

# New Digital Processor for Hearing Loss Compensation based on the Auditory System

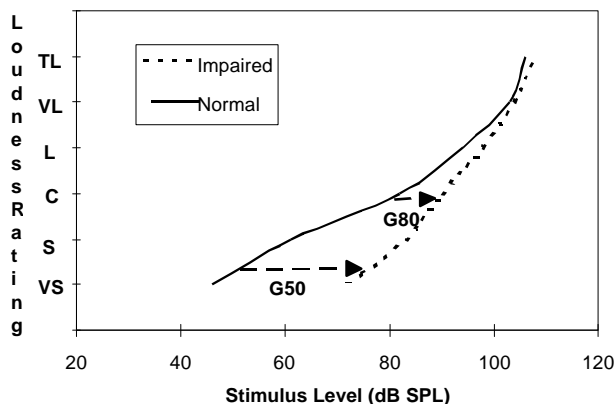
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## INTRODUCTION

ReSound Corporation introduced the first commercially available hearing device featuring multiband full dynamic range compression (MBFDRC) sound processing in 1989. Developed by AT&T Bell Labs and ReSound, the patented MBFDRC circuit was based on a design by Villchur to compensate for loudness recruitment<sup>1</sup>.

Recruitment causes the loudness level of a signal to increase at a steeper rate for persons with sensorineural hearing loss than for those with normal hearing (see Fig 1). Fast-acting syllabic compression, as implemented in all ReSound products to date, is an attempt to restore loudness perception to normal; the compressors operate fast enough to provide the appropriate gain to individual phonemes.



**Figure 1** Loudness scaling curves, demonstrating the steeper loudness growth for persons with sensorineural hearing loss than for those with normal hearing. See Allen et al. (1990) for details on the scaling technique<sup>5</sup>.

There is a physiological rationale for MBFDRC. In a healthy auditory system, the outer hair cells compress the response of the basilar membrane in order to "squeeze" the 120dB dynamic range of acoustic stimuli into the 60dB dynamic range of the inner hair cells. With outer hair cell damage, which is the cause of hearing loss in most people with mild-to-moderate impairment, this compressive nonlinearity is lost and the basilar membrane

responds in a more linear manner with reduced dynamic range.

By "pre-" compressing the signal in the hearing device, the compressive nonlinearity and dynamic range of the overall transduction system is restored. The rationale and consequences of MBFDRC are discussed in detail in Pluinage<sup>2</sup>. Comprehensive reviews of all compression schemes can be found in Hickson and Dillon<sup>3,4</sup>.

This article describes the reasoning behind Digital Cochlea Dynamics™ processing, ReSound's newest multiband compression system for sensorineural hearing loss compensation, made possible by the latest digital signal processing technology. This new approach attempts to: 1) more accurately mimic normal basilar membrane processing; 2) increase the precision with which the frequency response of the compressive gain fits the hearing loss; and 3) create a more robust multiband compression system than has been previously available.

## PHYSIOLOGICAL BASIS FOR DESIGN

The Digital Cochlea Dynamics processor, like its predecessor the MBFDRC circuit, seeks to replicate the lost physiological function of healthy outer hair cells. To do this, it uses a low compression threshold, compression ratios in the range of 1:1 to 3:1, and fast syllabic-rate attack and release times. In addition to addressing recruitment, this new approach includes a heavily overlapping filter bank, which also has a physiological basis for its design.

The human auditory system is commonly thought of as contiguous, non-overlapping critical-band filters. In reality, however, the auditory system has approximately 3500 filters that span the audible spectrum, corresponding to the number of inner hair cells in each ear<sup>6</sup>. These filters are *heavily overlapping*, such that two filters might have center frequencies that differ by only 1 Hz, yet their bandwidths could be over 100 Hz wide. This overlap provides fine frequency resolution while maintaining relatively wide auditory filters. For this reason, Digital Cochlea Dynamics processing also has heavily overlapping filters. These provide fine control of the gain's frequency response while better preserving the spectral contrast of the compressed sound.

## FLEXIBLE FREQUENCY SHAPING

One benefit of having many bands in a compression system is the flexibility they provide for setting gain and compression across frequencies. Digital Cochlea Dynamics processing can produce non-monotonic insertion gains as well as prescriptions for any number of arbitrary response shapes. This flexibility in programming allows the hearing health care professional to fit the new ReSound digital hearing devices to a wide variety of hearing losses, including flat, ski-slope, reverse-slope, cookie-bite, and even reverse cookie-bite configurations.

In multiband compression systems that have a small number of bands, the characteristics of the individual filters are important because of their effect on (and limited capacity for) frequency shaping. The hearing professional needs to understand these characteristics to fit different hearing losses appropriately. In Digital Cochlea Dynamics processing, the large number of filters and the overlapping nature of the bands means the response at any given frequency to be determined not by one band but by the cumulative processing of all bands that overlap at that frequency. The hearing professional can simply specify the gain requirements for 50dB SPL and 80dB SPL signal levels at the standardized, octave-band center frequencies. The fitting system then determines the programming parameters of the individual bands to produce the desired effect at these specified frequencies.

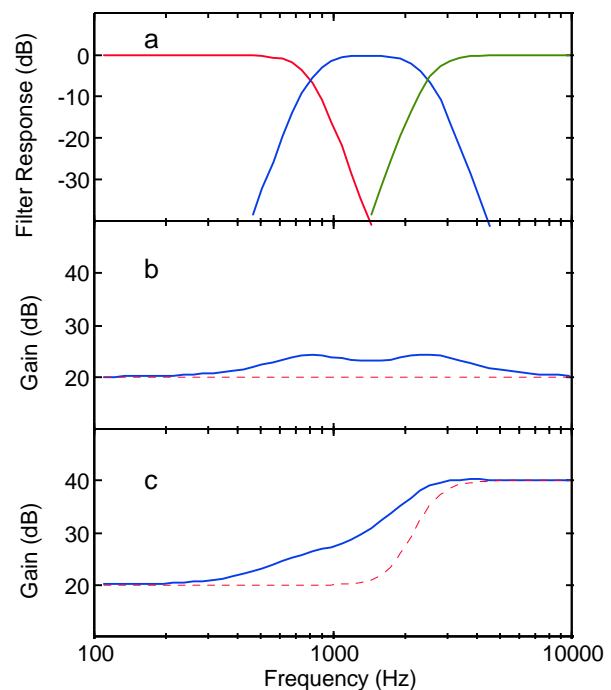
In the vast majority of cases, this approach enables the practitioner to focus on the overall effect of the processing for the patient instead of on the actual filter characteristics, such as center frequency, filter slopes, and crossover frequencies. As sound processing for hearing loss compensation becomes increasingly sophisticated (thanks in part to digital signal processing), the characteristics of the physical structures underlying the compressors are of less concern to the dispenser than is the overall effect for the patient.

## ROBUST MULTIBAND COMPRESSION PROCESSING

Having several compressor bands provides more control over the frequency shaping of the gain and compression. It also preserves the frequency-specific effect of the compression, where the gain at any frequency is determined by the signal power only in that frequency region. With multiple bands, for example, the gain applied to the second formant of a vowel is *not* determined by the higher-level first formant. However, the nonlinear nature of compression can cause irregularities in the transition regions between bands<sup>7</sup>. Hearing devices based on the 14-band Digital Cochlea Dynamics approach

will be the first multiband compression systems to address this issue.

Figure 2a shows three bands of a hypothetical compression system that divides the spectrum into low-, mid- and high-frequency regions. The filters shown here have been designed such that the response is flat across frequencies when equal gain is applied to each band. For example, if the compressors in each band of Figure 2a were programmed for 20dB gain, then the resulting gain function after summing the outputs of each filter together would be 20dB at all frequencies.



**Figure 2** *a)* is a typical 3-band compression system. *2b)* shows the system response to pure-tone (solid line) and broad-band (dotted line) stimuli when programmed for 20dB gain in all bands. *2c)* shows response to pure-tone (solid line) and broad-band (dotted line) stimuli when programmed for high-frequency emphasis. Note that amplification at the transition frequencies is significantly less for a broad-band signal.

Now consider the case where each compressor in Figure 2a is programmed with 3:1 compression and a gain of 20dB for 65dB SPL input. If the response of the hearing device is measured with a broadband stimulus so that the level of the signal in each band is 65dB SPL, then the gain applied at all frequencies is 20dB, as expected and as shown by the dashed line in Figure 2b. If the response of the same hearing device is measured with a 65dB SPL pure-tone frequency sweep, however, the hearing device

behaves very differently. The gain function measured using a pure tone is shown by the solid line in Figure 2b. Notice how the gain is higher than expected: up to 4dB at the frequencies where the bands cross over. This "over amplification" will always occur for narrowband signals that fall in the cross-over region between two compressive bands.

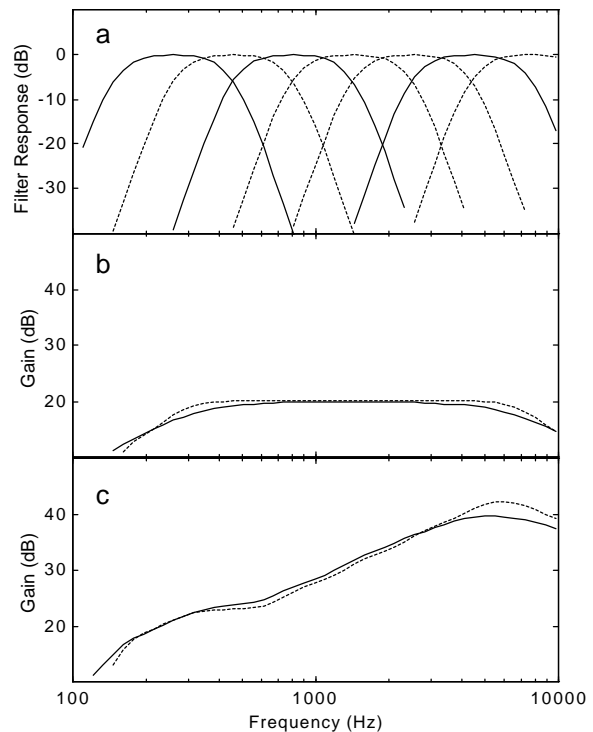
This effect is exacerbated when different bands are programmed for different gains. This is demonstrated in Figure 2c, where each band has been programmed for 3:1 compression, but the highest band has 20dB more gain than the other bands. The gain applied to the pure tone is up to 10dB higher than the gain applied to the broad-band signal. The filters can be designed to minimize the "over amplification" effect by adjusting the slopes of the filter roll-off and the phase responses. However, the gain applied to a pure tone will still be greater than the gain for a broad-band stimulus. This effect also becomes more significant as the number of bands in the compressor increases, since the number of cross-over regions increases. Digital Cochlea Dynamics processing has been designed to eliminate the "over amplification" effect.

Lindemann demonstrated theoretically that a filter bank for compression is similar to sampling in the frequency domain and that the "over amplification" can be treated as an aliasing effect<sup>8</sup>. Without going into the mathematical details, one solution for eliminating "over amplification" is to simply have more *overlapping* filter bands. This is in contrast to the usual "side-by-side" or non-overlapping arrangement of a traditional bank of filters. In most multiband processing, only the "skirts" of adjacent filters overlap (see Figure 2a). As discussed previously, the overlapping design is more consistent with the natural filter structure of the human auditory system.

Figure 3a shows a simplification of the basic overlapping filter bank design in the ReSound Digital Cochlea Dynamics processor. With this design, at least two filters have a significant effect on the compression and gain at any given frequency. Figures 3b and 3c show the response of this system to broad-band and pure-tone stimuli, respectively. Note that the responses to the broad-band and narrow-band stimuli are similar in shape and that only minor differences occur at the very ends of the frequency range where just one filter processes the stimuli. The increased overlap of the filters has eliminated the "over amplification" of the gain function in the transition regions.

In this way, the gain response of the Digital Cochlea Dynamics hearing device is independent of the acoustic stimulus and, therefore, it more accurately reflects the gain prescribed by the hearing professional. Since the device does not overamplify narrow-band stimuli,

standardized characterizations performed with tonal stimuli (e.g., 2cc coupler measurements), can accurately calculate the real ear insertion gain using coupler-to-real-ear transforms. This is not the case for compression systems with non-overlapping filters. For the situation in Figure 2c, a coupler measurement made with a tone sweep would incorrectly result in an overestimate of the insertion gain.



**Figure 3** a) Compression system with overlapping filter bank. b) System response to pure-tone (solid line) and broad-band (dotted line) stimuli when programmed for 20dB gain in all bands. c) System response to pure-tone (solid line) and broad-band (dotted line) stimuli when programmed for high-frequency emphasis. Note that the response is as predicted and independent of the stimulus bandwidth.

## CONCLUSION

The Digital Cochlea Dynamics processor in the new ReSound digital hearing system provides some significant benefits including:

- More accurate reflection of the physiological processing of a healthy auditory system;
- Greater fitting flexibility for a wide range of audiograms;

- Elimination of "over amplification" effects in the transition regions;

The new ReSound digital hearing system also features software algorithms for digital directional processing using multiple microphones, feedback cancellation and noise reduction, all of which are implemented in a fast Fourier transform (FFT) structure. These other algorithms will be discussed in a follow-up article.

## REFERENCES

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