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TEMPORALLY ENHANCED SPEECH IS MORE INTELLIGIBLE IN REVERBERANT ENVIRONMENTS

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ABSTRACT

Reverberation causes degradation in speech comprehension, especially for elderly people, the hearing-impaired and non-native listeners. In order to prevent intelligibility degradation, we developed several pre-processing techniques, where signals are processed before being radiated through the loudspeakers of a public address system. The two main techniques are modulation filtering and steady-state suppression, both of which enhance the temporal dynamics of speech. Previously we had found that the important frequencies of temporal dynamics for speech perception, the modulation frequencies, lie between 1-16 Hz (Arai et al., 1999). Therefore, we configured the modulation filtering to emphasize these modulation frequencies (Kusumoto et al., 2005). In this paper, we mainly discussed steady-state suppression (Arai et al., 2001, 2002), which suppresses steady-state portions of speech to reduce overlap-masking and improve speech intelligibility for young, elderly, and non-native listeners in reverberant environments (Hodoshima et al., 2006a; Miyauchi et al., 2005; Hodoshima et al., 2006b). Especially, we discussed how the “pre-processing” technique, such as the steady-state suppression, is effective with recent results including speech-rate slowing with the steady-state suppression exceeding a simple speech-rate slowing approach (Arai et al., 2005).
KEYWORDS: Speech enhancement, Reverberation, Speech Intelligibility, Steady-State Suppression

INTRODUCTION

Reverberation is a large problem to overcome in attempting to provide a barrier-free environment for speech communication. This is especially true for hearing-impaired and/or elderly people [1], as well as non-native listeners [2]. Acoustic variations existent in different examination rooms make impossible a perfectly fair comparison of speech comprehension for different listeners tested in different rooms. Thus, designing acoustics of a room becomes increasingly important for creating a universal environment.

It is known that strong reverberation affects speech intelligibility. Although early reflections often help speech intelligibility (the Haas effect, e.g., [3]) late reflections degrade speech intelligibility [4]. Because multi-purpose halls are used not only for music performances but also for lecture and other activities involving speech, such rooms would preferably be designed to enhance speech perception, while not degrading music quality. However, there is a conflict, as music and speech are best enjoyed or understood under opposite acoustic environments; i.e., longer reverberations are preferable for music while they have a negative affect on speech perception.

Several approaches have been proposed and discussed to improve speech intelligibility degraded by reverberation. Some approaches are based on architectural acoustics and others on electrical acoustics. In the latter case, there are two main approaches: 1) having a device at the listener's side, and 2) having a device at the public access (PA) side.

In case 1, several "post-processing" techniques have been proposed for dereverberation using a single microphone or a microphone array [5-8]. In this case, the listener has a device at his/her side through which he/she listens to processed speech sounds with earphones. Case 2 is called "pre-processing"; speech enhancement is done before speech signals are broadcast into a room, through a device embedded in the PA system of the room. In this case, a speech signal can be directly captured from a microphone, or recorded and stored in a playback system, etc. Because the signal is not yet reverberant, and therefore, has original speech features without smearing, the signal may be effectively enhanced. This is the great advantage of case 2 over case 1.

One of the early attempts at pre-processing was done by Langhans and Strube [9], where they filtered the temporal envelope in each subband of a speech signal. We call it "modulation filtering" because we can model a speech signal as the modulation of a source signal by the dynamics of speech organs, and it filters the modulation pattern of speech. In their report, however, they could not obtain any significant improvement in terms of speech intelligibility.

We applied modulation filtering as a pre-processing technique as Langhans and Strube [9] did and conducted several perceptual experiments [10-13]. These studies are based on the fact that reverberation has a lowpass-type modulation transfer function (MTF) [14]. The modulation spectrum of speech, which is defined as a frequency analysis of the temporal envelopes of a speech signal, usually has a peak around 4 Hz, and it is the reflection of the syllable-based temporal property of speech [15,16]. In reverberant environments, the modulation spectrum of speech has a peak in a much lower modulation-frequency range and the peak has a much lower modulation index. Therefore, in the studies above, we enhanced the temporal envelopes of speech by the modulation filters having the inverse characteristics of the MTF.

While conventional modulation filtering used linear processing for the temporal envelopes of speech, we proposed the steady-state suppression technique where the filtering is non-linear processing [17,18]. This technique directly reduces "overlap-masking" due to reverberation, and when we used it as a pre-processing technique, we showed in [17,18] that it prevents the degradation of speech intelligibility in a reverberant environment.

Self-masking and overlap-masking are two major causes of degradation of speech intelligibility in reverberant environments [19,20]. Self-masking occurs within each segment resulting in a modification of the segment itself. Additionally, the transition, such as the onset and offset of a segment, is smeared. Overlap-masking occurs among segments where the reverberation tails of the previous segments mask the following segments. The effect of overlap-masking is greater when a previous segment is more powerful than a following segment, as in the case of a vowel followed by a consonant. Therefore, steady-state suppression suppresses the power of the stronger steady-state portions, such as those found in vowels, which would typically have the stronger overlap masking effect on following weaker segments. This technique is rational because the information in the steady-state portions of the speech signal is relatively redundant with that in the transient segments [21,22]. Furthermore, neither steady-state suppression nor modulation filtering changes the total length of an utterance. Maintaining the original utterance length is particularly important when the talker is in the same room from which processed speech is broadcast.

This type of pre-processing can also be applied when the talker is absent from the room in which processed speech is broadcast. In these situations, speech sounds can be recorded and/or processed in advance. A typical example of such a condition would be an emergency broadcast in a tunnel. In this case, the time scale of the speech signal can also be modified. Moreover, we empirically know that speaking slowly helps to increase speech intelligibility, particularly in a large hall with a long reverberation time (RT). Bolt and MacDonald [19] reported that speech intelligibility is greatly increased by speaking slowly in reverberation. Thus, we can also design a new algorithm that stretches a speech signal using a time-scale modification technique to decrease the speech rate.

Stretching a speech signal alone is not the best solution, though, because in and of itself stretching does not significantly reduce the amount of overlap-masking. Instead, isolating each transition part of the slowed speech might be more effective. Therefore, we proposed two new approaches: one is the "zero-padding" approach and the other is the steady-state suppression after speech-rate slowing [23,27].

In this paper, we discuss the effectiveness of various aspects of the steady-state suppression technique. The first experiment deals with simulated reverberation for young normal listeners [24]. The second and third experiments are for elderly [25] and non-native listeners [26], respectively. The fourth experiment involves speech-rate slowing [23].

EXPERIMENT 1: SIMULATED REVERBERATION

In this experiment, Hodoshima et al. [24] explored the effect of steady-state suppression under various reverberant conditions. In order to simulate reverberant conditions in small to medium-sized halls, stimuli were convolved with various artificial impulse responses with RTs of 0.4-1.3 s.

Stimuli. The original speech samples consisted of 24 nonsense consonant-vowel (CV) syllables embedded in a Japanese carrier phrase. The vowels were /a, i/, and the consonants were /p, t, k, b, d, g, s, ʃ, h, dz, dʒ, tʃ, m, n/. The speech samples were obtained from the ATR speech database of Japanese. The eight different reverberant environments were simulated by taking a convolution of speech samples with eight impulse responses with RTs of 0.4, 0.5, 0.7, 0.9, 1.0, 1.1, 1.2 and 1.3 s. The impulse responses were created from the sample with an RT of 1.1 s (measured in Hamming Hall, Tokyo, Japan) by multiplying an exponential decay. Stimuli were the original signals with reverberation (Unprocessed) and the processed signals with reverberation (Processed).

Subjects. Twenty-four young normal-hearing subjects (14 males and 10 females, aged 18 to 26 years) participated in Experiment 1a. Twenty-four young normal-hearing subjects (11 males and 11 females, aged 19 to 27 years) participated in Experiment 1b. All were native Japanese speakers.

Procedure. The experiment was conducted in a soundproof room. Stimuli were presented diotically through headphones (STAX SR-303) connected to a computer. The sound level was adjusted to each subject's comfort level during the training session prior to the experiment. A stimulus was presented in each trial and the subjects were instructed to select one of the 24 CVs displayed on the computer in Kana orthography. The experiment was carried out at each subject's pace. For each subject, 240 stimuli were presented randomly (5 impulse responses with RTs of 0.9, 1.0, 1.1, 1.2 and 1.3 s x 24 CVs x 2 processing conditions) in Experiment 1a. In Experiment 1b, 288 stimuli were presented randomly (6 impulse responses with RTs of 0, 0.4, 0.5, 0.7, 0.9 and 1.0 s x 24 CVs x 2 processing conditions). A dry condition in Experiment 1b is not discussed in this paper.

Results. Figure 1 shows the two experimental results. In Experiment 1a, a 2 x 5 ANOVA for repeated measures with processing and impulse response confirmed the significant main effects of processing [$p < 0.01$] and the impulse response [$p < 0.01$]. The interaction effect was also significant [$p = 0.02$]. For the comparison of the mean percent correct values between with and without processing conditions, t-tests were performed for each impulse response. A statistically significant difference was obtained at RT of 1.0 s [p=0.02], 1.1 s [p<0.01] and 1.2 s [p<0.01]. In Experiment 1b, a 2 x 5 ANOVA for repeated measures confirmed the significant main effects of processing [p<0.01] and impulse response [p<0.01]. The interaction effect was also significant [p=0.006]. For comparison of the mean percent correct values obtained with and without processing, t-tests were performed for each impulse response. A statistically significant difference was obtained at RT of 0.7 s [p<0.01] and 0.9 s [p<0.01].

Discussion. The results showed that statistically significant improvements were obtained by steady-state suppression for RTs 0.7-1.2 s. This demonstrates that steady-state suppression is a pre-processing method effective for improving syllable identification because it reduces the effect of overlap-masking under specific reverberant conditions.

EXPERIMENT 2: FOR ELDERLY LISTENERS

In this experiment, Miyauchi et al. [25] evaluated the effect of steady-state suppression with fifty elderly people. They conducted a listening test under the same simulated reverberant conditions as in Experiment 1.

Stimuli. The speech samples from Experiment 1 were also used in Experiment 2, except for the target vowel (only /a/ was used in Experiment 2). The two different reverberant environments were simulated by taking a convolution of speech samples with two impulse responses. RTs of the impulse responses were 1.0 s and 1.3 s (measured at the largest lecture hall at Sophia University, Tokyo, Japan). The impulse response with an RT of 1.0 s was the same as the one with an RT of 1.0 s in Experiment 1. Stimuli were the original signals with reverberation (Unprocessed), and the processed signals with reverberation (Processed).

Subjects. Fifty elderly subjects (21 males and 29 females, aged 56 to 90 years) participated in Experiment 2. All were native speakers of Japanese. Sixteen people were classified as presbycusis and 22 people were classified as normal hearing based on their audiograms.

Procedure. The procedure was the same as in Experiment 1, except for the number of CVs displayed on the computer (14 CVs in Experiment 2), a reply from subjects (subjects wrote their answers on answer sheets in Experiment 2) and the total number of stimuli (56 in Experiment 2). The stimuli were presented randomly (2 reverberant conditions x 14 CVs x 2 processing conditions).

Results. Table 1 shows the results of Experiment 2. A t-test indicated that the correct rate of the processed signal was significantly higher than that of the unprocessed signal in each reverberant environment [$p<0.01$].

Discussion. The results confirm that steady-state suppression is effective for not only young people as shown in the previous experiment but also elderly people in specific reverberant conditions. Comparing the results of Experiments 1 and 2, we see that the degree of improvement in speech intelligibility for elderly people was greater than for young people.

EXPERIMENT 3: FOR NON-NATIVE LISTENERS

In this experiment, Hodoshima et al. [26] investigated whether steady-state suppression improved consonant identification for non-native listeners under the same simulated reverberant environment as Experiment 1. This experiment also compared the effect of steady-state suppression on consonant identification by native and non-native listeners.

Stimuli. The same version of the Modified Rhyme Test (MRT) used in Kusumoto et al. [13] was used in this experiment. The target English words developed by Kruel et al. [28] were embedded within a carrier phrase. All 6 lists, each composed of 50 monosyllabic words, were used. The three different reverberant environments were simulated by taking the convolution of speech samples with the three impulse responses used in Experiment 1 (RTs of the impulse responses were 0.4, 0.7 and 1.1 s). Stimuli were the original signals with reverberation (Org) and the processed signals with reverberation (Proc).

Subjects. Twenty-four native speakers of English (14 males and 10 females, aged 18 to 50 years) as native-listeners and twenty-four native speakers of Japanese (6 males and 18 females, aged 20 to 30 years) as non-native listeners participated in the experiment. All were normal hearing people.

Procedure. Each subject was tested with all six MRT lists. The lists were assigned to each of the six conditions (2 processing conditions x 3 reverberant conditions) and counterbalanced across subjects. Additionally, 50 sentences randomly selected from across the six lists were used as stimuli in a dry condition. The procedure was the same as in Experiment 1, except for the places the experiment was conducted for native listeners (the stimuli were presented through headphones, AKG K271, in Experiment 3), words displayed on the computer (six words rhyming with the target word in Experiment 3) and the total number of stimuli (350 in Experiment 3). The stimuli were presented randomly (3 reverberation conditions x 14 CVs x 7 processing conditions). The 300 reverberant stimuli were randomly presented first, followed by the 50 randomly presented stimuli in the dry condition. The dry condition is not discussed in this paper.

Results. Figure 2 shows the experimental results with initial and final word position shown separately. Open squares represent native listeners' mean percent correct for the unprocessed stimuli (nl_org). Filled squares represent native listeners' mean percent correct for the steady-state suppressed stimuli (nl_proc). Open circles represent non-native listeners' mean percent correct for the unprocessed stimuli (nnl_org). Filled circles represent non-native listeners' mean percent correct for the steady-state suppressed stimuli (nnl_proc). A 2 x 3 x 2 mixed ANOVA was carried out with listener group as a nonrepeated factor, RT, consonant position, and processing as repeated variables, and percent correct as the dependent variable. Results show that native listeners had a higher percent correct than nonnative listeners [$p < 0.01$]. The percent correct was also reliably higher for the 0.4 s RT than the 1.1 s RT [$p < 0.01$], and higher for initial consonants than for final consonants [$p < 0.01$]. No reliable difference in percent correct was observed between the steady-state and unprocessed conditions. In addition to these main effects, significant interactions were observed between consonant position and listener group [$p < 0.01$] and between consonant position and reverberant condition [$p < 0.01$]. Other interactions were not significant.

Discussion. The results showed that native listeners performed better than non-native listeners in all conditions used in Experiment 3. The results also showed that the mean percent correct decreased as RT increased, and was higher in initial consonants than in final consonants. In addition, the difference between the mean percent correct for initial and final consonants was larger for non-native listeners than for native listeners in reverberant conditions. Although there were no significant differences between unprocessed and steady-state suppressed stimuli, and no significant interaction between the effect of the steady-state suppression and listener group under the reverberant conditions used in the current study, the effect of steady-state suppression differed in consonant position, RT and listener group. These findings imply that a pre-processing technique would be required which helps non-native listeners to identify consonants as well as native listeners do.

EXPERIMENT 4: WITH SPEECH-RATE SLOWING

In this experiment, Arai [23] and Nakata et al. [27] applied steady-state suppression after slowing the speech rate of a signal. To decrease the speaking rate, Praat [29] was used, which applies the PSOLA (Pitch-Synchronous Overlap and Add) method for time-scale modification. The PSOLA method can modify speech rate without changing the fundamental or formant frequencies of the original speech signal [30].

Stimuli. The speech samples from Experiment 2 were also used in Experiment 4. The three different reverberant environments were simulated by taking the convolution of speech samples with three impulse responses. RTs of the three impulse responses were 2.0 s (Rev1), 2.8 s (Rev2; measured at a lecture hall from the database by the Sub Working Group on Research in Speech Transmission Quality of the Architectural Institute of Japan) and 3.6 s (Rev3; measured at St. Ignatius Church, Tokyo, Japan). Rev1 was created from Rev2 by multiplying an exponential decay as in Experiment 1. Stimuli were original speech samples (Unprocessed), speech samples processed by speech-rate slowing (SRS) and speech samples processed by steady-state suppression after speech-rate slowing (SRS+SSS).

Subjects. Twenty-four young normal-hearing subjects (8 males and 16 females, aged 19 to 24 years) participated in the experiment. All were native Japanese speakers.

Procedure. The procedure was the same as in Experiment 1, except for the number of CVs displayed on the computer (14 CVs in Experiment 4) and the total number of stimuli (294 in Experiment 4). The stimuli were presented randomly (3 reverberation conditions x 14 CVs x 7 processing conditions). Only three processing conditions are discussed in this paper, although the total number of processing conditions was seven, including the four other processing conditions based on the zero-padding technique proposed in [23].

Results. Table 2 shows the experimental results. Multiple comparison of the three out of seven processing conditions showed significant differences between Original and SRS+SSS [$p < 0.05$] at RT of 2.8 s, Original and SRS [$p < 0.05$] at RT of 2.0 s, and Original and SRS+SSS [$p < 0.01$] at RT of 2.0 s. Also, the difference between SRS and SRS+SSS was significant [$p < 0.05$] at RT of 2.0 s.

Discussion. Steady-state suppression after speech-rate slowing yielded superior speech intelligibility as compared to the simple speech-rate slowing approach (at RT of 2.0 s, the correct rate of SRS+SSS was significantly higher than that of SRS). This result confirmed that overlap-masking was effectively reduced by suppressing the steady-state portions of speech while slowing down the speech rate.

CONCLUSIONS

In this paper, we discussed the effectiveness of various aspects of the steady-state suppression technique, especially the effect of various reverberant conditions, the effect on the speech intelligibility for various types of listeners, and the effect of speech-rate slowing. As results, we can conclude that properly designing a speech amplification system of a room is definitely effective in terms of preserving the intelligibility of speech for barrier-free speech communication; and further, it should be combined with other aspects of design of room acoustics for universal environment.

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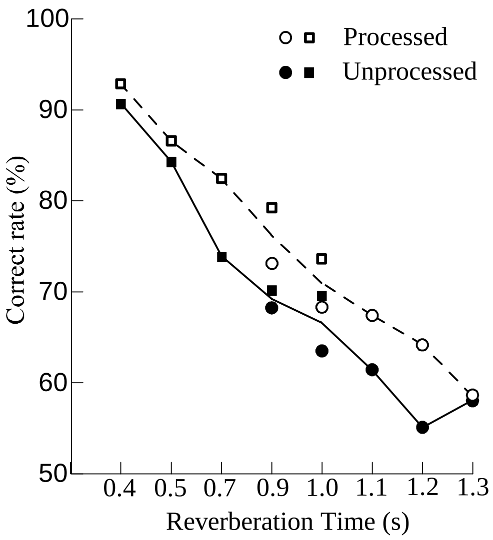

Figure 1. Results of Experiment 1 (from [24])

Table 1. Results of Experiment 2 (from [25])

RT [s]	Unprocessed	Processed
1.1	34.4	45.8
1.3	37.3	44.9

Table 2. Results of Experiment 4 (from [27])

RT [s]	Unprocessed	SRS	SRS+SSS
2.0	45.2	57.7	70.2
2.8	43.5	50.3	56.0
3.6	44.6	42.6	50.9

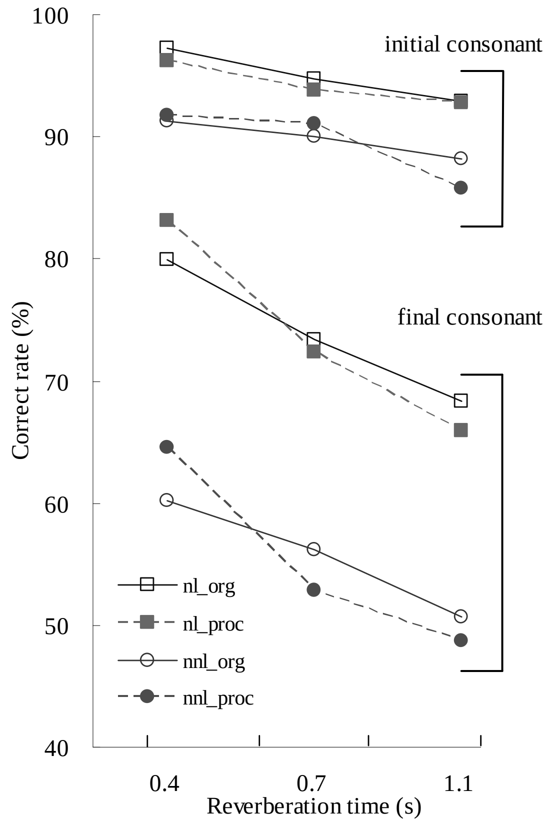

Figure 2. Result of Experiment 3 (from [26])