

Research Note

Evaluation of a Telephone Speech-Enhancement Algorithm Among Older Adults With Hearing Loss

Christina M. Roup,^a Gayla L. Poling,^{a,b} Kimberly Harhager,^{a,c}
Ashok Krishnamurthy,^a and Lawrence L. Feth^a

Purpose: In this study, the authors evaluated a processing algorithm aimed at improving speech recognition via the telephone among older adults with sensorineural hearing loss (SNHL).

Method: Thirty older adults with SNHL participated. Speech recognition was measured in quiet using the Modified Rhyme Test (MRT; Kreul et al., 1968) and the Speech Perception in Noise (SPIN; Bilger et al., 1984) sentences, and in noise using the Quick Speech in Noise (QSIN; Killion et al., 2004) test. Each test was presented via the telephone with and without processing.

Results: Significant improvements in recognition performance due to processing were observed for the SPIN and QSIN. The improvement on the QSIN was significantly greater than on the MRT and SPIN, likely because the MRT and SPIN sentences were

presented in quiet, whereas the QSIN was presented in noise. Significant improvements in recognition performance were observed for both an offline version and a real-time version of the algorithm relative to the unprocessed condition, although no difference was noted between the 2 versions.

Conclusions: Results indicate that preprocessing the acoustic signal is a viable method of improving speech recognition via the telephone. The algorithm has the potential to benefit older adults with SNHL who struggle to communicate via the telephone with or without hearing aids.

Key Words: hearing impairment, speech recognition, telephone

For older adults, communication via the telephone may be complicated by reduced audibility of the telephone signal due to hearing loss (Cruikshanks et al., 1998). This complication can be further exacerbated by two limitations inherent to telephone communication itself: a limited acoustic signal from the telephone and total reliance on the auditory signal due to a lack of visual cues. The frequency response range of a landline telephone is commonly accepted as 300–3300 Hz (Rodman, 2003). The reduced bandwidth of the telephone signal results in missing acoustic energy needed to correctly recognize speech. The dynamic range of the telephone is also

limited in that an amplified acoustic signal will likely be peak clipped, thereby introducing nonlinearity and/or distortion. In addition, communication via the telephone forces the listener to rely solely on the auditory signal without the benefit of visual cues. Considerable improvement in speech recognition occurs when visual speech cues are available to supplement the auditory information (Helfer, 1998; MacLeod & Summerfield, 1987). The lack of visual cues, reduced audibility due to hearing loss, and a limited acoustic signal all interact, causing difficulty understanding speech over the telephone.

Existing options available for improving the audibility of the telephone signal for listeners with hearing impairment (HI; e.g., amplified telephones, telephones with a volume control, hearing aids with and without telecoils) are dependent on their use by the individual with HI. Another option is to preprocess the acoustic signal before it reaches the listener with HI. The option of amplifying the telephone signal through preprocessing is particularly relevant for community service agencies (e.g., community assistance programs) and medical organizations (e.g., telehealth) that experience difficulty communicating with older adults with HI. The idea of

^aThe Ohio State University, Columbus, OH

^bNorthwestern University, Evanston, IL

^cNationwide Children's Hospital, Columbus, OH

Correspondence to Christina M. Roup: roup.2@osu.edu

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preprocessing a telephone signal is not new. For example, Terry et al. (1992) used digital signal processing to provide frequency shaping and compression to improve the audibility and intelligibility of the telephone signal for listeners with HI. Results revealed significant improvements in speech intelligibility for the frequency shaping with and without compression compared with the control condition. Terry et al. also obtained preference judgments regarding processing strategies presented through the telephone. The majority of subjects preferred a form of preprocessing over linear amplification. The results from Terry et al. suggest that preprocessing of the telephone signal is a viable method of improving speech recognition for listeners with HI.

The Departments of Speech and Hearing Science and Electrical and Computer Engineering at The Ohio State University have collaborated to develop a computer-based telephone speech-enhancement algorithm (TSEA; Komattil, 2004; Natarajan, 2002; Poling, 2004; Sheffield, 2000). The TSEA is based on a hearing aid compression-processing algorithm aimed at preserving spectral contrast within the speech signal (Tejero-Calado, Rutledge, & Nelson, 2001) that was chosen, in part, to amplify the signal without introducing peak clipping. Motivation for development of the TSEA stems from the need to improve telephone communication between community service agencies (e.g., community assistance programs, telehealth) and older adults with HI. In a community assistance program situation, the TSEA is intended to be activated on the “counselor end” in order to assist with speech recognition on the “client end.” The TSEA can also be implemented in this manner within the emerging telehealth domain, which has become increasingly utilized over the past decade (Koch, 2006). Difficulty communicating with health care professionals via the telephone has been reported as a concern by older adults (Iezzoni, O’Day, Killeen, & Harker, 2004). The ability of a health care professional to enhance the speech signal via the telephone has the potential to greatly improve communication with older adults with HI. The purpose of the present study, therefore, was to evaluate the effectiveness of the TSEA at improving speech recognition of older adult listeners with HI via the telephone in a group of older adults with sensorineural hearing loss (SNHL). A secondary purpose of the study was to compare telephone recognition performance between the original offline version of the algorithm and an updated version that can be run in real time.

Method

Subjects

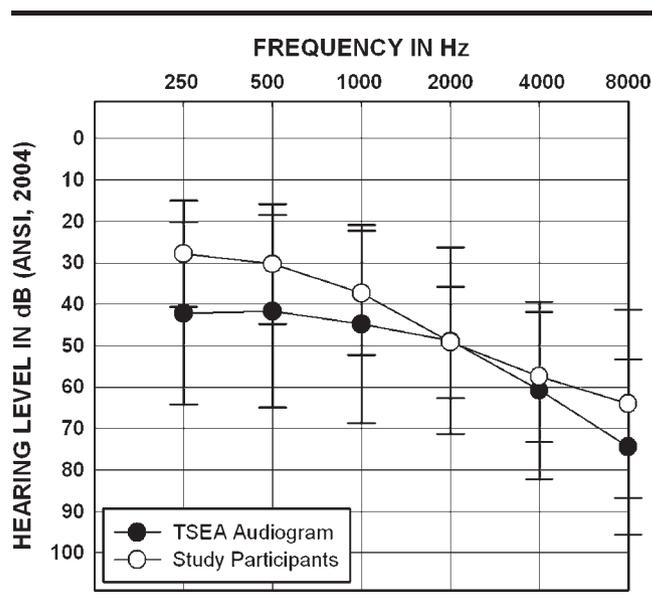
Thirty older adults (16 female, 14 male), ranging in age from 55 to 70 years, participated in the present

study. Twenty of the subjects (11 female, 9 male) participated in Experiment 1, and all 30 participated in Experiment 2. Inclusion criteria included (a) a bilateral, moderate to severe SNHL (i.e., air and bone conduction thresholds within 10 dB at 250–4000 Hz); (b) normal otoscopic findings; (c) tympanometry within normal limits (Wiley et al., 1996); and (d) English as a first language. Figure 1 presents mean pure-tone thresholds for all subjects. All subjects were recruited from The Ohio State University Speech-Language-Hearing Clinic research database and employee e-mail advertisements. The present study was approved by the Behavioral and Social Sciences Institutional Review Board at The Ohio State University.

TSEA

The TSEA is based on a hearing aid compression-processing algorithm (Tejero-Calado et al., 2001) adapted by Natarajan (2002) for the present study. The TSEA is a multichannel compression algorithm aimed at preserving spectral contrast within the speech signal. The processing reduces the typical amplitude variations in speech, allowing relatively low-amplitude consonant sounds to be amplified to audible levels while higher amplitude vowel sounds remain comfortably loud. It also preserves the spectral contrast by maintaining spectral peak-to-valley ratios between loud and soft variations in speech. Specifically, the speech signal sampled at 8000 Hz is split into frames of 32 ms with 50% overlap between the frames. Each frame is then passed through

Figure 1. Mean pure-tone thresholds for all study participants (open circles) and the pure-tone thresholds used in the telephone speech-enhancement algorithm (TSEA; filled circles). Error bars indicate SD. ANSI = American National Standards Institute.



a Hamming window, and a 512-point fast Fourier transform is used to compute the spectral content of each frame. Further processing of the frame is done only if the spectral content of the frame is above a noise threshold.

Each non-noise frame is divided into three channels on the basis of its spectral content. The gain applied to each channel is computed independently from the input audiogram and the frequency response characteristics of the telephone¹ receiver so as to raise the signal level above the threshold of audibility while maintaining the spectral peak-to-valley ratio. In a given channel, let D_{NH} and D_{HI} be the dynamic ranges of a listener with normal hearing (NH) and a listener with HI, respectively. Assume that the relative threshold of the listener with HI is T and that the relative SPL in the channel before gain is S_N . Gain (g) is computed as $g = D_{HI}/D_{NH}$, and the relative sensation level (SL) is computed as $SL_{HI} = g \times SL_{NH} + T$. The determination of gain by the algorithm is further illustrated in Figure 2. Specifically, the unprocessed (original) spectrum is shown as a dashed curve, and the processed spectrum is shown as a solid curve. The threshold of audibility for the average hearing loss implemented by the algorithm is depicted as a solid line. Point A in Figure 2 at approximately 2000 Hz would be audible for a listener with normal thresholds, but it falls below the threshold for the average listener with HI. At 2000 Hz, the algorithm calculates a ratio of the dynamic range of the HI ear (D_{HI}) to the dynamic range of the NH ear (D_{NH}). Then, the level of A is expressed in dB SL (above normal threshold). The algorithm determines the desired SL for the HI ear as $SL_{HI}/SL_{NH} = D_{HI}/D_{NH}$. To determine the gain to be applied to level A, the value of A is multiplied by D_{HI}/D_{NH} and is added to the threshold value at 2000 Hz. Thus, $A^* = (D_{HI}/D_{NH}) \times A + \text{threshold}$.

To avoid any audible distortion due to discontinuities in the signal, the gain variation between adjacent frames is smoothed. The processed speech is synthesized by performing an inverse fast Fourier transform, and the individual frames are recombined by an overlap add method to smooth the discontinuities at the frame boundaries. Compression characteristics of the TSEA included (a) an attack time of 5 ms with a release time of 100 ms and (b) a compression threshold of 50 dB SPL. There is no fixed compression ratio in the TSEA. Rather, the compression ratio is determined frame by frame on the basis of the input signal.

For the purposes of the present study, average hearing thresholds were used to define the gain within each channel of the algorithm. The intended purpose of the TSEA is to enhance the speech signal before it reaches the listener with HI, and the device would therefore be activated by a service provider. In this scenario, the service provider would have no prior knowledge of the individual's audiogram. Therefore, the average audiogram was based on hearing thresholds (250–8000 Hz) compiled from 100 individuals over the age of 60 years tested at the Columbus Speech and Hearing Center (Columbus, OH) between 1996 and 2000 (see Figure 1).

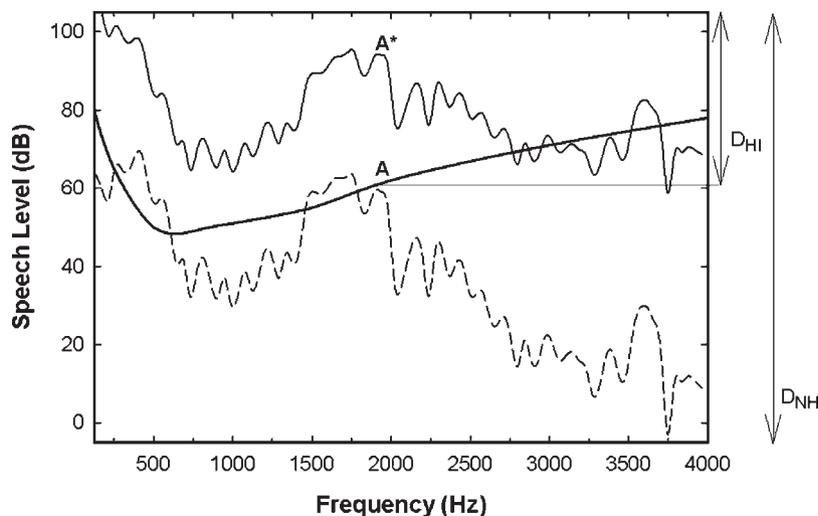
There are two versions of the TSEA: (a) Version 1, a preliminary version coded in MATLAB that cannot be run in real time, and (b) Version 2, written in C code to run on a real-time signal processing server platform developed by FutureCom Technologies. Specifically, the real-time signal processing system was an implementation of the algorithm on telephone interface cards as part of a commercial interactive voice response platform.

Stimuli

Speech recognition was measured in each of the following categories: (a) phoneme discrimination, (b) word recognition, and (c) sentence recognition. Phoneme discrimination was measured with the Modified Rhyme Test (MRT; Kreul et al., 1968). The MRT is a closed-set, 50-item test, with each item consisting of a set of six monosyllabic words that vary in either the initial or final consonant. Word recognition was measured with the revised Speech Perception in Noise (SPIN) test sentences (Bilger, Nuetzel, Rabinowitz, & Rzeczkowski, 1984). The SPIN includes eight lists of 50 sentences, with each list divided into 25 high-predictability and 25 low-predictability sentences. Although the SPIN was designed to be given in the presence of background noise, the sentences were given in quiet for this experiment. In addition, for the purposes of the present study, the number of correct responses was collapsed across SPIN sentence type to determine an overall percent correct. And finally, sentence recognition was measured with the Quick Speech-in-Noise (QSIN) test (Killion, Niquette, Gudmundsen, Revit, & Banerjee, 2004). The QSIN is a test of speech recognition in background noise (i.e., four-talker babble). Multiple signal-to-noise ratios (SNRs) are used to determine a measure of SNR loss. Specifically, six sentences are presented, each at a different SNR. The SNRs decrease in 5-dB steps from 25 dB to 0 dB for each successive sentence. Listeners are asked to repeat the entire sentence, and five key words are scored. For the purposes of the present study, the number of correct responses was collapsed across all SNRs to determine an overall percent correct.

¹The frequency characteristics of the telephone network used in the present study were measured using a type 2 frequency response in which the input varied in order to achieve a constant output. The output was set to the maximum possible before clipping occurred. The frequency response characteristics of the telephone network were incorporated into the algorithm with the average audiogram in order to shape the output.

Figure 2. Illustration of the compression algorithm depicting the original (dashed curve) and processed (solid curve) spectrum of a speech frame. The solid line depicts the audibility thresholds for the average hearing loss used by the algorithm. The dynamic ranges of normal hearing (D_{NH}) and average hearing impairment (D_{HI}) are illustrated on the y -axis. Point A at approximately 2000 Hz represents an audible signal for a listener with normal hearing, and point A* represents the same point amplified by the algorithm.



Recordings of each speech test (MRT, SPIN, and QSIN) were obtained from commercially available CDs. The audio files for each test were extracted from their respective CDs, digitized at a sampling rate of 44.1 kHz (quantization bit rate of 16), and down-sampled to 8 kHz to be compatible with the TSEA. The stimuli were then stored on a desktop PC hard drive.

Procedure

For Experiment 1, speech recognition was measured for the three tests (MRT, SPIN, and QSIN) under two listening conditions—unprocessed and processed—with the offline version of the TSEA. The order of test delivery (type of test) was randomized, and the listening conditions (unprocessed vs. processed) were counterbalanced to minimize order effects. The MRT and the SPIN test were presented in quiet, with 50 trials per condition. The QSIN was presented as intended, in a multitalker babble with six sentences (30 key words) per condition. For Experiment 2, speech recognition was measured using the QSIN under three listening conditions: unprocessed, processed with Version 1 (original), and processed with Version 2 (real time). For Experiments 1 and 2, the QSIN sentences were processed through the algorithm, and the multitalker babble was mixed with the speech after processing. The decision not to process the multitalker babble was based on the assumption of noise present on the HI listener's end. The multitalker

babble was attenuated manually (PA4 attenuator, Tucker Davis Technologies) between each sentence to create the various SNRs (25–0 dB). All speech stimuli as well as the multitalker babble for the QSIN were presented (a) through an Audigy 2NX sound card (Creative Technology) from a desktop PC that was coupled to a portable telephone interface (Microtel, Gentner Communications Corporation); (b) through a telephone landline; and (c) to a telephone receiver (Model T-905C, Cortel). Participants were seated in a sound-treated booth (Model 402ATR, IAC) and were familiarized with each test before administration. All stimuli were presented to the ear that the listener used for everyday telephone communication. Subjects responded verbally to the speech stimuli, and their responses were recorded as percent correct or incorrect. All speech-recognition testing was completed unaided.

The unprocessed speech stimuli were digitized at the 8-kHz sampling rate to match the sampling rate used on the telephone network. To maximize waveform fidelity and minimize quantization errors, the sample amplitudes were normalized. For each sample, the peak value of the amplitude was rescaled so that it was just less than the maximum output available from the sound card. This resulted in an average presentation level of 70–74 dB SPL. The presentation levels were monitored acoustically from the telephone receiver placed directly over the external ear canal on the pinna of a Knowles electronics manikin for acoustic

research (KEMAR) and held in place with adhesive strapping. The compressive processing of these speech samples reduced the peak factor of each waveform. Thus, when the processed waveform was normalized before being delivered to the telephone network, the average level measured was 82–85 dB SPL.

Results

Experiment 1. Table 1 includes *Ms* and *SDs* for recognition performance for the three speech tests (MRT, SPIN, and QSIN) across the two listening conditions (unprocessed and processed). As seen in Table 1, small improvements in telephone speech recognition due to TSEA processing were seen for the phoneme-discrimination task (MRT) and the word-recognition task (SPIN). The greatest improvement occurred for the sentence-based speech-in-noise task (QSIN).

The percentage data were transformed into rationalized arcsine units before statistical analysis (Studebaker, 1985). To determine whether the TSEA resulted in a significant ($p < .05$) improvement in telephone speech recognition, the transformed data were subjected to a two-way repeated measures analysis of variance (ANOVA) with processing condition and speech type as within-subjects factors. Results revealed a significant main effect for processing condition, $F(1, 19) = 100.7, p < .05$. Post hoc Bonferroni *t* tests revealed significant improvements in telephone speech recognition for the SPIN and the QSIN due to the processing algorithm. The difference in phoneme discrimination between the unprocessed and processed conditions for the MRT was not significant. The ANOVA also revealed a significant main effect for speech

type, $F(2, 38) = 144.5, p < .05$. Post hoc Bonferroni *t* tests revealed significantly better processed recognition performance for both the MRT and SPIN as compared with the QSIN. The difference in processed performance between the MRT and SPIN was not significant.

Mean difference scores (processed vs. unprocessed) were also calculated (see Table 1) and compared across the three speech types using a one-way ANOVA. Results revealed a significant main effect, $F(2, 57) = 24.9, p < .05$. Tukey's honestly significant difference post hoc analysis revealed significantly more improvement in QSIN performance relative to both MRT and SPIN performance. No significant differences were found in improvement between the MRT and SPIN.

Experiment 2. Table 2 includes means and *SDs* for the QSIN across the three listening conditions: unprocessed, processed with Version 1, and processed with Version 2. As can be seen in Table 2, both versions of the TSEA resulted in improved sentence recognition as compared with the unprocessed condition. The difference in speech recognition performance between the algorithm versions, however, was minimal (63.8% and 69.0%, respectively).

In order to determine whether the improvement in sentence recognition differed across conditions, the arcsine-transformed (Studebaker, 1985) data were subjected to a one-way ANOVA. Results revealed a significant main effect of condition, $F(2, 87) = 30.4, p < .05$. Tukey's honestly significant difference post hoc analysis on the effect of condition revealed significant ($p < .05$) improvements between the unprocessed condition and the two algorithm versions. Significant differences in recognition performance were not present between the two algorithm versions ($p > .05$).

Table 1. Mean speech recognition performance (in percent) for the three speech tests in Experiment 1 across the two listening conditions: unprocessed and processed.

Condition	MRT (%)	SPIN (%)	QSIN (%)
Unprocessed			
<i>M</i>	80.5	70.3	24.8
<i>SD</i>	8.6	15.9	13.9
Processed			
<i>M</i>	85.2	81.2	56.7
<i>SD</i>	7.7	12.9	21.4
Difference score ^a			
<i>M</i>	4.7	10.9	31.8
<i>SD</i>	8.0	7.2	17.0

Note. MRT = Modified Rhyme Test; SPIN = Speech Perception in Noise test; QSIN = Quick Speech-in-Noise test.

^aDifference (in percent) between the processed and unprocessed scores for each speech measure.

Discussion

The purpose of Experiment 1 was to evaluate the effectiveness of a processing algorithm designed to enhance communication via the telephone. Results of Experiment 1 revealed significant improvements in speech recognition due to processing by the TSEA. Specifically,

Table 2. Mean sentence recognition performance (in percent) for the QSIN in Experiment 2 across the three listening conditions: unprocessed, processed with algorithm Version 1, and processed with algorithm Version 2.

Value	Unprocessed (%)	Version 1 (%)	Version 2 (%)
<i>M</i>	30.9	63.8	69.0
<i>SD</i>	19.9	21.0	19.7

mean performance improved by 10.9% for the SPIN test and 31.9% for the QSIN test. Mean performance improved by 4.7% for the MRT; however, this difference was not significant. Similar results were reported by Terry et al. (1992), who used frequency shaping to improve audibility of a speech signal filtered to mimic the frequency response of a telephone (i.e., 300–3000 Hz). The authors reported improvements in phoneme discrimination on the order of 11.8% for frequency shaping alone and 15.8% for frequency shaping plus compression in a group of listeners with SNHL. The results from the present study and those from Terry et al. (1992) suggest that preprocessing the speech signal is a viable method of improving speech recognition via the telephone.

The amount of improvement observed in the present study was not equal across the levels of speech assessed (i.e., phoneme, word, and sentence). Specifically, the difference in mean improvement due to processing between the MRT and the SPIN test was not significantly different. However, the QSIN test resulted in significantly greater improvement due to processing than either the MRT or the SPIN test. A greater improvement due to processing on the QSIN test is unsurprising given that it was presented in the presence of multitalker babble. The fact that equal amounts of improvement due to processing were not seen across the three levels of speech tested may be a reflection of different levels of test difficulty.

As part of the development of the TSEA, Version 1 tested in Experiment 1 was programmed in C code to run in real time on a signal processing server platform developed by FutureCom Technologies (i.e., Version 2). It was expected that no differences would be observed in speech recognition performance between the two versions of the TSEA. Results of Experiment 2 verified that, indeed, similar levels of improvement were observed for both versions of the TSEA relative to the unprocessed condition. Performance differences between the two algorithm versions were small and not significant. Version 2 of the TSEA, therefore, was as effective at improving speech recognition via the telephone as the original algorithm. Version 2 has the added benefits of real-time implementation and multichannel capabilities. Specifically, the speech signal can be processed as it is presented or spoken. In addition, this real-time system is capable of handling multiple calls simultaneously.

The ability to communicate effectively via the telephone in today's society is vital for many reasons, including avoiding isolation, maintaining independence, and ensuring safety and security (Dimmick, Sikand, & Patterson, 1994; Murphy, 1999). The issue of effective telephone communication takes on even greater importance when considering the evolution of telehealth. Telehealth allows in-home patient care by health care providers, overcoming common issues for older adults such as

difficulty with transportation (Mormer & Mack, 2003; Wakefield, 2003). Many older adults with hearing loss, however, have expressed concerns regarding their ability to effectively communicate about their health care via the telephone (Iezzoni et al., 2004). Options for improving telephone communication currently focus on the HI user's end (i.e., amplified telephones or hearing aids). The success of these strategies, however, depends on the individual with HI having knowledge of and access to such devices. The focus of the present study, therefore, was to examine the efficacy of a system that can be implemented by community service agencies and health care providers that routinely communicate with older adults with HI. The development of a preprocessing algorithm such as the TSEA is an important first step toward the long-term goal of providing improved communication via the telephone for older adults with HI.

Other telecommunication technologies, such as mobile telephones and voice-over-Internet protocols (VoIP), such as Skype, also may potentially benefit from a preprocessing algorithm. Mobile telephone signals can be affected by the same limited bandwidth as landline telephone signals and may therefore benefit from a preprocessing strategy such as the TSEA. VoIP technology, on the other hand, takes advantage of web camera technology, allowing the listener to use lip and facial cues during communication. A preprocessing strategy such as the TSEA may not be as beneficial in this situation. Future research efforts will also be aimed at evaluating the algorithm with additional communication technologies, such as mobile telephones and VoIP.

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