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**Prediction of speech intelligibility based on a correlation metric in the envelope power spectrum domain**

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**Introduction**

A powerful tool to investigate speech perception is the use of speech intelligibility prediction models. Recently, a model was presented, termed correlation-based speech-based envelope power spectrum model (sEPSM)\(^b\) [1], based on the auditory processing of the multi-resolution speech-based Envelope Power Spectrum Model (mr-EPSM)\(^c\) [2], combined with the correlation back-end of the Short-Time Objective Intelligibility measure (STOI) [3]. The sEPSM\(^d\) can accurately predict NH data for a broad range of listening conditions, e.g., additive noise, phase jitter and ideal binary mask processing.

The sEPSM\(^d\) model includes audibility thresholds, such that sensitivity loss can be accounted for beyond the audiogram, but other types of hearing impairment (HI) cannot be simulated using this framework. However, speech perception can vary greatly among listeners even when hearing sensitivity is similar. Therefore, the predictive power of the sEPSM\(^d\) back-end was further investigated in combination with a more realistic auditory pre-processing front-end adopted from the computational auditory signal processing and perception model (CASP) [4]. Here, the speech-based CASP (sCASP) was incorporated based on the audiogram, but other types of hearing impairment (HI) were also considered.

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**sEPSM\(^d\) model**

The sEPSM\(^d\) model consists of an auditory pre-processing front-end and a modulation envelope extraction and metric computation back-end. The front-end includes a Gammatone filterbank, with a fixed linear gain and a time window of 1 ms to determine the modulation depth of 1. and the speech-like but non-semantic international speech test signal (IEEE 1240-2014). The models were evaluated in conditions with: additive noise, reverberant speech, and phase jitter distortions of 0.75 Hz. The model can now serve as foundation for the development of a HI model, since the CASP-based framework allows for fitting to individual hearing impairments.

**Towards prediction of HI data**

The CASP model offers more flexibility to model hearing impairments, beyond the audiogram, due to the Dual Resonance Non-linear filterbank (DRNL) [5]. The model has been shown to account for psychophysical data from individual HI subjects.

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**Test conditions**

The models were evaluated in conditions with:

- Speech mixed with stationary or non-stationary interferers: Speech shaped noise (SSN), which was also used to fit the model; Amplitude modulated SSN (AMSSN) with \( f_{c,mod} > 8 \) Hz and modulation depth of 1. and the speech-like, but non-semantic international speech test signal (IEEE 1240-2014).
- Noisy speech in the presence of reverberation: \( T_{60} = 0, 0.4, 0.7, 1.3 \) and 2.3 s.
- Noisy speech subjected to different types of non-linear processing:
  - Ideal Binary Mask processing (IBM) with four interferers.
  - Phase jitter distortion \( \left( r = \text{R}_c(\alpha(\Theta(t))) / \text{R}_c(\alpha(\Theta(t))) \right) \text{ with } \Theta(t) = [0, 2\pi] \), \( \alpha = 0.125 \).

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**Fitting of the models**

The models are fitted per speech material to the condition of clean speech with SSN by fitting a sigmoid function between the model outputs and the human scores.

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**Results**

**Human data** vs. **sEPSM\(^d\)** vs. **STOI** vs. **sEPSM\(^d\)** vs. **sCASP**

**Summary of results**

The sCASP model provides similar (and in some conditions better) results than the sEPSM\(^d\).

The model can now serve as foundation for the development of a HI model, since the DRNL-based framework allows for fitting to individual hearing impairments.

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**Outlook**

- Investigate the model’s ability to account for individual hearing impairments using the parameters available in the CASP framework.
- Consider additional processing stages that could account for inner hair-cell loss and auditory nerve deafferentation (Sumner et al., 2002; López-Poveda and Barrios, 2013), as they are likely to be determinant in speech-in-noise related tasks.
- Determine the conditions on which the HI model will be tested with special focus on supra-threshold distortions that might be challenging for HI subjects.

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