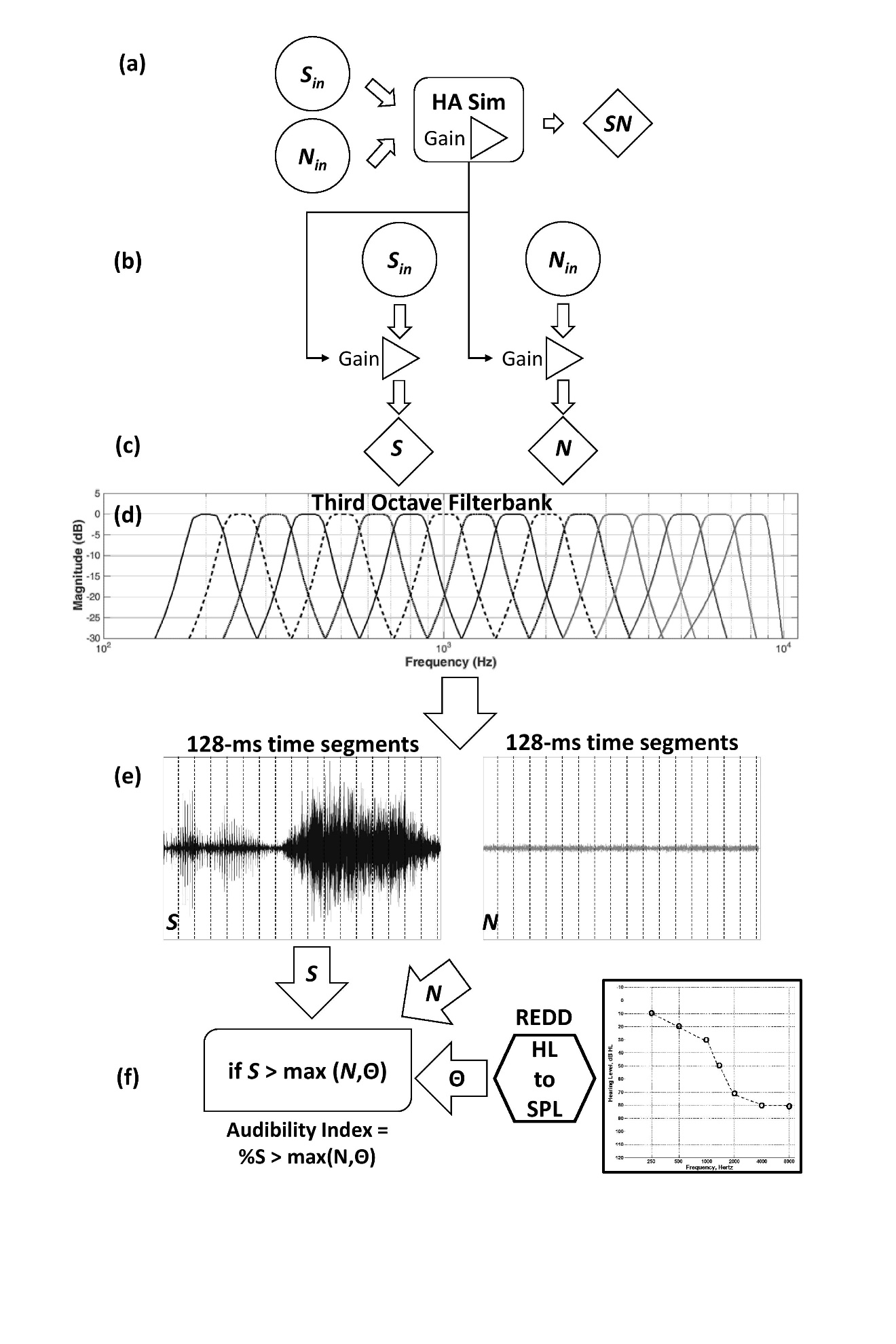
**Appendix B**

**Modified Audibility Index (AUD)**

Figure B1 shows the steps involved in computing audibility using the modified audibility index (AUD; Alexander & Rallapalli, 2017). Stimuli that were presented to each listener for each amplification method were re-processed in the hearing aid simulator with identical parameters and procedures. The instantaneous gain values obtained from the speech-in-noise mixtures were then applied to the constituent speech and noise files individually. This was done to ensure that any changes to the output SNR that took place after amplification were accounted for while computing audibility. For linear processes like the ones used here, this method is equivalent to the phase inversion method that has been used to separate speech and noise signals from a signal-processed mixture (Alexander & Rallapalli, 2017; Alexander & Masterson, 2015; Hagerman & Olofsson, 2004; Naylor & Johannesson, 2009; Souza et al., 2006).

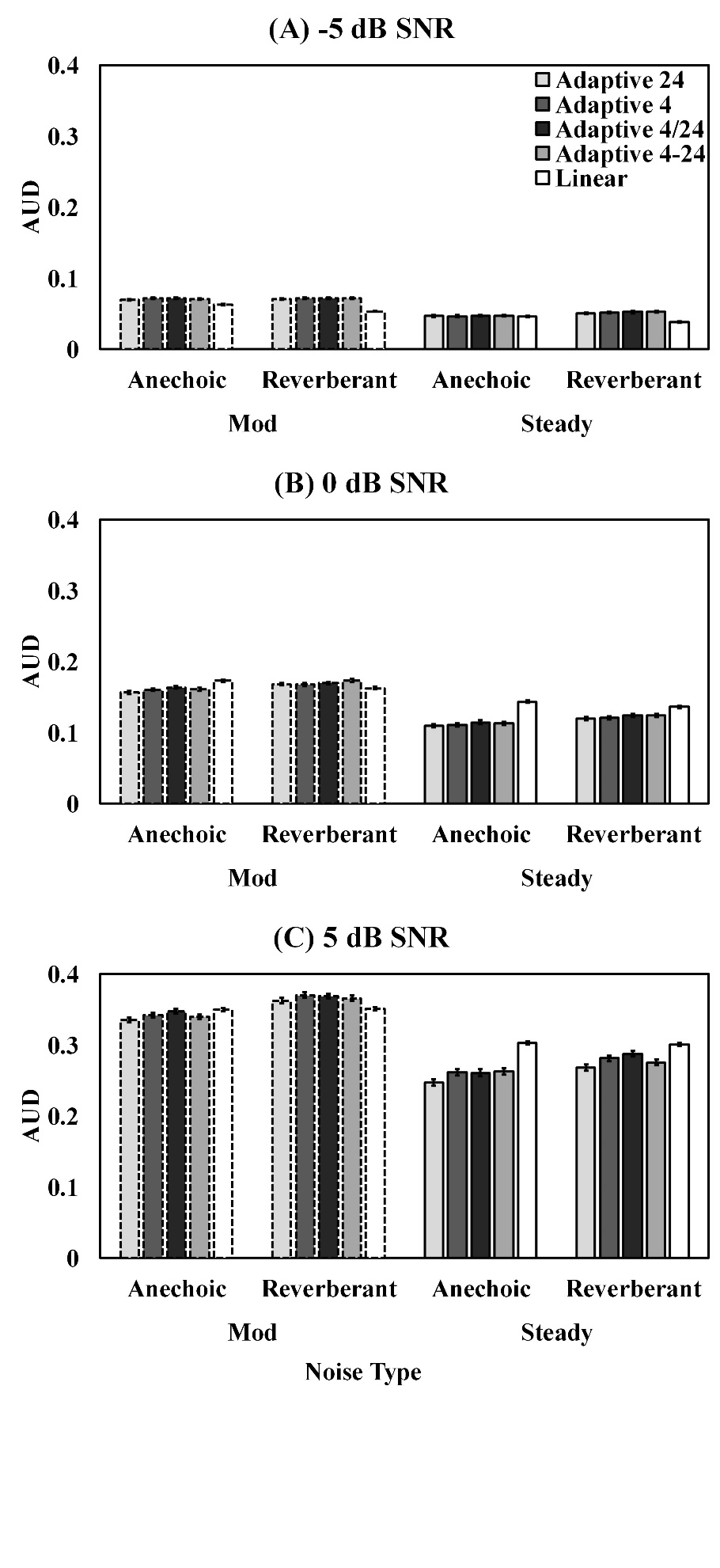
For the reverberant condition, the hearing aid gains obtained from the reverberated speech-in-noise signal were applied to the direct speech signal along with its early reflections (< 50 ms) to compute the ‘speech levels.’ These same gains were applied to direct noise signal and all of its reflections plus the late reflections (> 50 ms) from speech to compute the ‘noise levels.’ This was done for two reasons. First, to account for potential improvements in speech intelligibility due to increased intensity of speech from early reflections (e.g., Bradley, Sato, & Picard, 2003; Srinivasan, Stansell, & Gallun, 2017). Second, to avoid overestimation of audibility of the speech signal with reverberation because the late reflections add to the signal level, but do not add to intelligibility (Bradley, Reich, & Norcross, 1999), effectively contributing to the overall noise in the signal.



**Figure B1.** Steps used to compute audibility for the test sentences. (a) Input speech (*Sin*) and noise (*Nin*) signals were processed in the hearing aid simulator (HA Sim). (b) Time-varying gain from each hearing aid channel for the speech-in-noise mixture (*SN*) was saved and applied to *Sin* and *Nin* separately (c) to produce amplified *S* and *N*. (d) *S* and *N* were separated into narrowband signals using a 1/3 octave filterbank. (e) Narrowband signals were divided into 128-msec time segments. (f) The SPL of the speech segment was compared to the SPL of the noise and to the interpolated audiometric thresholds in dB SPL. The proportion of audible speech segments was used as the index of audibility.

After applying the appropriate gains, the speech and noise signals were individually divided into narrowband signals by passing them through a 1/3 octave filterbank (ANSI S1.11, 2004). The narrowband signals were further divided into 128-msec time segments, using Hann windows with 50% overlap. The estimated real-ear SPL (derived from the headphone transfer functions on KEMAR) of the speech signal in each segment was compared to the estimated real-ear SPL of the audiometric threshold at the center frequency of the analysis filter and to the estimated real-ear SPL of the corresponding noise segment. Thresholds for the center frequencies were obtained by linear interpolation of the audiometric thresholds in the octave frequencies ranging from 250 Hz to 8000 Hz for the test ear. The index of audibility (AUD) was the proportion of audible speech segments between the 1st and 70th percentiles (the peaks and valleys, respectively) that were above the estimated real-ear SPL of the audiometric threshold or the noise, whichever was higher. AUD reported in this study is the mean audibility (i.e., proportion of audible segments) across the 1/3 octave filters. This measure is useful in understanding the operational differences across the different amplification methods in a given environmental context.

AUD across amplification methods is shown in Figure B2. A repeated-measures ANOVA was used to determine the effects of amplification method, noise type, reverberation condition, and SNR on audibility (Table B1). Effect size and post-hoc analyses were conducted using the methods described above. There were significant main effects for each of the factors, which were qualified by significant interactions[[1]](#footnote-2),[[2]](#footnote-3). As the level of noise increased (decreased SNR), the effects of environmental acoustics on AUD (noise type and reverberation) and signal processing (amplification method) became smaller. At -5 dB SNR, differences across amplification methods were significant for both types of noise, but the effect was greater for modulated noise (*F*[4,140] = 39.43, *p* < 0.001, ηp2 = 0.155) than for steady noise (*F*[4,140] = 7.07, *p* < 0.001, ηp2 = 0.032). However, at 0 dB and 5 dB SNR, the effect of noise type on AUD across amplification methods was the opposite. Specifically, differences in AUD across amplification methods were significant in steady noise but not in modulated noise (p>0.05). Post-hoc tests among the different amplification methods revealed that the significant effect of amplification method at -5 dB SNR was present because AUD with the adaptive amplification methods was always greater than with linear amplification, regardless of the noise type (p<0.05). On the other hand, a significant effect of amplification method in steady noise at 0 dB and 5 dB SNRs occurred because AUD with linear amplification was always greater than AUD with all the adaptive amplification methods (p<0.001). At 5 dB SNR, AUD with Adaptive 24 was significantly lower than AUD with Adaptive 4 and Adaptive 4/24 in steady noise.



**Figure B2.** Mean audibility or “AUD” across the test sentences averaged across thirty-six audiograms (listeners). See Figure 3 caption in the main document for a complete description.

Table BI. ANOVA for the main effects and interactions of amplification method (AMP), noise type (Noise), reverberation condition (Reverb), and SNR on audibility (AUD).

|  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- |
| Source | dfsource | dferror | F | p | ηp2 |
| **SNR** | 2 | 70 | 17617.0 | <.0001\*\*\* | 0.942 |
| **Noise** | 1 | 35 | 1770.72 | <.0001\*\*\* | 0.451 |
| **SNR \* Noise** | 2 | 70 | 218.81 | <.0001\*\*\* | 0.169 |
| **Reverb** | 1 | 35 | 52.77 | <.0001\*\*\* | 0.024 |
| **SNR \* Reverb** | 2 | 70 | 21.40 | <.0001\*\*\* | 0.019 |
| **Noise \* Reverb** | 1 | 35 | 0.00 | 0.9893 | 0.000 |
| **SNR \* Noise \* Reverb** | 2 | 70 | 1.50 | 0.2305 | 0.001 |
| **AMP** | 4 | 140 | 7.79 | <.0001\*\*\* | 0.014 |
| **SNR \* AMP** | 8 | 280 | 9.23 | <.0001\*\*\* | 0.033 |
| **Noise \* AMP** | 4 | 140 | 14.25 | <.0001\*\*\* | 0.026 |
| **SNR \* Noise \* AMP** | 8 | 280 | 2.77 | 0.0058\*\* | 0.010 |
| **Reverb \* AMP** | 4 | 140 | 10.04 | <.0001\*\*\* | 0.018 |
| **SNR \* Reverb \* AMP** | 8 | 280 | 0.51 | 0.8472 | 0.002 |
| **Noise \* Reverb \* AMP** | 4 | 140 | 0.38 | 0.8206 | 0.001 |
| **SNR \* Noise \* Reverb \* AMP** | 8 | 280 | 0.20 | 0.9905 | 0.001 |

\*\*\*p-values with statistical significance <0.001 \*\*p-values with statistical significance <0.01

Recall that AUD accounts for the proportion of audible speech above an individual’s audiometric threshold and the noise level in a given segment. In other words, the output SNR also plays a role in determining the AUD. At negative SNRs, the signal is dominated by noise and therefore there is an increase in AUD with compression, especially with modulated noise because more gain is provided to the softer speech segments.

At the positive SNR, compression causes a reduction in output SNR, especially with steady noise (cf. Alexander and Masterson, 2015), thereby negatively affecting audibility relative to linear amplification. This occurs because lower-level noise segments during the pauses of speech are given more gain than the higher-level speech in surrounding spectro-temporal regions. At the highest SNR, only the Adaptive 24 method resulted in significantly lower audibility for speech compared to Adaptive 4 and Adaptive 4/24. This is not unexpected because AUD takes into account the output SNR which has been shown to decrease with increasing number of channels (Alexander & Masterson, 2015).

The two-way interaction between SNR \* reverberation condition was also significant. Separate ANOVAs at each SNR showed that AUD was greater for the reverberant condition than the anechoic condition at 0 dB (*F*[1,35] = 7.31, *p* < 0.05, ηp2 = 0.008) and 5 dB (*F*[1,35] = 20.65, *p* < 0.001, ηp2 = 0.023) SNR. However, the effect of reverberation condition on AUD was not significant at -5 dB SNR, likely due to floor effects.

An increase in audibility is expected to be most advantageous when the input level of speech is relatively low, such as at a low SNR. While there was an overall improvement in speech audibility (as measured by the AUD) for the adaptive compression methods over linear amplification, there were no significant differences among the adaptive methods at a negative SNR. This suggests that no additional audible information was available to the listeners to improve speech recognition in these conditions. Given that amplification method had the smallest effect size compared to other conditions[[3]](#footnote-4), including SNR, noise type, and reverberation, the differences in audibility were not sufficient to negatively affect speech recognition ability. A small effect size for amplitude compression has been reported in a previous study that used similar background noise conditions (Alexander & Masterson, 2015). Therefore, it is not surprising that the addition of reverberation to the noise in this study would have further minimized differences among the amplification methods (Reinhart et al., 2017).

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1. The three-way interaction between SNR \* noise type \* amplification method was significant because the effect of amplification method varied by noise type to different extents depending on the SNR. This was shown by a significant interaction between amplification method and noise type at all SNRs with effect size decreasing from 5 dB SNR (*F*[4,140] = 8.49, *p* < 0.001, ηp2 = 0.038) to -5 dB SNR (*F*[4,140] = 2.66, *p* < 0.05, ηp2 = 0.012). [↑](#footnote-ref-2)
2. Although there was a significant interaction between reverberation condition \* amplification method, post-hoc t-tests at each reverberation condition showed no significant effect of amplification method on AUD (*p* > 0.05) probably because of the very small effect size (ηp2 < 0.01). [↑](#footnote-ref-3)
3. The average difference in AUD between the adaptive compression methods and linear amplification (small effect size) was 0.01. Similarly, the average difference in speech recognition between the adaptive compression methods and linear amplification was 2.7 RAUs (or 2.4%). In comparison, the average increase in AUD from the least to the highest SNR (large effect size) was 0.26 and the improvement in speech intelligibility was by 75.79 RAUs (or 62.57%). [↑](#footnote-ref-4)