Inverse correlation of intelligibility of speech in reverberation with the amount of overlap-masking

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1. Introduction

Late reflections often help speech intelligibility [1], whereas early reflections often help speech intelligibility, and this is called the Haas effect (e.g., [2]). It has been reported that the main cause of degradation in speech intelligibility in reverberant environments is overlap-masking [3–5]. Because of overlap-masking, reverberant components of prior speech segments mask successive segments. As a result, speech segments following reverberating segments are more difficult to understand. As the energy of the prior segments increases, the effect of overlap-masking also increases. This is particularly important when the preceding segment is a vowel, which has more power, and the subsequent segment is a consonant, which has less power [6,7].

A number of researches have proposed and discussed how the intelligibility of speech in reverberation can be estimated from an impulse response of a room. Reverberation time, such as $T_{60}$, is a simple objective parameter for estimating reverberation [8]. Speech intelligibility usually decreases as $T_{60}$ becomes longer, but different rooms having the same $T_{60}$ might yield different degrees of speech intelligibility. One example is the case where $T_{60}$ is the same in different rooms, but the energy ratios of the direct-to-reverberated sounds are different. The Deutlichkeit value, such as $D_{50}$ [9,10] and Clarity, such as $C_{50}$ [9,11], take this direct-to-reverberation ratio into account. The speech transmission index (STI) is another parameter that is widely used to measure speech intelligibility objectively [12,13]. STI is based on the fact that the modulation transfer function depends on reverberation [14].

The intelligibility of speech also depends on the speech signal itself. To reduce overlap-masking, Arai et al. [6,7] proposed “steady-state suppression” as a preprocess for speech signals in reverberant environments. Strange et al. [15] showed that the information in steady-state portions of a speech signal was relatively insignificant compared with the information in transient portions. Additionally, steady-state portions usually have more energy compared with transient portions. In the “steady-state suppression” technique, overlap-masking is reduced by estimating and suppressing steady-state portions of speech that have high energy but are less important for speech perception, such as the nuclei of syllables. From the results of several experiments in simulated and real sound fields, we found that when we apply this process between a microphone and loudspeaker, it significantly improves speech intelligibility in reverberant environments (reverberation times of 0.7–1.3 s) [e.g., 6,7,16,17].

The conventional measures for estimating the intelligibility of speech in reverberation, which are based on the impulse response of a room, are independent of the speech signal itself. Therefore, they do not reflect the effect of any preprocesses, including nonlinear processing (e.g., the steady-state suppression technique), which are designed to be applied to the original speech signal. There are several empirical STI approaches for predicting the intelligibility of nonlinearly processed speech [18–21]. These approaches could handle nonlinearly processed speech, such as cochlear- implant processed speech.

In this study, we propose a new intelligibility measure that can take into account the effect of nonlinear preprocesses, with a view toward using the steady-state suppression technique to reduce the amount of overlap-masking (OLM) in reverberant environments. In particular, we show how this measure correlates with speech intelligibility.

2. Proposed measure for estimating intelligibility of speech

First, we focus on a target syllable within an arbitrary sentence. Figure 1 shows a conceptualized speech waveform. In this figure, $s(t)$ denotes a target syllable, whereas $p(t)$ denotes a sequence of pretarget syllables. We define $t = 0$ at the boundary between $p(t)$ and $s(t)$ on the horizontal (time) axis. Then, a new measure, the signal-to-OLM ratio $S_{OR}$, is defined as

$$S_{OR} = 10 \log_{10} \frac{\int_0^T |s(t) * h_{50}(t)|^2 \, dt}{\int_0^T |p(t) * h(t)|^2 \, dt} [\text{dB}],$$

where $h(t)$ is the impulse response of a room and $h_{50}(t)$ is the first 50 ms of the impulse response. In this case, the direct sound starts at $t = 0$ in the impulse response. The variable $T$ is...
the duration for calculating the measure. We used $T = 150\text{ ms}$ in the following experiment because in this study, we focus particularly on the signal-to-OLM ratio within the initial consonant of a target syllable that is located at $t = 0\text{–}150\text{ ms}$ of $s(t)$ in the experimental setup of this study.

3. Evaluation

To evaluate how the speech intelligibility correlates with the amount of overlap-masking, we compared the correct rate obtained from the following perceptual experiment with $\text{SOR}$.

3.1. Reverberant conditions

We conducted a perceptual experiment under artificial reverberant conditions by convolving speech samples with impulse responses. The impulse responses were artificially synthesized from white noise by multiplying temporal envelopes. Table 1 shows ten impulse responses with different combinations of $T_{60}$ and $D_{50}$.

3.2. Speech samples

The original speech samples consisted of 14 nonsense consonant-vowel (CV) syllables (target syllables) embedded in a Japanese carrier phrase, “Daimoku to shite wa _____ to iimasu” (It is called ____ as a title). The vowel was /a/ and the consonants were /p, t, k, b, d, s, j, h, dz, dʒ, tj, m, n/. The speech samples were obtained from the ATR Speech Database of Japanese. The same carrier sentence was used for all targets. The beginning position of the target vowel was adjusted to 150 ms from the offset of the pretarget carrier phrase. The ratio of the root-mean square (RMS) in the carrier phrase to that in the CVs was 1:0.7. Finally, we prepared original speech samples and processed speech samples by steady-state suppression following the method of Arai et al. [6,7] and further, added a step to avoid suppressing relatively longer continuants where the spectral moment is higher than 3,750 Hz [22], such as sibilant consonants.

3.3. Perceptual experiment

The stimuli were the speech samples convolved with each of the ten impulse responses used in this study.

3.3.1. Participants

Twenty-two young people with normal hearing (15 males and 7 females, aged 20 to 26 years) participated in the experiment. All were native speakers of Japanese.

3.3.2. Procedure

The experiment was conducted in a soundproof room. Stimuli were presented diotically through headphones (STAX SR-303) connected to a computer via the digital-to-analog (D/A) converter of a digital audio amplifier (MA-500U, Onkyo) that was connected to the computer via a USB interface. The sound level was adjusted to each participant’s comfort level during a training session prior to the experiment. A stimulus was presented in each trial and the listeners were instructed to select one of 16 options, including 14 CVs, vowel /a/, and ‘others,’ displayed on the computer screen. The experiment was carried out at each listener’s pace. For each listener, 280 stimuli were presented randomly (10 reverberation conditions $\times$ 14 CVs $\times$ 2 processing conditions).

3.4. Results and discussion

Table 2 shows the mean percentage of correct responses in the perceptual experiment (the second column) and the $\text{SOR}$ value (the third column) for the ten reverberant conditions. The first and second rows of each cell show the speech intelligibility of nonprocessed speech samples and speech samples processed by steady-state suppression, respectively. We can see that steady-state suppression improves speech intelligibility as well as increases the $\text{SOR}$ value under most of the experimental conditions used in this study.

Figure 2 shows the scatter plot of the speech intelligibility versus the $\text{SOR}$ value for each pair of a nonprocessed condition (unfilled circle) and a processed condition (filled circle); a line connects the two conditions. The correlation coefficient among all of the points in this figure is 0.7025, which shows a high correlation between speech intelligibility and the $\text{SOR}$ value.

4. Conclusions

In this study, we investigated the correlation between the intelligibility of speech in reverberation and the amount of overlap-masking (OLM) due to reverberation. There was a high correlation between the results of a perceptual experiment and the values of the newly proposed intelligibility measure, $\text{SOR}$, defined as the signal-to-OLM ratio. In other
words, the intelligibility of speech in reverberation was inversely correlated with the amount of overlap-masking.

Two advantages of using our proposed measure are that: 1) it reflects the reverberation characteristics of a room, as contained in the impulse response of the room, and 2) it also reflects the characteristics of the speech signal itself, as well as the effect of any preprocesses, including nonlinear processing (e.g., steady-state suppression), applied to the original speech signal.

This time, we did not divide the speech signal into frequency bands but treated the signal as one band. However, we can also determine SOR by filter bank analysis based on the auditory filter. In this case, a more realistic amount of overlap-masking is estimated and the correlation between the intelligibility of speech and the amount of overlap-masking might be improved. The correlation can also be calculated for each consonant. In the future, we will use “model speech,” such as amplitude-modulated white noise, instead of actual speech signals.

Acknowledgments

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References


Table 2  Mean percentages of correct responses of the perceptual experiment (the second column) and the SOR value (the third column) for the ten reverberant conditions. The first and second rows of each cell show speech intelligibility of nonprocessed speech samples and speech samples processed by steady-state suppression, respectively.

<table>
<thead>
<tr>
<th>Condition</th>
<th>Intelligibility [%]</th>
<th>SOR [dB]</th>
</tr>
</thead>
<tbody>
<tr>
<td>Rev1</td>
<td>69.2</td>
<td>−20.0</td>
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<tr>
<td></td>
<td>72.7</td>
<td>−15.3</td>
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<tr>
<td>Rev2</td>
<td>48.4</td>
<td>−22.6</td>
</tr>
<tr>
<td></td>
<td>53.9</td>
<td>−18.2</td>
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<tr>
<td>Rev3</td>
<td>58.1</td>
<td>−18.8</td>
</tr>
<tr>
<td></td>
<td>61.4</td>
<td>−14.4</td>
</tr>
<tr>
<td>Rev4</td>
<td>58.8</td>
<td>−15.0</td>
</tr>
<tr>
<td></td>
<td>62.3</td>
<td>−10.7</td>
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<td></td>
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<td></td>
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</tr>
<tr>
<td></td>
<td>60.4</td>
<td>−12.9</td>
</tr>
</tbody>
</table>

Fig. 2  Scatter plot of speech intelligibility versus the SOR value for each pair of a nonprocessed condition (unfilled circle) and a processed condition (filled circle); a line connects the two conditions.
[13] IEC 60268-16 Ed. 3.0, Sound system equipment — Part 16: Objective rating of speech intelligibility by speech trans-
mission index (2003).
mission Index (TNO Human Factors, Soesterberg, 2002).