

DIGITAL HEARING AIDS - THE WAY OF THE FUTURE

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First developments in digital processing of speech sound were done in 1960's in Bell Laboratories [29]. In recent years, the use of digital components in the design of hearing aids (HA's) is fast becoming a standard rather than an exception. Various HA's on the market, that utilize digital circuits, have proven to provide increased flexibility and efficiency in both fitting and hearing aid evaluation [1,2,25]. Although the availability of digital signal processing (DSP) techniques that are applied to the incoming signal do not vary much in the commercially available units, it is evident that research in DSP is the most promising area in future hearing aid development. Already new generation of HA's brings higher levels of satisfaction from the end user-hearing impaired [2,17,24,26].

Signal Processing can be viewed as any manipulation of a signal that alters its characteristics; whether extracting, enhancing or otherwise modifying said information [3,4,30]. These changes are conducted in order to help the hearing aid wearer to better discriminate between speech and noise; essentially then to increase signal to noise ratio (S/N). To achieve these results the consumer has available to him/her two basic approaches: automatic/adaptive digitally controlled analogue systems, and digitally programmable HA's. It should be noted that however complex a HA may seem, generally five approaches to signal processing are in use: variable gain, equalization, compression, limiting, and steady state noise reduction [8]. Following is a description of the major types of DSP on the market.

Automatic/Adaptive DSP circuits

This form of DSP; decreasing certain frequencies, while trying to leave the so called speech frequencies untouched, has been shown to increase speech intelligibility by 15% in the presence of low frequency noise [3], but so far gives no increased benefit to the HA wearer while in the presence of competing noise of similar broadband spectra, such as cafeteria noise [7].

The **Argosy Manhattan II Circuit** automatically alters its frequency response as a result of continuous sampling of sounds in the environment. As the noise SPL increases, high frequency gain decreases and low frequency output also decreases. Adjustments to the low frequency potentiometer provides up to 40 dB of gain reduction at 500 Hz [4].

The **Siemens 283** changes its overall frequency response to compensate for the loss in speech intelligibility caused by low frequency noise. The incoming signal is divided into two channels, the low frequency; up to 800 Hz which contains a compression circuit and the linear high frequency channel, which contains an adjustable high-pass filter from 800-1600 Hz [3].

From **Intellitech** the **Zeta Noise Blocker II**, a digital microchip integrated into a hearing aid circuit, samples incoming signals and analyzes the rate of frequency change, becoming active when the presence of noise is detected and applying digitally controlled attenuation by four analog filters. This circuit has been shown to be more effective in increasing S/N in the presence of high frequency competition [1,9].

Although the **K-AMP** is an amplifier, it does adaptively affect the linearity of a signal, offering approximately 25 dB maximum gain for sounds below 40 dB SPL, with gain gradually reduced to 0 dB as input level increase. The K-AMP will operate to 110 - 115 dB SPL input without distortion, and is noted as only amplifying quiet sounds. Loud transient sounds that represent a problem to many HA users are passed without amplification. These transients cause the **Telex Communication Adaptive Compression Circuit**®, included in the K-AMP, to quickly drop to 0 dB, with a recovery time almost as fast; resulting in little affect on the ongoing gain [5,6].

Digitally Programmable HA's

Digitally programmable HA's process sound in the analogue domain,

but are controlled by digital circuitry. They have the ability to be reprogrammed by an external unit, thus allowing quick comparisons of different settings in order to determine patient's preferences [1,10]. Paired-comparison techniques can be used with these types of HA's to improve the fitting process [22].

The **Audiotone System 2000** (from Dalberg "The Dolphin System™") was the first programmable unit available on the market (1988). Its programming features include, maximum output, gain, high frequency cut (25 dB at 5 kHz), low frequency cut (30 dB at 500 Hz), input compression and frequency dependent output limiting [1,11].

The **Maico/Bernafon PHOX** (programmable hearing operation system) allows the programmability of the following features: gain, high and low frequency cut, slope of the high and low frequency response, a high frequency emphasis filter (cuts gain below 1500 Hz), a 3 kHz peak to emulate the canal resonance, option X a patient-activated noise reduction system, which cuts high and low frequency gain an additional 18 dB per octave, and an **automatic gain control (AGC)** input compression circuit [1,11,12,13,25].

Siemens Triton 3000 is a three channel compression HA. Ability is provided to program the gain in each channel (up to 36 dB), the AGC in each channel as well as two crossover points (300-1400 Hz & 1-5 kHz) with a minimum bandwidth of 1 octave and maximum of 3 octaves [1,11,25].

Resound's Personal Hearing System (PHS) offers a programmable two channel compression unit, with a compression range of 3:1 - 1:1, and an ultrasonic remote control for reprogramming. Programmable crossover frequencies for the high and low frequency bands are between 400-4000 Hz, with gain variable from 0 - 40 dB for each band. Fitting information can be stored in memory cartridge, as well as transferred to a PC. ReSound also incorporates an input compression limiter to control maximum output [1,11,14,25].

3M Corporation offers the **Memory Mate**, a programmable two channel compression HA. Programmable functions include; overall gain, dispenser-adjustable crossover frequency between 500-4000 Hz, and an eight memory client selected RAM. The RAM stores different frequency responses, for different environments [1,11,25].

The **Widex Quattro** allows for programmability of gain, maximum power output, output compression on/off, low and high-cut filters, plus an additional low-cut filter, the inverse presbycusis adaption filter. The HA also contains four memory choices selectable from a small FM remote control, which also operates as a programming unit, with the insertion of a programming key [1,11,15,16,25].

The **Ensoniq Sound Selector HA** has a programmable thirteen band equalizer that is adjustable in 1 dB increments up to 40 dB in each band and up to 60 dB overall gain. The thirteen bands are divided as such: lowest two in 1 1/2 and 1 octave bands respectively, the next three are 1/2 octaves, and the following eight are 1/3 octave bands. This is noted to generate a smooth response virtually free of distortion caused by resonance peaks. Also standard is 2:1 frequency-independent input compression and a directional microphone [1,11,17,18,19,25].

The **Nicolet Phoenix**, the only fully digital HA, has been taken off the market. Based upon the fitting, different frequency response characteristics, and a noise reduction algorithm were programmed (by the manufacturer) into the HA. The user was then able to select one of three "modes", by selecting one of three buttons on the aid, each set up for different listening environments. The **Phoenix VP** (variable processor) has five frequency response curves programmed into the first button, five degrees of noise reduction in the second, and the third button set to reverse the effect of the first two. The **Phoenix Plus** offers an additional 15 dB of gain [1,24].

Other programmable hearing instruments are offered by:

Audioscience (2-channel), Beltone Electronics Corp. (1-channel), GN Danavox (3-channel), Philips Hearing Instruments (3-channel), Rexton Inc. (1-channel) and Starkey Laboratories Inc. (1-channel) [25].

Conclusions

● In the past decade digital HA research and development has resulted in a number of improvements in clarity and S/N ratio (in certain environments). Most of current ASP (automatic signal processing) devices are still limited however by the number of frequency bands; thus are generally only effective in the presence of low frequency competition. All but Ensoniq have limited their approach to a few distinct processing channels. We should note when considering masking by noise, it is well accepted that the frequency range of the human auditory system is divided into 24 discreet channels, or critical bands (for each ear) [20] and each cochlear neuron responds to a narrow range of frequency stimuli [21]. It would seem appropriate then to at least separate the frequency range into like divisions. Philips has recently developed the Digital Compact Cassette (DCC), which utilizes 32-band digital processing for perceptual coding of audio signals [27]. This technology could be very easily adopted to make 32-band HA's. However, DCC will not be accepted as a new recording format, since recordable optical discs will be released soon to the consumer market.

● We still lack sufficient understanding as to the nature of many hearing impairment problems and their relationship to one another [2,5,29]. Generally hearing problems are divided into: conductive, cochlear, eighth nerve and central nervous system disorders [2]. However the number of the distinctive disorders may reach easily into the hundreds [2], and each may require a specific signal processing algorithm [2,24]. Further research is still needed to explore the usefulness of compression systems [23]. Information theory can be applied to calculate inherent channel capacity for the ear [28]. On the basis of this theory analysis of a hearing impaired communication channel could be performed and the most appropriate information coding obtained.

● Digital Signal Processing workstations (for example NeXT computer) should be used to perform further psychoacoustic tests in order to learn more about the human auditory system. Also a Digital Master HA can be simulated on these types of computers and used to design and check different DSP strategies to be used in HA's.

● The complexity of processing which is needed to address many hearing disorders requires highly sophisticated signal processing. DSP offers substantial improvements over analog techniques [30] along with unmatched flexibility and precision to adopt the processing to individual requirements of each patient [2]. Also the paired-comparison judgment technique may be used more effectively with this technology for precise HA fitting [22].

● DSP should complement rather than substitute for signal processing which is performed in the auditory system (in other words it should be transparent when not needed). This will allow the best signal processor so far - the human brain - to extract information most efficiently [18].

● The following DSP techniques could be used in future HA's : arbitrary filtering and frequency shaping, arbitrary gain (as function of frequency and signal amplitude), frequency shifting, feedback control, noise reduction (various techniques), peak clipping or limiting etc [24,30]. Also multichannel, parallel processing can be done with DSP improving speed and sophistication of sound processing. "Smart HA's" with adaptive algorithms and performing logical operations can be build around DSP technology to further improve HA's capabilities.

In our opinion DSP is still an underexplored technology in the area of HA's, but this may change in the near future with anticipated benefits to the hearing impaired.

HA's however sophisticated never would be a panacea for hearing impairment. Hearing impairment reduces information channel capacity (from outside world to auditory system) and this can't be restored with a hearing aid. HA's can only help to better utilize the remaining information channel capacity.

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