel by the four uppermost bands. The cut-off frequency between LF and HF was 1600 Hz. The gain in the LF and HF channels was controlled by the volume control (VC). In DynEar, the VC was active only in the HF channel. The LF and HF signals served as inputs to two independent compressors with adjustable attack and release times.

## New Fitting Algorithms

### *LinEar Algorithm*

The basic characteristics underlying LinEar were the use of a prescribed frequency response, chosen to be POGO II (Schwartz et al., 1988), as a starting-point and then allowing the subjects to individually adjust the gain in the LF and HF channels for best sound quality. In the LF channel this adjustment ($LFgain_{LIN}$) aimed at the most natural perception of their own voice and in the HF channel ($HFgain_{LIN}$) at as clear a reproduction as possible of external sounds.

We chose to adjust the compression limiting in the LF and HF channels to 12 dB *lower* than the pure-tone uncomfortable loudness level (UCL) to account for the loudness summation for broadband (HF) signals at discomfort levels (Bonding & Elberling, 1980). This relatively low setting of the kneepoint might cause some compression for loud speech. In those situations LinEar will behave like a two-channel compression hearing aid. When field-testing the algorithm, a broadband manual volume control was available for the user.

### *DynEar Algorithm*

DynEar was developed on the basis of field experiences using LinEar and with influences from research on syllabic compression (Olsen, 1992; Plomp, 1988). The basic characteristics were:

1. LF channel: Individual frequency shaping and wide dynamic range (syllabic) compression in order to restore normal loudness levels for LFs *and* to reduce the potential risk that loud LF speech elements (vowels) mask faint HF speech elements (consonants).
2. HF channel: DynEar used the same HF channel as LinEar, that is, linear gain with individual frequency *shaping* (POGO II) and a compression limiter with low threshold (UCL-12 dB). This relatively low setting of the limiting threshold might cause some compression for loud sounds in HF.

In order to allow for changes in gain with external variations when field-testing the algorithm, a manual gain control that only operated in the HF channel, $HFgain_{DYN}$ was available for the user.

### *RangeEar Algorithm*

RangeEar was developed on the basis of field experiences using DynEar and LinEar and with influences from

Table I. *Summary of the three fitting algorithms*

| | LF channel | | HF channel | |
|---|---|---|---|---|
| | Compression | Frequency response | Compression | Frequency response |
| LinEar | Compression limiting (UCL-12 dB) | POGO II+ $\text{LFgain}_{\text{LIN}}$ | Compression limiting (UCL-12 dB) | POGO II+ $\text{HFgain}_{\text{LIN}}$ |
| DynEar | Wide dynamic range (syllabic) | Restoration of normal loudness | Compression limiting (UCL-12 dB) | POGO II+ $\text{HFgain}_{\text{DYN}}$ |
| RangeEar | Wide dynamic range (syllabic) | Restoration of normal loudness | Wide dynamic range (AVC/slow AGC) | POGO II |

research on AVC (slow AGC) systems (Plomp, 1994). The basic characteristics were:

1. LF channel: The same LF channel as in DynEar, that is, individual frequency shaping and wide dynamic range (syllabic) compression.
2. HF channel: Individual frequency shaping (POGO II) and slow-acting AGC in the HF channel in order to increase audibility for weak sounds while avoiding discomfort from loud sounds as compared to linear gain *and* to compensate for external variations while allowing internal variations to remain unaltered.

The RangeEar algorithm did not include any user-operated controls, that is, the algorithm was fully automatic. Table I summarizes the main characteristics of the three fitting algorithms.

## Experiments

Three experiments are reported here: (1) DynEar vs LinEar. Pilot study (Lunner et al., 1997a); (2) DynEar vs LinEar. Main study (Lunner et al., 1997a); and (3) RangeEar vs DynEar/LinEar (Lunner et al., 1997b)

Experiments 1–3 were made as a series of field studies comparing a new fitting algorithm with the best possible reference algorithm using two algorithms implemented in the same instrument. The field studies were designed to answer the following questions: (1) Is the test algorithm preferred over the reference algorithm? (2) Are there any particular conditions (e.g., different listening environments, audiometric configurations) for which one of the algorithms is preferred over the other? If so, could this be verified through objective and/or subjective measures?

## Methods

### *Subjects*

The subjects were selected from the patient files of the clinic and asked about their willingness and ability to participate in the study. They all fulfilled the following criteria: well motivated, working or active retired subjects, experienced hearing aid users (of several years' duration). Most hearing losses were moderate. Modest constraints were set on their audiogram configurations, since the frequency shaping via the filterbank was expected to cover a wide range of hearing loss configurations, from mild to moderately severe. Table II summarizes the number of subjects participating (N), number of female (F) and male (M) subjects, as well as mean, min and max age of the subjects.

### *Field Test Procedure*

The two algorithms for comparison were fitted and fine-tuned separately before the actual field test. In the field test, as well as in the laboratory tests, the subjects had to refer to the algorithms as 'A' or 'B', not knowing which fitting algorithm was associated with the letters (blind test). The first two weeks of the field test were training weeks (access to only one of the algorithms, A or B, one week each) and the last two weeks were

Table II. *Number of subjects participating (N), number of female (F) and male (M) subjects, as well as mean, min and max age of the subjects*

| Experiment | N | F | M | Mean age | Min age | Max age | Note |
|---|---|---|---|---|---|---|---|
| 1. DynEar vs LinEar, pilot | 13 | 2 | 11 | 52 | 24 | 72 | Six of the subjects had participated in an earlier experiment |
| 2. DynEar vs LinEar, main | 26 | 8 | 18 | 54 | 25 | 73 | Ten of the subjects had participated in Experiment 1 |
| 3. RangeEar vs DynEar/LinEar | 13 | 4 | 9 | 53 | 26 | 74 | All subjects had participated in Experiment 2 |

comparison weeks (access to both fittings via the algorithm switch). During the field test the subject filled out questionnaires to assess performance. After the field test, preference between A and B was determined and performance was tested in the laboratory.

### Questionnaires

Rating scales were used in questionnaires to rate the sound quality of a number of specified sound environments. The rating scales were Clearness/Dullness, Softness/Sharpness and Overall impression. The subjects filled out three sets of questionnaires at home, one each for algorithm A and B during the training weeks, and one for both algorithms A and B during the comparison weeks.

### Preference Determination

*Logger Preference.* It was assumed that the relative time of usage (as measured by the logger) reflects the algorithm that was preferred. A relative time of usage of more than 60% classified the subjects as having *logger preference* for an algorithm.

*Subjective Preference.* At the very end of the trial period the subjects' preferences for algorithm A or B were assessed by means of a structured interview at the laboratory. The subjects were asked to state overall preference and preference in specified situations using three alternatives: A is best/B is best/I cannot tell any difference.

### Laboratory Tests

Speech recognition in noise was measured by determining the speech-to-noise ratio (S/N) that yielded 40% correct recognition of test words using lists of 10 low-redundancy 5-word sentences (Hagerman & Kinnefors, 1995). The speech material, as well as the competing speech-shaped noise, was recorded on CD and presented through one single frontally located loudspeaker in an audiometric test room. The noise level was variable and performance was tested at a single speech level, 65 dB SPL (C-weighted equivalent level), in Experiment 2, and at two speech levels, 60 and 75 dB SPL, in Experiment 3. In Experiment 2, the subjects' own aids were also tested as an additional measure. These were all binaurally fitted conventional single-channel linear hearing aids with peak clipping.

## Results

In summary, Experiments 1–3 represent about 60 man-months of field-testing experience.

### Experiments 1–2. DynEar vs LinEar

*Experiment 1. Pilot Study.* Preference based on use time (logger preference) showed that no general preference for the DynEar or LinEar algorithms, i.e. no *general good-for-all rationale* was found. However, two groups with different preferred use could be formed from the logger data: six subjects showed LinEar preference and five DynEar preference (and two whose preference could not be determined). Via discriminant analysis, the two preference groups could be identified by auditory dynamic range measures at 500, 1000 and 3000 Hz. Two classification functions were formed from the analysis:

$$\mathrm{C_{DYN}} = 0.78 \times \mathrm{DYNA}_{500} - 0.48 \times \mathrm{DYNA}_{1000} + 0.22 \times \mathrm{DYNA}_{3000} - 18.4 \tag{1a}$$

$$\mathrm{C_{LIN}} = 0.97 \times \mathrm{DYNA}_{500} - 1.08 \times \mathrm{DYNA}_{1000} + 0.68 \times \mathrm{DYNA}_{3000} - 20.7 \tag{1b}$$

The highest score among $C_{\mathrm{DYN}}$ and $C_{\mathrm{LIN}}$ determines the most probable preference, DynEar or LinEar. $DYNA_{nnnn}$ is the dynamic range, UCL-HTL, at frequency *nnnn*, mean values of left and right ears.

A significance test (Manly, 1986) for the ability of the discriminant function to separate between groups showed significance at the confidence level 99%. When the mean dynamic range was greater for the low and mid frequencies and narrower for the HFs, subjects preferred the DynEar fitting over the LinEar fitting. Figure 2 shows the mean dynamic range variables (a) and mean audiograms (b) for the two groups. When inserting individual data into 1a and 1b, all subjects were classified in accordance with their actual preference. Based on the results in Experiment 1, it was hypothesized that the individual preference for either the DynEar or LinEar algorithm could be predicted from the classification functions 1a and 1b.

To test this hypothesis a new group of 16 subjects was recruited for the main study. To confirm that the preference was stable over time, 10 of the subjects who participated in Experiment 1 also participated in Experiment 2.

*Experiment 2. Main Study.* The overall preference and the preferred use (logger preference) agreed, with a few uncertainties. The results showed 12 subjects having *DYNpref*, 13 *LINpref* and one of the new subjects being uncertain about his overall preference. The old group subjects were very consistent in their relative use time of the algorithms between Experiments 1 and 2.

Comparing the actual preference with the predicted preference based on Equation 1, it was found that the prediction was successful in 12 new cases out of 15. The results indicate that DynEar or LinEar preference can be predicted from auditory dynamic range measures. Furthermore, it was shown that the algorithm which provided the best performance was also the preferred algorithm:

1. Across all environments, *DYNpref* subjects rated

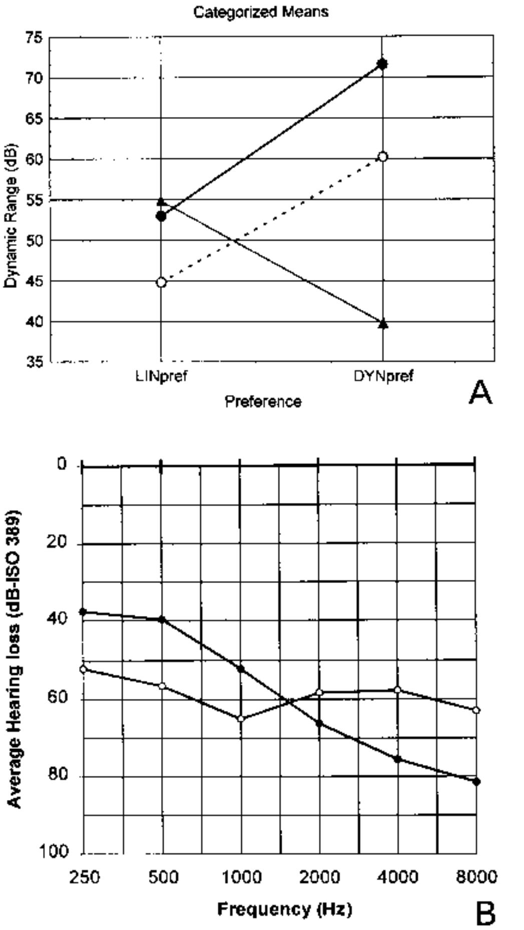


*Fig. 2.* A. Mean dynamic ranges at 500, 1000 and 3000 Hz for the *LINpref* ($n = 6$) and *DYNpref* ($n = 5$) subjects (● = $\text{DYNA}_{500}$, ○ = $\text{DYNA}_{1000}$, ▲ = $\text{DYNA}_{3000}$). B. Mean hearing threshold levels for the *LINpref* and *DYNpref* subjects. Average of left and right ears (○ = LINpref, ● = DYNpref).

DynEar clearer ($p < 0.005$, Least Significant Difference Test, LSD) and with higher overall impression ($p < 0.005$, LSD), and *LINpref* subjects rated LinEar clearer ($p < 0.005$, LSD) and with higher overall impression ($p < 0.005$, LSD).

2. In the speech recognition in noise measurements, an interaction effect ($p < 0.05$, ANOVA) was found between the tested algorithm and the preferred algorithm, with the best performance for the preferred algorithm.

The user-adjusted HF gain, $HFgain_{DYN}$, was notably systematic for the subjects who preferred the DynEar fitting. The results indicated an interaction between the LF and HF channels: Through multiple regression analysis, the actually used real-ear insertion gain in the

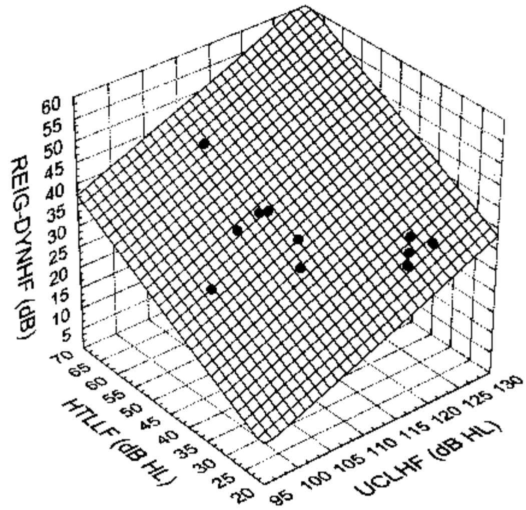


*Fig. 3.* Three-dimensional plot of the plane defined by the regression equation (2). Filled points: individual data.

HF-channel could be described from the hearing threshold levels in the LF channel and from the uncomfortable loudness levels in the HF channel (85% of the variance explained), see Equation 2 and Figure 3. However, the hearing threshold levels in the HF channel were found to be uncorrelated to the actually used real-ear insertion gain in the HF channel, which was unexpected.

$$\text{REIG}_{\text{DYNHF}} = -93.1 + 0.68 \times \text{HTL}_{\text{LF}} + 0.83 \times \text{UCL}_{\text{HF}} \quad (2)$$

Both DynEar and LinEar presented benefit over the subjects' own linear, peak-clipping aids in the non-blind follow up:

1. The preferred algorithms resulted in significantly lower speech recognition in noise thresholds ($p < 0.01$, paired *t*-test). Based on the normative data presented by Hagerman (1984), the difference corresponds to about 20–25% difference in word recognition score.
2. Twenty-five of the subjects judged the digital aid to be superior to their own aids, while one subject judged them equal.

Thirteen of the subjects who participated in Experiment 2 participated in Experiment 3. Since none of the algorithms showed superior performance in general, the LinEar algorithm was used as reference for the LinEar preference subjects and DynEar for the DynEar preference subjects.

### *Experiment 3. RangeEar vs DynEar/LinEar*

As in the previous experiment, the overall preference and the logger preference agreed. On a group basis, no situations were found with preference different from the overall preference. Of the 13 subjects, 6 preferred the RangeEar fitting and 4 the DynEar fitting. Two subjects preferred the LinEar fitting and one had equal preference for RangeEar and LinEar.

By means of discriminant analysis, the three groups with stated preference could be identified by auditory dynamic range measurements in the LF and HF ranges:

$$\text{RangeEar} = -0.01 \times \text{DYNA}_{\text{LF}} + 1.04 \times \text{DYNA}_{\text{HF}} - 27 \tag{3a}$$

$$\text{C}_{\text{DynEar}} = -0.06 \times \text{DYNA}_{\text{LF}} + 0.70 \times \text{DYNA}_{\text{HF}} - 17 \tag{3b}$$

$$\text{C}_{\text{LinEar}} = -0.11 \times \text{DYNA}_{\text{LF}} + 1.30 \times \text{DYNA}_{\text{HF}} - 35 \tag{3c}$$

The highest score among $C_{\text{RangeEar}}$, $C_{\text{DynEar}}$ and $C_{\text{LinEar}}$ determines the most probable preference—RangeEar, DynEar or LinEar. $DYNA_{\text{nn}}$ is the dynamic range, UCL-HTL, in channel *nn*, mean values for left and right ears.

A significance test (Manly, 1986) for the ability of the discriminant function to separate between groups showed significance at the confidence level 96%. Preference was correctly described in 11 out of 12 cases (with stated preference) using classification functions obtained in the discriminant analysis.

The mean auditory dynamic range was broader at LFs, and narrower at HFs for those who preferred the RangeEar or DynEar fitting compared to those who preferred the LinEar fitting. Furthermore, the difference between RangeEar and DynEar preference was determined by differences in the HF range, with the DynEar preference subjects showing a narrower dynamic range (Fig. 4). Some evidence was found that the algorithm giving best performance was also the preferred algorithm.

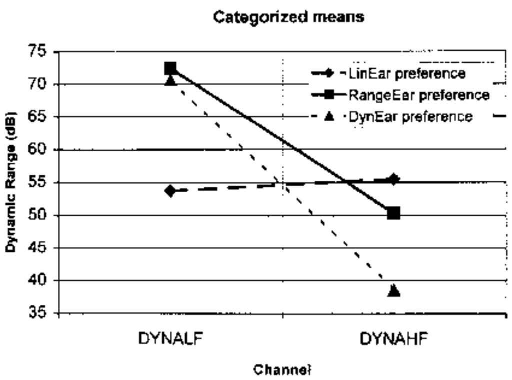


*Fig. 4.* Mean dynamic ranges in LF and HF for the RangeEar preference subjects ($n = 6$), DynEar preference subjects ($n = 4$), and LinEar preference subjects ($n = 2$) subjects. Average of left and right ears.

Interaction effects were found between the tested algorithm and the preferred algorithm in the ratings of overall impression ($p < 0.01$, ANOVA) as well as clearness ($p < 0.05$, ANOVA) (across all environments). The simple effects showed that LinEar preference subjects rated overall impression and clearness of LinEar highest ($p < 0.05$ for both, LSD), whereas RangeEar preference subjects rated overall impression of RangeEar highest ($p < 0.05$, LSD). No significant differences were found for the DynEar preference subjects. The measurements of speech recognition in noise did not show any differences between the algorithms.

## Discussion

*Results do not indicate one general good-for-all rationale.* On the basis of the results in these experiments, we cannot conclude that any one of the new algorithms was generally superior to the others. In Experiment 2, DynEar was preferred by 12 subjects and LinEar by 13; and one was uncertain about his overall preference. However, in Experiment 3, six subjects preferred the RangeEar algorithm, four the DynEar algorithm, two the LinEar algorithm and one was uncertain about his overall preference. In other words, 10 out of 12 subjects with stated preference in Experiment 3 preferred the option of having a wide dynamic range syllabic compressor in the LF channel and having the gain in the HF channel adjustable, either manually (DynEar) or automatically (RangeEar).

*Preference is Stable Over Time.* The relative use times were very consistent for the subjects participating in both Experiment 3 and Experiment 4. Most subjects had differences in the order of a few percent. The subjective and objective preference agreed with some uncertainties in Experiments 2 and 3. No contradiction between logger preference and subjective preference was ever found for any subject. In conclusion, the results show that the stated preference was clearly reproducible and stable over time.

*Preference and Performance Agree.* The preference and performance agreed. The results indicate that preferred fittings obtain higher ratings concerning overall impression and clearness of sound quality. We found some evidence that speech intelligibility in noise was better for the preferred fittings when comparing DynEar and LinEar, but we were unable to show significant differences when comparing RangeEar to DynEar or LinEar.

In conclusion, the results show that sound quality and possibly speech recognition in noise were important factors affecting preference. The results indicate that preference is a reliable measure of the success of an algorithm.

*Preference Depends on Auditory Dynamic Range and Can Be Predicted.* It was found that the average auditory dynamic range was wider at LFs and narrower at HFs for those who preferred the RangeEar or DynEar fitting compared with those who preferred the LinEar fitting. Furthermore, the difference between RangeEar and DynEar preference was determined by differences in the HF range, with the DynEar preference subjects showing a narrower dynamic range. The preference for RangeEar was probably associated with increased audibility of weak sounds in the HF channel while avoiding uncomfortably loud sounds.

Through classification functions we could make post hoc predictions that were 100% correct when predicting DynEar and LinEar preference, and 92% correct post hoc predictions for LinEar, DynEar and RangeEar preference. However, one will *always* obtain better predictions in post hoc predictions as compared to a priori predictions, since in the post hoc analysis the model (classification functions) is based on the *same* subjects as used to evaluate the model. Therefore, *new* subjects have to be used to validate the model.

This was made in Experiment 2. The results indicated that 80% correct a priori predictions could be made for DynEar vs LinEar preference. Therefore, the classification functions may be used clinically as a preselection tool between DynEar and LinEar. We still have to perform a new study on new subjects to verify the prediction strength of the classification functions also including RangeEar.

*Results Contradict Conventional Wisdom.* On the basis of current experiences with wide dynamic range compression, which show a general benefit for compression (e.g. Benson et al., 1992; Moore et al., 1992), it was surprising that the subjects with the most severe hearing loss in the LF channel preferred LinEar. However, we note that LinEar signal processing may act as a two-channel compression device for loud input sounds, because of the low compression threshold setting, so it may not be appropriate to compare LinEar with conventional linear hearing aids.

*Why is Wide Dynamic Range Compression Most Successful for Those with Greatest Dynamic Range in LF?* Since the loudness model was used to determine compression ratios for RangeEar and DynEar LF, both RangeEar and DynEar preference subjects were consequently provided with less compression in LF than those who preferred LinEar. Thus the benefit from syllabic compression in LF was largest for the subjects with the greatest dynamic range who were fitted with the lowest compression ratio (CR) in LF. Subjects with a narrow dynamic range in LF obtained more benefit from linear gain than from syllabic compression with relatively large CRs.

Compression ratios above 2 may be detrimental for speech recognition in noise (Plomp, 1994). The LinEar preference subjects used an average compression ratio close to 3 in the DynEar or RangeEar LF channel. Thus, the reason for the LinEar preferences may not be the absence of LF syllabic compression *per se*, but simply that when applied to subjects with narrow LF auditory dynamic range, too much compression was used. Better results might have been obtained if the LF compression ratios for this group of subjects had been lower than those prescribed by the underlying loudness model.

Thus, as dynamic range (in LF) changes from normal to moderately reduced, it is beneficial to normalize loudness. However, beyond a certain point there should be no further increase in compression ratio. However, it seems more parsimonious to suggest that a moderate degree of compression seems good for everyone and that if CR has to be varied at all then the best basis for doing so could be something other than the loudness function. If CR should not be varied on the basis of loudness normalization, then there must be other reasons for the success of wide dynamic range compression in the LF channel in a majority of our test subjects.

As we try to improve audibility in the LFs, we may run into masking problems at HFs. We have no reason to suspect that the effects of masking by amplification are less for a hearing impaired subject. On the contrary, Gagné (1988) and Trees & Turner (1986) suggest that hearing impaired subjects, at least those with HF hearing loss, suffer from an excess upward spread of masking.

The differences in auditory dynamic range between preference groups also suggest that, whatever phenomenon is causing the difference, subjects with sloping loss compared to those with more flat configuration seem to have more benefit of compression in the LF channel.

Danaher et al. (1973) found that a person's ability to discriminate between second-formant (F2) transitions with and without the presence of an F1 masker is related to the configuration of the hearing loss. Subjects with sloping audiograms showed masking effects of F1, while some subjects in the flat group showed negligible masking effects; others in the same group showed substantial masking.

Furthermore, Summers & Leek (1997) found that by presenting synthetic consonant-vowel stimuli at moderate and high signal levels F1 attenuation of up to 18 dB led to increasing performance among hearing-impaired subjects. Benefit associated with F1 attenuation was particularly evident for listeners with steep increases in audiometric thresholds between the first and second formant regions of the test stimuli.

One may hypothesize from these findings that the subjects in the LinEar preference group (more flat loss) were less susceptible to upward spread of masking, and thus did not benefit from syllabic compression in the LF channel, while the RangeEar preference and DynEar preference subjects were susceptible to upward spread of masking and therefore could benefit from the compression.

Thus, we may assume that compression in the LF channel is successful for the sloping group because of the lower gain provided at high input levels resulting in more release of masking in the HFs compared to linear gain. Furthermore, the fast regulation of gain by syllabic compression may be beneficial because it releases from forward masking in the LF channel as well as upward spread of masking. This suggests that we have to search for an optimal compromise between making sounds audible and avoiding masking due to too much LF gain.

*Evidence Supporting the Hypothesis that Compression in LF Gives Release of Masking in HF.* In Experiment 2 we had the opportunity to have an independent HF gain control. Using the logger we could follow the actual use of this control over time, and from the data we calculated the average HF gain used for each subject with DynEar preference.

The data were analysed across subjects and we found a strong correlation between the actually used gain in HF and the hearing thresholds in LF (and UCL in HF). The regression equation found suggests for example that a subject with a $HTL_{LF}$ of 50 dB HL used a 6.8 dB higher gain in HF than a subject with a $HTL_{LF}$ of 40 dB HL (assuming the $UCL_{HF}$ equal). Since the gain provided in LF was directly related to the hearing thresholds in LF, via the loudness model, we may assume that the gain used in HF was determined by the gain used in LF. Furthermore, we found no correlation between the gain actually used in HF and the hearing thresholds in HF. Of course the subjects would adjust the gain to make sounds audible, but what the results suggest is that the influence from the hearing thresholds in the LF (and thus gain through the loudness model) channel is more important than the influence from hearing thresholds in the HF channel. A likely explanation for this finding is upward spread of masking.

These data suggest that we cannot treat the LF and HF channels independently. This implies that the regulation of HF gain should be influenced by the actual sound levels in the LF channel, to avoid upward spread of masking by the amplified LF signal.

*Convenience?* In the above explanations about reasons for RangeEar, DynEar or LinEar preference we have ignored the possible explanation that some subjects simply appreciated the convenience of having a completely automatic hearing aid, while others appreciated the ability to manually control either the whole bandwidth or just the higher frequencies. With such reasoning, the differences in preference should be attributed to completely subjective and partly non-auditory considerations. However, the significant correlation between auditory dynamic range data and preference is an argument against convenience being of any major importance with regard to preference.

## Conclusions

The results with the experimental wearable instruments do not indicate one general good-for-all rationale, but rather different rationales resulting in preference and performance which depend on the hearing loss configuration. The results show that sound quality and possibly speech recognition in noise are important factors affecting preference, and they indicate that preference is a reliable measure of the success of an algorithm.

Furthermore, it is shown that the preference for one of the new algorithms can be predicted, with good reliability, based on auditory dynamic range measurements. LinEar preference subjects have a relatively flat hearing dynamic range frequency response, whereas DynEar and RangeEar preference subjects have a more sloping configuration. The different preferences of

DynEar and RangeEar seem to be determined by the amount of loss in the HFs, the DynEar preference subjects having the most severe losses. It is hypothesized that the different preferences are affected by different susceptibility to masking of HF sounds by amplified LF sounds, and that the success of most of the subjects using syllabic compression in the LF channel is due to the release of upward spread of masking.

## Acknowledgements

This work was supported by research grants from the Nordic Development Centre for Rehabilitation Engineering (NUH), the Swedish National Board for Technical Development (NUTEK), the Swedish Research Council for Engineering Sciences (TFR), Hörselskadades Riksförbund and Stiftelsen Tysta Skolan. The project is a cooperation between the Division of Technical Audiology, Linköping University, Sweden, and Oticon Research Unit, Eriksholm, Snekkersten, Denmark.

## References

Barfod J. Multichannel compression hearing aids: experiments and considerations on clinical applicability. Scand Audiol 1978; (Suppl. 6): 315–40.

Byrne D, Dillon H. The National Acoustic Laboratories' (NAL) new procedure for selecting the gain and frequency response of a hearing aid. Ear Hearing 1986; 7: 257–65.

Benson D, Clark TM, Johnson JS. Patient experiences with multiband full dynamic range compression. Ear Hearing 1992; 13: 320–30.

Bonding P, Elberling C. Loudness summation across frequency under masking and in sensorineural hearing loss. Audiology 1980; 19: 57–74.

Danaher EM, Osberger MJ, Pickett JM. Discrimination of formant frequency transitions in synthetic vowels. J Speech Hear Res 1973; 16: 439–51.

Dillon H. Compression? Yes, but for low or high frequencies, for low or high intensities, and with what response times? Ear Hearing 1996; 17: 287–307.

Gagné JP. Excess masking among listeners with a sensorineural hearing loss. J Acoust Soc Am 1988; 83: 2311–21.

Hagerman B. Clinical measurements of speech reception threshold in noise. Scand Audiol 1984; 13: 57–63.

Hagerman B, Kinnefors C. Efficient adaptive methods for measurements of speech reception thresholds in quiet and in noise. Scand Audiol 1995; 24: 71–7.

Hawkins DB, Naidoo SV. Comparison of sound quality and clarity with asymmetrical peak clipping and output limiting compression. J Am Acad Audiol 1993; 4: 221–8.

Hawkins DB, Walden BE, Montgomery AA, Prosek RA. Description and validation of an LDL procedure designed to select SSPL90. Ear Hearing 1987; 8: 162–9.

Lunner T, Hellgren J. A digital filterbank hearing aid: design, implementation and evaluation. Proceedings of the International Conference on Acoustics, Speech and Signal Processing (ICASSP 91), Toronto, Canada 1991; 5: 3661–4.

Lunner T, Hellgren J, Arlinger S, Elberling C. A digital filterbank hearing aid: predicting user preference and performance for two signal processing algorithms. Ear Hear 1997a; 18: 12–25.

Lunner T, Hellgren J, Arlinger S, Elberling C. A digital filterbank hearing aid: three Digital Signal Processing algorithms—user preference and performance. Ear Hear 1997b; 18: 373–87.

Mangold S, Leijon A. Programmable hearing aid with multichannel compression. Scand Audiol 1979; 8: 121–6.

Manly BFJ. Multivariate statistical methods: a primer. New York: Chapman & Hall, 1986.

McCandless GA, Lyregaard PE. Prescription of gain and output (POGO) in hearing aids. Hear Instr 1983; 34: 16–21.

Moore BCJ. Perceptual consequences of cochlear hearing loss and their implications for the design of hearing aids. Ear Hear 1996; 17: 133–60.

Moore BCJ, Johnson JS, Clark TM, Pluvinage V. Evaluation of a dual-channel full dynamic range compression system for people with sensorineural hearing loss. Ear Hear 1992; 13: 349–70.

Olsen J. From acoustic amplification to hearing loss compensation. Hear Instr 1992; 43: 12–8.

Ono H, Kanzaki J, Mizoi K. Clinical results of hearing aid with noise-level-controlled selective amplification. Audiology 1983; 22: 494–515.

Plomp R. The negative effect of amplitude compression in multi-channel hearing aids in the light of the modulation-transfer function. J Acoust Soc Am 1988; 83: 2322–7.

Plomp R. Noise, amplification, and compression: considerations of three main issues in hearing aid design. Ear Hear 1994; 15: 2–12.

Schwartz DM, Lyregaard PE, Lundh P. Hearing aid selection for severe-to-profound hearing loss (POGO II). Hear J 1988; 41: 13–7.

Steinberg, JC, Gardner, MB. The dependency of hearing impairment on sound intensity. J Acoust Soc Am 1937; 9: 11–23.

Summers V, Leek MR. Intraspeech spread of masking in normal-hearing and hearing impaired listeners. J Acoust Soc Am 1997; 101: 2866–76.

Trees DE, Turner CW. Spread of masking in normal subjects and in subjects with high frequency hearing loss. Audiology 1986; 25: 70–83.

Van Dijkhuizen JN, Festen JM, Plomp R. The effect of frequency-selective attenuation on the speech-reception threshold of sentences in conditions of low frequency noise. J Acoust Soc Am 1991; 90: 885–94.

Villchur E. Signal processing to improve speech intelligibility in perceptive deafness. J Acoust Soc Am 1973; 53: 1646–57.