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A Novel Design for Restoration of Suppression with Noise Reduction Capability in Hearing Impaired Listeners

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Abstract—The cochlea is the primary sensory organ for hearing. Its three major signal-processing functions are frequency analysis, dynamic-range compression (DRC) and amplification. The cochlea implements these functions in a concurrent manner that does not allow completely separate characterization of each function. Common forms of hearing loss are manifestations of mutual impairment of these major signal-processing functions. Gamma tone filters are widely used in computational auditory models for modelling the peripheral filtering in the cochlea. Filters of this type are then combined to form a computational auditory filter bank. The filter bank was used as the peripheral filtering stage of the computational auditory model. Suppression of the filtered output plays an important role in the coding of speech and other complex stimuli and it has been suggested to result in the enhancement of spectral contrast of complex sounds, such as vowels. The individual outputs of the suppression stage are then combined to obtain an output signal with suppressive effects. The restoration of both normal suppression and normal loudness has the potential to improve hearing-aid performance and user satisfaction. Also single channel speech enhancement at the output of the auditory motivated Gamma tone filter bank is performed for benefits in terms of listener preference and Speech Intelligibility that improves speech perception in presence of background noise.

Keywords- Cochlea,Distortion product optoacoustic emission (DPOAE),Compression,Gammatone,Hearing aids,Spectral En- hancement,Noise Reduction.

I. INTRODUCTION

The peripheral auditory system transduces airborne sounds into a robust and meaningful neural code. Many of the mechanisms involved in the sound-transduction process perform nonlinear signal transformations, and some show a strong dependence on the frequency of the stimuli being processed. The combination of strong frequency-dependence and nonlinearity can lead to complex interactions in the systems response to many sounds. One such interaction is demonstrated in the phenomenon of two tone suppression where the responses evoked by one tone can be reduced quite substantially when a second tone is presented at the same time. Nonlinear interactions of this type are of potential importance in many areas of hearing research in the peripheral encoding of speech stimuli and in determining the masking effects of background noise.

People can no longer hear softer sounds when the hair cells in the inner-ear responsible for amplifying them are damaged. But they continue to hear louder sounds in almost the same way as people with normal hearing. This phenomenon is known as reduced dynamic range. It means that different levels of sound must be amplified by different amounts in order to compensate for lost soft sounds without making louder sounds. Dynamic-range compression is used in hearing aids and cochlear implants to compensate for the limited dynamic range of the impaired auditory system. An important by-product of Dynamic Range Compression (DRC)is suppression, which contributes to psychophysical simultaneous masking.

Two-tone suppression is a nonlinear property of healthy cochleae in which the response (e.g., basilar-membrane displacement and/or neural-firing rate) to a particular fre- quency is reduced by the simultaneous presence of a second tone at a different frequency [1 -3].Because these invasive measurements cannot be made in humans, suppression must be estimated by other physiological or psychophysical pro- cedures. Distortion product optoacoustic emission (DPOAE) suppression [16] is one of these procedures and can be used to provide a description of the specific suppressive effect of one frequency on another frequency.

While the active response to a pure tone provides frequency selectivity, exquisite sensitivity, and wide dynamic range, its intrinsic nonlinearity causes tones of different frequency to interfere with one another in the cochlea. Multiband DRC hearing aids attempt to restore DRC but currently do not attempt to restore normal suppression. DRC alone (i.e., without suppression) may reduce spectral contrasts by reducing gain for spectral peaks while providing greater gain for spectral troughs. This paper describes a hearing-aid signal-processing strategy that aims to restore normal cochlear two-tone suppression, with the expectation that this would improve spectral contrasts for signals such as speech. The implementation of suppression in this strategy was inspired by measurements of DPOAE suppression tun-ing curves (STC).The processes of DRC, amplification, and suppression are not implemented separately in this strategy, but are unified into a single operation.

The prescription of amplification for the method is based on measurements of categorical Loudness Scaling (CLS) [25] for tones, and is intended to restore normal growth of loudness for any type of signal. The strategy is computationally efficient and could be implemented with current hearing aid technology to restore both suppression and loudness growth. Categorical loudness scaling deter- mines the loudness in the whole auditory dynamic range in terms of categories like inaudible, very soft, soft, medium etc. as a function of the stimulus level. This makes cat- egorical loudness scaling interesting for the diagnosis of recruitment (pathological reduction of the auditory dynamic range) and for the fitting of hearing aids with dynamic compression. The data, based on CLS, provide the basis of our amplification prescription strategy. The proposed strategy does not use audiometric thresholds, a model of loudness, or a model of speech intelligibility, but uses instead actual CLS data from each HI individual.

It is well known that background noise reduces the intelligibility [12] of speech and that the greater the level of background noise the greater the reduction in intelligibility. We are able to understand speech in a moderately noisy environment because speech is a highly redundant signal and thus even if part of the speech signal is masked by noise, other parts of the speech signal will convey suffi- cient information to make the speech intelligible, [7] or atleast sufficiently intelligible to allow for effective speech communication. There is less redundancy in the speech signal for a person with hearing loss since part of the speech is either not audible or is severely distorted because of the hearing loss. Background noise that masks even a small portion of the remaining, impoverished speech signal will degrade intelligibility significantly because there is less redundancy available to compensate for the masking effects of the noise. As a consequence, people with hearing loss have much greater difficulty than normally hearing people in understanding speech in noise.

Hearing aids allow for some degree of signal processing to reduce the effects of noise. The recent development of dig- ital hearing aids opens up substantial new possibilities with respect to the use of advanced signal processing techniques for noise reduction .Because of the particularly damaging effects of background noise on speech intelligibility for people with hearing loss (i.e., hearing-aid users) this problem is of critical importance.

In this work, an algorithm for single channel speech enhancement at the output of the auditorily motivated Gam- matone filter bank is established. The employed Wiener filter based single channel

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speech enhancement algorithm requires an estimate of the noise power spectral density. This noise power spectral density can for instance be estimated based on the a posteriori speech presence probability (SPP). In this paper, the statistical parameters of the speech presence probability estimator to the different temporal correlation at the output of individual Gammatone channels are optimized.
This paper is organized as follows (after Section I). Section II describes the Suppression tuning curves (STC) measurements. Section III describes Signal Processing strat- egy and spectral contrast enhancement. Finally Section IV and Section V describes the Experimental Results and Con- clusion.

II. SUPPRESSION TUNING CURVES MEASUREMENT

The idea behind psychophysical tuning curves stems from the measurement of physiological tuning curves where the level of a tone required to induce a fixed level of activity is measured as a function of frequency. In the behavioural counterpart, a low-level tone is used to excite a narrow region of the Basiliar membrane [9]. Then, at various frequencies around the signal, the level of a tonal or narrowband masker is adjusted to just mask the signal. Here, it is again assumed that masking occurs when a fix amount of excitation by the masker is present at the location of the tone. As the masker is farther off in frequency from the tone, less excitation is present, and correspondingly the masker levels will be higher at threshold. A tonal masker will produce beating near the signal frequency, providing the listener with an amplitude-modulation cue, which will result in higher thresholds and an unwanted sharpening of the tip of the curve that does not reflect frequency selectivity. This effect can be mitigated by using a narrow-band masker instead of a tone. Additionally, nonlinear interactions, such as suppression, may occur between the masker and the tone when presented simultaneously. Neither of these effects are present in the forward masking case, which causes important differences in the two measurements.
In the DPOAE suppression experiments, [5 - 6] DPOAEs were elicited in normal hearing (NH) human subjects by a pair of primary tones f_1 and f_2, ($f_2/f_1 \approx 1.2$), whose levels were held constant while a third, suppressor tone f_3 was presented. The suppressive effect of f_3 was defined as the amount by which its presence reduced the DPOAE level in response to the primary-tone pair. By vary- ing both the frequency and the level of f_3 ,information about the influence of the frequency relation between suppressor tone and primary tone (mainly f_2) on the amount of suppression was obtained. The DPOAE measurements that are used in the design of the model for human cochlear suppression include eight f_2 frequencies (0.5, 1, 1.4, 2, 2.8, 4, 5.6, and 8 kHz). For each f_2 frequency, up to 11 f_3 frequencies surrounding f_2 were used. There are many studies of DPOAE STCs, all of which are in general agreement. However, Gorga et al. provided data for the widest range of frequencies and levels in a large sample of humans with normal hearing. Thus, those data are used as the basis for the signal-processing strategy implemented here.
The suppression due to a single tone of the growth of its own cochlear response is what causes its response growth to appear compressive. The relative contribution of a suppressor tone at f_3 to the total compression of a tone at f_2 may be obtained as

$$L_2 = c_1 + c_2 L_3 \qquad (1)$$

Where c_1 & c_2 are regression coefficients, L_2 is the primary tone level & L_3 is suppressor tone level which is set as 40 dB SPL.

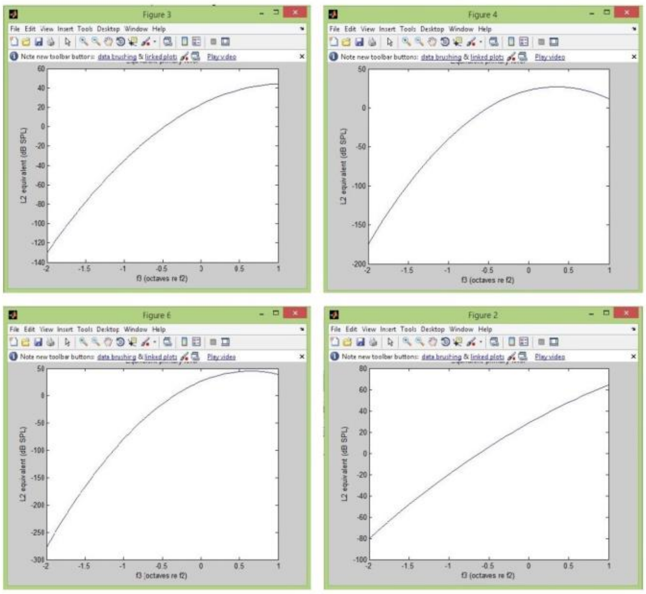

Figure 1. Curve fits of L_2 ***for different*** f_2

The coefficient c_2 describes the compression of f_2 (relative to compression at f_3) and is plotted in figure below. Note that L_2 and c_2 both have nearly linear dependence on frequency f_3 (when expressed in octaves relative to f_2) below $f_3 \approx f_2$ and that $c_2 \approx 1$ when $f_3 \approx f_2$

.We use these trends to generalize the dependences of $L2$ and c_2 on f_2 and f_3 , and obtain extrapolated c_2 and c_1 [obtained from extrapolated L_2 and c_2 that we use in the signal-processing strategy to determine frequency dependent gains. Extrapolation of c_2 and c_1 allows application of our signal-processing strategy at any frequency of interest, not just at the frequencies that were used in the DPOAE-STC measurements.

The extrapolation is a two-step polynomial-regression procedure that allows for the extension of the representa- tion of c_2 and c_1 from the available (data) frequencies to the desired (model) frequencies. First, separate polynomial regressions were performed to describe the f_3 dependence for both of the coefficients c_2 and L_2 at each of the eight f_2 frequencies.

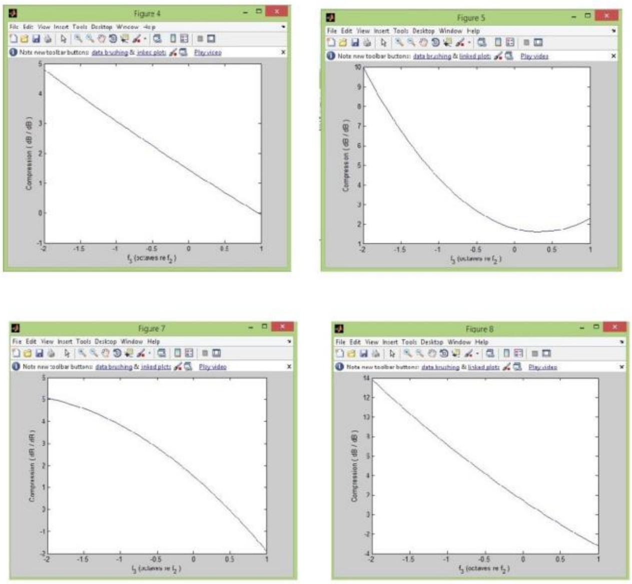

Figure 2. Curve fits of c_2 for different f_2

A second set of polynomial regressions were performed to describe the f_2 dependence of the coefficients of the 16 initial polynomials (8 for c_2 and 8 for L_2). The result of this two-step regression was a set of two polynomials that allowed calculation of values for c_1 and c_2 for any desired pair of frequencies f_2 and f_3

III. METHODOLOGY

The signal-processing simulation of human cochlear suppression along with speech in noise intelligibility consists of four main stages (as shown in Fig. 3) 1) Analysis 2) Noise Reduction 3) Suppression 4) Synthesis

A. Analysis

As applied in an auditory perception model , 4th-order linear Gamma tone filters can be used to mimic human auditory filters. The Gamma tone filter bank (GFB) was realized using cascaded first-order complex valued band- pass filters ,resulting in computationally efficient infinite impulse response band pass filters. The filter bank was used as the peripheral filtering stage of the computational auditory model. Application of the GFB for speech processing devices like hearing aids is desirable in order to allow for the use of more auditory based signal processing.

The outputs of the gamma tone filter bank [10] are complex band pass-filtered time-domain components of the input signal where the real part represents the band-limited gamma tone filter output and the imaginary part approximates its Hilbert transform. Thus, complex gamma tone filters produce an analytic representation of the signal, which facilitates accurate calculation of the instantaneous time-domain

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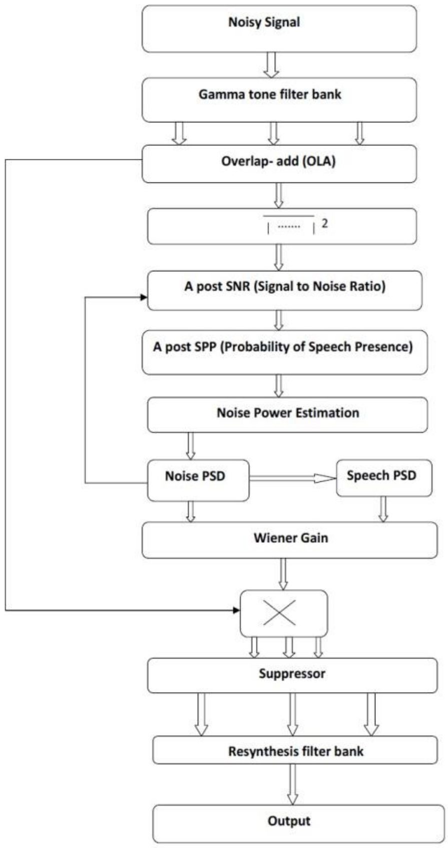


Figure 3. Block Diagram of the Proposed System

level. An advantage of the gamma tone filter bank over other frequency-analysis methods (e.g., Fourier transform, contin- uous wavelet transform) is that it allows frequency resolution to be specified as desired at both low and high frequencies. Also of interest is the fact that gamma tone filters are often used in psychophysical auditory models because of their similarity to physiological measures of basilar membrane vibrations. For the derivation of the bandwidth of an auditory
filter as a function of its center frequency, the concept of the equivalent rectangular bandwidth (ERB) of the auditory filters in the cochlea is used. The equivalent rectangular bandwidth or ERB is a measure used in psychoacoustics, which gives an approximation to the bandwidths of the filters in human hearing, using the unrealistic but convenient simplification of modelling the filters as rectangular band pass filters The ERB shows the relationship between the auditory filter, frequency, and the critical bandwidth. An ERB passes the same amount of energy as the auditory filter it corresponds to and shows how it changes with input frequency.

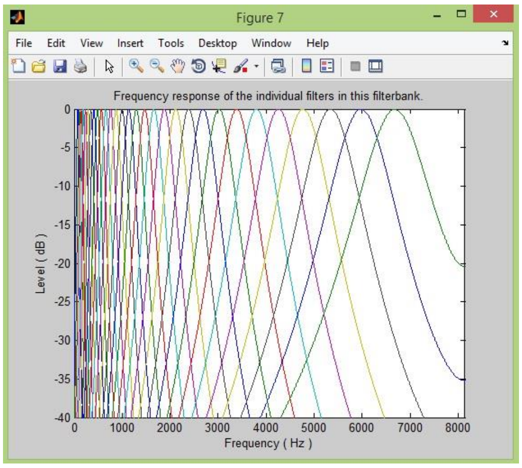


Figure 4. Magnitude frequency response of the Gammatone filterbank

The critical band is the band of audio frequencies within which a second tone will interfere with the perception of a first tone by auditory masking where f_C is the center frequency, l is the min bandwidth for lower frequency and q is the filter quality at large frequency.

B. Noise Reduction

After GFB analysis, Hann-windows w(n) with a constant duration of 32 ms, where successive segments overlap by 50%, were used in an overlap-add (OLA) manner. In this way, all estimates in the subsequent stages are updated each 16 ms. The resulting Wiener Gain from the noise reduction system is applied to each segment. In each GFB channel k and windowed segment l, a short-term period gram-like energy estimate of the (noisy) input signal Y is calculated.

C. Suppression

The suppression stage [4] of the model determines gains that are to be applied to each frequency band and that include suppressive effects. The gain applied to a particular frequency band is time-varying and is based on the instantaneous level of every filter bank output in a manner based on measurements of DPOAE STCs (Suppression tuning curves).

However, unlike in DPOAE suppression measurements where the suppressive effect of a suppressor frequency f3 on the DPOAE level in response to two primary tones (f1 and f2) was represented, the model represents the influence of a suppressor frequency fs which is equivalent to f3 on a probe frequency f3 which is equivalent to f2.Further,the notations fs and fp in the description of the model. The total suppressive

influence combines the suppressive effect of all frequency components into a single, equivalent level that would cause the same reduction in gain (due to compression) if it was the level of a single tone. This suppressive level is important in the design of the model. The typical decrement function to be reconstructed is obtained by subtracting the control condition which is the suppressive level for a one component stimulus Ls1 , from the suppressed condition which is the suppressive level for a two-component stimulus Ls2 , S1 and S2 are the suppressive intensities.

D. Synthesis

In the synthesis stage, the individual outputs of the suppression stage are combined to obtain an output signal with suppressive effects. The combination of the individual

frequency bands is designed to produce nearly perfect reconstruction when the suppressor applies zero gain to all channels. In this case, the output signal is nearly identical

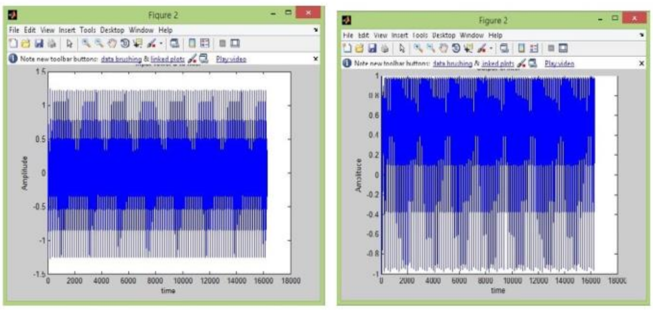
to the input signal, except for a delay that is equal to the filter bank target delay.

IV. RESULTS AND DISCUSSION

The performance of current MATLAB implementation of the suppression hearing-aid (SHA) signal processing, especially with regard to its ability to reproduce two-tone suppression, the suppressive-gain parameters were set to the following values: $L_{cs} = 0$, $L_{ce} = 100$, $L_{max} = 115$ dB SPL, $G_{cs} = 60$, and $G_{ce} = 0$ dB. In the simulations to follow, these settings were selected for application of SHA processing to a flat hearing loss of 60 dB.

The results after application of suppressive gain for vowels are illustrated in figures below. The results with vowel analysis shows that the suppression of output shows an increase in gain compared to the vowel input levels which is very desirable in hearing prosthetics. This hearing-aid signal-processing strategy performs two- tone suppression by considering the instantaneous output of all frequency channels when calculating the gain for a particular channel. This cross-channel influence in the calculation of gain is based on DPOAE-STC measurements and is applied instantaneously.

Vowel 'a' as input and processed output



Vowel 'i' as input and processed output

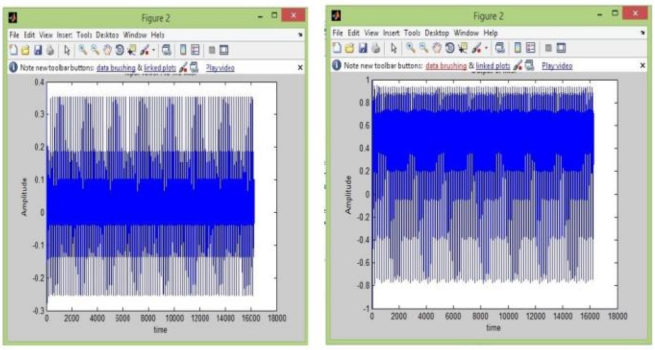


Figure 6. Suppressive gain demonstration for vowel inputs

This strategy has greater ecological validity compare which was also intended to restore two-tone suppression, because the design is based on DPOAE-STC data from NH subjects. However speech clarity or the proportion of a speaker's output that a listener can readily understand s important in perception of speech for hearing impaired listeners.

The working hypothesis is that loss of suppression is a significant contributor to abnormal loudness summation in HI ears. Therefore, integration of suppression and nonlinear gain based on loudness of single tones has the potential to compensate for loudness summation. The loudness data used for prescribing gain define the level of a single tone that will restore normal loudness in HI individuals.

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The suppression describes how the level of one tone affects the level of another tone at a different frequency.

We expect that this combined effect will generalize to loudness restoration for broadband stimuli, thus compensating for loudness summation. In this work, we present an algorithm for single channel speech enhancement at the output of the auditory motivated Gamma tone filter bank. The employed Wiener filter based single channel speech enhancement algorithm requires an estimate of the noise power spectral density. This noise power spectral density can for instance be estimated based on the a posteriori speech presence probability.

Enhancement of spectral contrasts [4] potentially could improve speech perception in the presence of background noise. A spectral-contrast measure was defined as the aver- age of the three formant peak minus the average of the two intermediate minima, in order to quantify the spectral contrast.SCE measures for the synthetic vowels a,e,i,o,u are presented in Table I when the input level was 70 dB SPL

Vowel	Spectral Contrast of Input (Unprocessed)	Spectral Contrast of Output (Unprocessed)
a	18.9376 dB	17.1121 dB
e	10.4083 dB	17.6106 dB
i	13.9091 dB	17.1298 dB
o	10.8119 dB	17.7613 dB
u	14.8486 dB	16.1644 dB

Figure 7. Spectral Contrast Enhancement for five synthetic vowels

V. CONCLUSION

The proposed hearing-aid signal-processing strategy unifies compression with cross-channel interaction in its calculation of level dependent gain. In this combined model, gain at each frequency is dependent to varying degrees on the instantaneous level of frequency components across the entire audible range of frequencies, in a manner that realizes cochlear-like two-tone suppression. . Although not specifically an element of the design, the presence of suppression apparently results in the preservation of local spectral contrasts, which may be useful for speech perception in background noise [11] . The proposed strategy is computationally efficient enough for real-time implementation with current hearing-aid technologies. Benefits in terms of listener preference and speech intelligibility are obtained as results which is important in hearing aid applications.

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