MODULATION SPECTRUM AS AN ADDITIONAL QUANTIFIER OF HEARING AID PROCESSING

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INTRODUCTION

While an acoustic signal is quantified by its three dimensions - frequency, intensity and time, much of the attention in hearing aid dispensing has been invested in the optimization of the output in the frequency and intensity dimensions. This is evidenced by the fact that the standards issued by the ANSI (S-3.22, ANSI, 2003) on hearing aids have stringent requirements on measuring/reporting the frequency-intensity output of the hearing aids. The only measurement of temporal processing is the measurement of attack and release time using a sinusoidal signal. The discussion of various advanced hearing aid features such as wide dynamic range compression (WDRC), digital noise reduction (DNR), and directional microphone has also centered around the goal of improved audibility or enhanced signal-to-noise ratio (SNR). The deliberate alteration of the signal to optimize audibility and/or SNR can lead to inadvertent alteration of the temporal characteristics of the input signal. The effect of some processing may enhance the temporal envelope, while others may result in suboptimal listening conditions, particularly to listeners with greater degrees of hearing loss or reduced working memory capacity (Souza et al 2015). Thus, the goal in hearing aid fitting should not just be optimizing audibility and/or SNR, but to achieve such objectives with minimal distortion (or even enhancement) of the temporal information also. In the absence of a formalized tool to quantify temporal envelope changes, we propose the inclusion of a measurement of modulation index (MI) derived from modulation spectrum of the input as a means to study the changes in modulation characteristics as a consequence of the signal processing in hearing aids.

SPEECH MODULATION

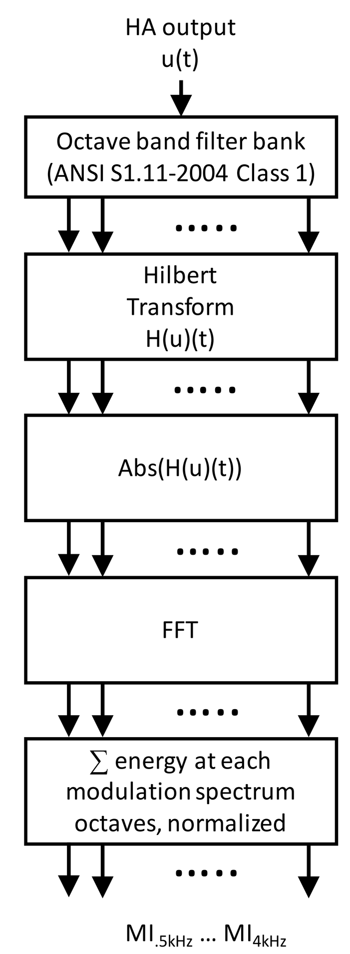
Amplitude envelope cues play an important role in speech intelligibility. Such cues could be modified by signal processing within the hearing aids. Envelopes can be characterized by temporal modulation spectrum, which displays the magnitude of the modulation (or modulation index) as a function of the modulation frequencies. The temporal modulation frequency describes the rate at which the signal envelope changes, whereas the magnitude or modulation depth describes the amount of changes in the amplitude envelope. Modulation rates to which listeners are most sensitive vary from 0.1 Hz to about 40 Hz with a peak sensitivity around 2 - 6 Hz, which is also close to the syllable rate of natural speech. Modulation rates between 2 and 50 Hz convey information on consonant manner, voicing, and vowel identity as well as tempo and rhythm.

Temporal envelope may also play a role in speech recognition in adverse situations. The correlation of amplitude envelopes between nearby channels for a single speaker may help listener to bind the different channels into a single perceptually coherent stream, which may help listeners to segregate target speaker from the background in the presence of competing speech.

Reliance on amplitude envelope cues may be increased for listeners with moderate to severe cochlear hearing loss, who have limited ability to utilize the temporal fine structure information of speech and may have broader auditory filters. Another group of listeners for whom the importance of temporal envelope may be heightened are those with a reduced working memory capacity. Maintaining the fidelity of the signal envelope may be especially important for those with a limited working memory capacity.

We propose using the modulation spectrum to characterize the temporal envelope of hearing aid output. We will first describe the method to calculate the modulation spectrum across octave bands. We will then report data from the measurements that evaluated the effects of individual hearing aid features on modulation index. Finally we will show how modulation index may be used to compare effects of hearing aid processing on temporal envelope between commercial hearing aids.

METHODS



Derivation of modulation spectrum across octave frequenices (MSaOF)

Output signals were limited to the one-octave bands using a filter bank with center frequencies at .5, 1, 2, and 4 kHz (ANSI S1.11-2004 Class 1 filterbank).

The filter bank outputs were analyzed using Hilbert transform to create an "analytic signal", which the magnitude represents the amplitude envelope of the signal.

Envelope signals were analyzed using Fast-Fourier Transform (FFT). The energy in each FFT bin were summed with the energy in the adjacent bins so that the summed energy was obtained for one octave bandwidths.

The energy values were divided by the energy at 0 Hz to provide a normalized "modulation index" value.

Modulation data at each octave was displayed at 4 Hz modulation frequency across the octave bands (see Figure 1).

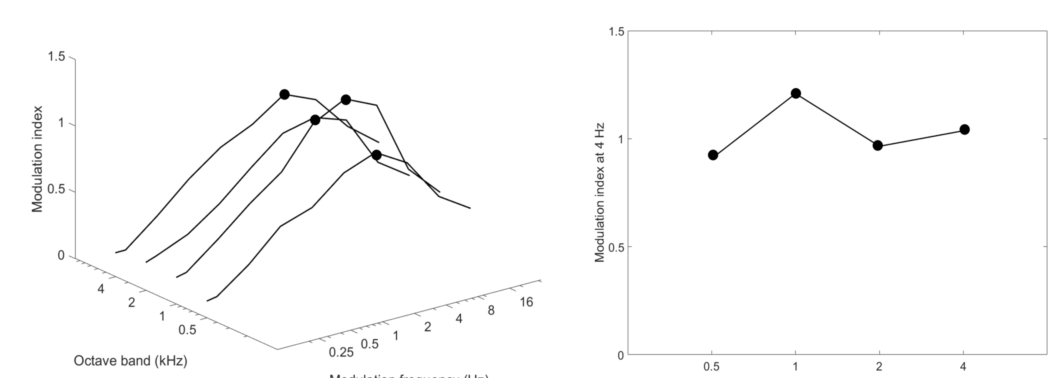


Figure 1: Modulation spectrum data (left) will be represented as the peak index occuring at 4 Hz modulation frequency across octave frequencies (right).

Hearing aids: Two "state-of-the-art" commercially available receiver-in-canal (RIC) type hearing aids (HA1 and HA2) from two manufacturers. Among the advanced features both devices included multi channel compression (15 and 16 channels), digital noise reduction, adaptive directional microphone, transient noise reduction, inter-ear data exchange, and feedback management system. HA1 also included an environmental classifier. The effects of individual features was assessed using HA1.

Hearing aid settings: HAs were programmed to flat 40 dB HL (isolated features test) and 30, 40, 50, and 60 dB HL (.5, 1, 2, and 4 kHz) (all features test) using manufacturers' fitting formulae, and default settings from the fitting software.

Stimuli: Speech (ISTS) at 60, 70, and 80 dB SPL from 0° in quiet or in the presence of continuous noise (ICRA-noise) from 0° or 180° at 5, 10 and 15 dB SNR.

Test setup: Electroacoustic measurements with KEMAR using fully occuding earmolds.

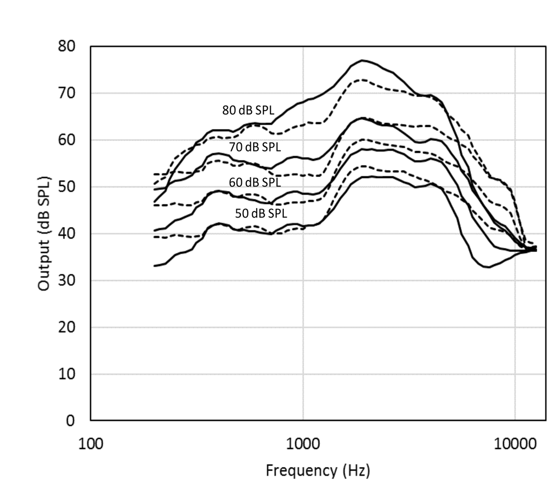


Figure 2: LTASS at the output of the two commercial hearing instruments included in the current study at 1/3-octave bands for speech input presented at 50, 60, 70, and 80 dB SPL from 0°.

FEATURES IN ISOLATION (HA1)

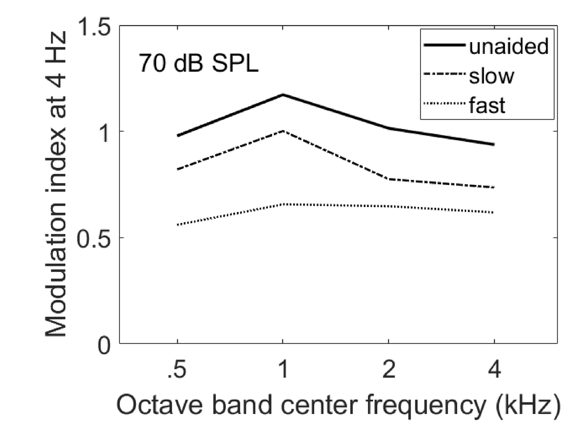


Figure 3: The effect of compression speed.

COMPRESSION SPEED. AT_{slow} = 12ms, RT_{fast} = 1000ms, AT_{fast} = 5ms, RT_{fast} = 50ms. Speech at 70 dB SPL in quiet. The faster compression speeds allow the hearing aid to react to syllabic rate changes in the level of the input signal, and thus reduce the natural dynamic variation in the speech. This was reflected as lower MI with fast than with slow acting compression.

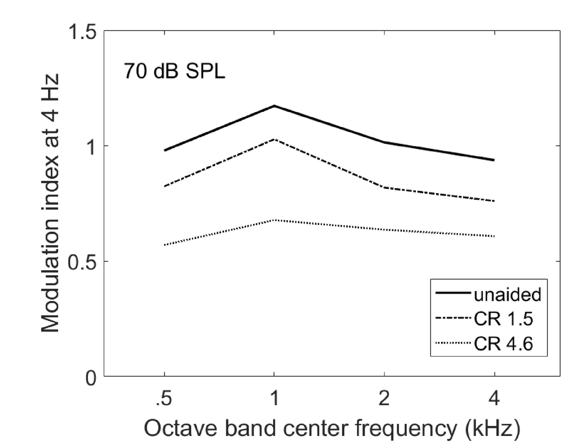


Figure 4: The effect of compression ratio.

COMPRESSION RATIO. Compression ratios 1.5 and 4.6, AT = 5ms, RT = 50ms. Speech at 70 dB SPL in quiet. The increase in compression ratio resulted in decrease of MI across all modulation frequencies. The same effect was measured for all input levels from 50 to 80 dB SPL (not displayed). The effect was only observed with fast acting compression. The MI was the same for CR = 1.5 and 4.6 with SAC.

NOISE REDUCTION. No difference between processing conditions measured in quiet (left). In the presence of noise (Speech at 80 dB SPL and noise at 15 dB SNR from 0°) the traditional noise reduction ("NR on") and no noise reduction ("NR off") conditions resulted in identical MI. When using the NR algorithm, which attempts to maximize the speech intelligibility index (SII) ("NR SE"), the MI was higher than with other processing conditions and unaided condition. Same observations were measured also for 5 and 10 dB SNR (not displayed).

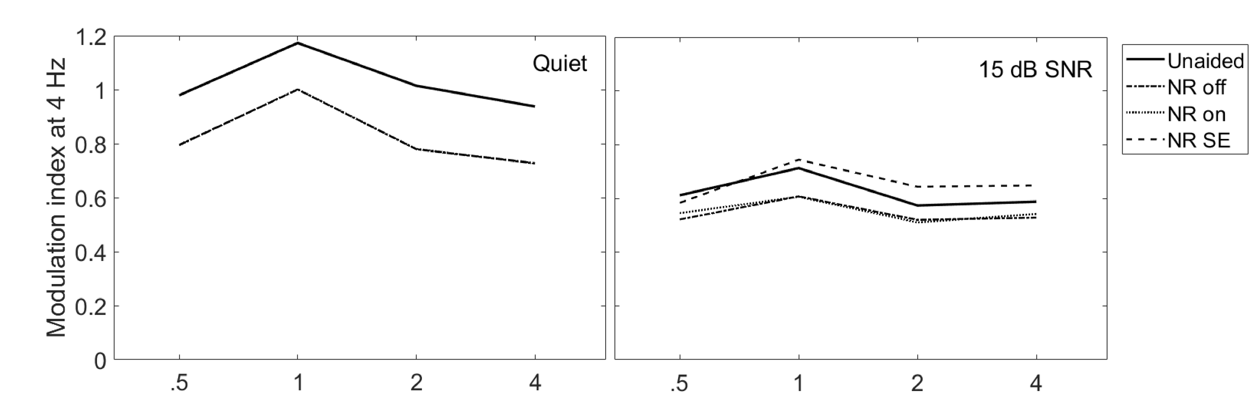


Figure 5: The effect of digital noise reduction algorithm.

DIRECTIONAL MICROPHONE. Speech at 70 dB SPL from 0° and noise from 0° (figure left) and 180° (figure right) at 10 dB SNR. No difference was measured between microphone modes when noise originated from 0°, When noise originated from 180° the MI was the lowest in omnidirectional mode due to absence of pinna shadowing effects. Directional microphone increased the MI above unaided condition due to reduction of background noise.

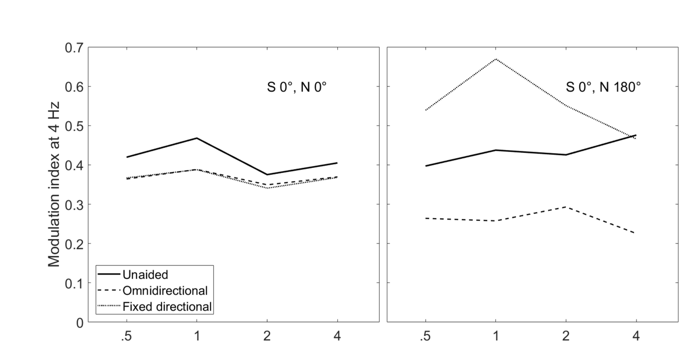


Figure 6: The effect of directional microphone.

COMPARISON OF TWO DEVICES

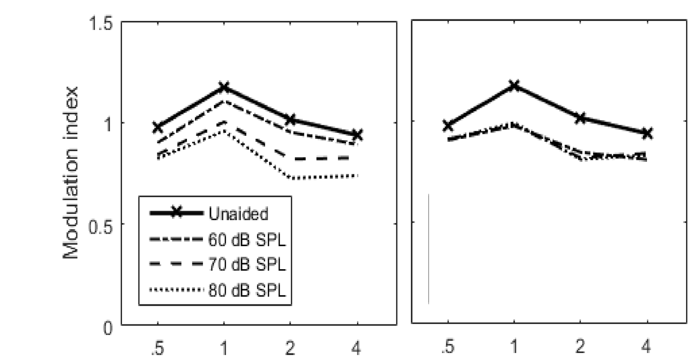


Figure 7: Modulation index for HA1 (left) and HA2 (right) with speech at 60, 70 and 80 dB SPL in the quiet.

SPEECH IN QUIET. For HA1 the MI was dependent on the input level, whereas HA2 resulted in the same MI for all input levels. This indicates non-adaptive processing with HA2 with same processing applied regardless of the input level. In quiet, the features that were triggered by noise were likely not active. Therefore, the likely source of reduction in MI was wide dynamic range compression (WDRC). Faster compression speeds result in lower MI. Therefore, a hypothesis is that the HA2 included relatively fast fixed speed compression. HA1 on the other hand may have included fast processing time constants only at louder levels.

SPEECH IN NOISE. Differences in MI were also ween in noise between the two devices. HA1 resulted in higher MI than HA2 when noise originated from 0° for both 70 and 80 dB SPL speech levels and all SNRs. Increase in MI may reflect decrease in background noise. When noise originated from 180° the aided MI was generally higher than the unaided MI for HA1 at both 70 and 80 dB SPL speech input. With HA2 the increase in MI occurred only at 80 dB SPL speech level suggesting higher activation level for directional microphone than with HA1.

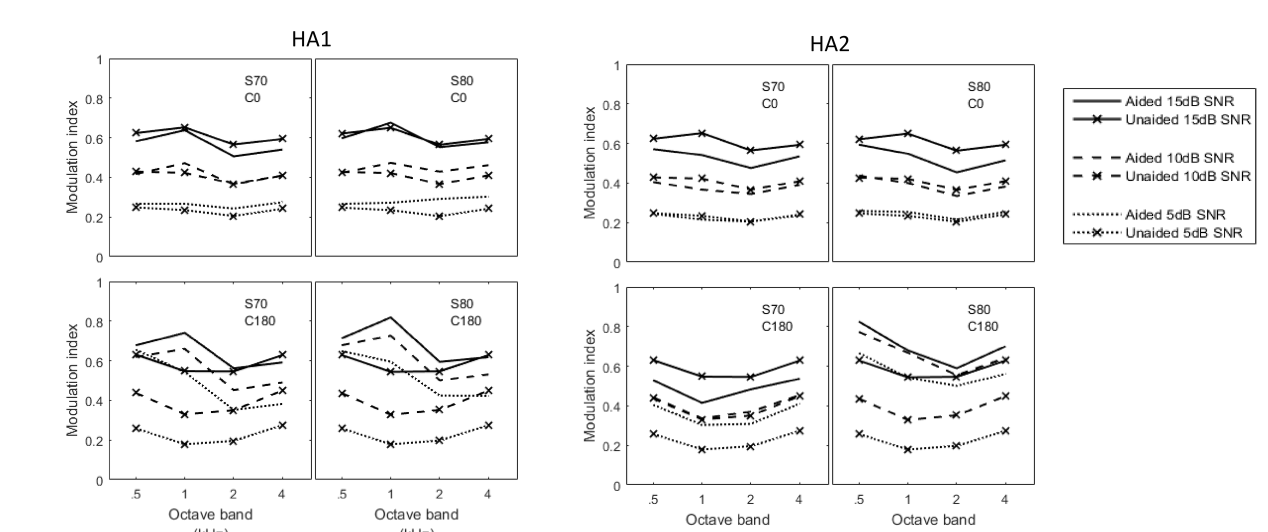


Figure 8: Modulation index for HA1 (left) and HA2 (right) with speech presented at 70 or 80 dB SPL in the presence of noise from 0° or 180°.

DISCUSSION

This study demonstrated that the measurement of modulation spectrum may be used as a source of additional information on the effects of hearing aid processing on temporal envelope characteristics. Measurement of modulation spectrum allows the use of natural speech signal in quiet or in noise at levels which activate the advanced features in a manner that is expected in real life use. In the absence of a formalized tool to investigate temporal processing in commercial hearing aids, the inclusion of a measurement of modulation spectrum as a means to study the changes in modulation characteristics of the input as a consequence of the hearing aid signal processing could be considered.

REFERENCES

Souza, P., Arehart, K., Shen, J., Anderson, M., & Kates, J. (2015). Working memory and intelligibility of hearing-aid processed speech. Front Psychol, 6:1-14.
Gallun, F., & Souza, P. (2008). Exploring the role of the modulation spectrum in phoneme recognition. Ear Hear, 29:800-813.