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Effect of Release Time on Preferred Gain and Speech Acoustics

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Abstract

Eighteen experienced hearing aid users with mild to moderate sensorineural hearing loss were fit with a digital hearing instrument. An adaptive procedure was used to determine their preferred gain setting for continuous speech under six conditions. Release time (RT) was set to 40, 160, or 640 msec. A prerecorded speech stimulus was presented in quiet or in the presence of multitalker babble (10 dB signal-to-babble ratio); all other compression variables were fixed. Real-ear data obtained with settings for each condition suggest that RT did not affect gain preference; however, subjects preferred higher gain in the presence of the multitalker babble. The RMS amplitudes of 30 phonemic units were calculated using ear canal recordings of the speech stimulus for each subject in each condition. Altering RT resulted primarily in decreased amplitude with increased RT, but this effect was not predictable across listeners or conditions.

Key Words: Adaptive procedure, compression, compression release time, compression threshold, gain, hearing aid, phoneme amplitude, wide dynamic range compression

Abbreviations: AAI = Aided Articulation Index; AT = attack time; NAL-R = National Acoustic Laboratory—Revised; REIG = real ear insertion gain; RMS = root mean square; RT = release time; VCV = vowel consonant vowel; WDRC = wide dynamic range compression

Sumario

Se le adaptó un auxiliar auditivo digital a dieciocho sujetos con experiencia en el uso de amplificación y con una hipoacusia sensorineural leve a moderada. Se utilizó un procedimiento adaptativo para determinar su configuración preferida de ganancia para escuchar lenguaje continuo bajo seis condiciones. El tiempo de liberación (RT) se estableció a 40, 160 o 640 milisegundos. Se presentó un estímulo lingüístico pre-grabado, tanto en silencio como en presencia de ruido de conversación de hablantes múltiples (tasa de señal-nivel de habla de 10 dB): todas las otras variables de compresión permanecieron fijas. Los datos de oído-real obtenidos con ajustes para cada condición sugieren que el RT no afectó la ganancia preferida; sin embargo, los sujetos prefirieron ganancias mayores en presencia de ruido de conversación con hablantes múltiples. Se calcularon las amplitudes del RMS de 30 unidades fonémicas, utilizando registros en el conducto auditivo de estímulos de lenguaje

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para cada sujeto en cada condición. La alteración del RT produjo, primariamente, una disminución en la amplitud conforme incrementó el RT, pero el efecto no fue predecible en los diferentes sujetos o condiciones.

Palabras Clave: Procedimiento adaptativo, compresión, tiempo de liberación de la compresión, umbral de compresión, ganancia, auxiliar auditivo, amplitud fonémica, compresión de rango dinámico amplio

Abreviaturas: AAI = Indice de Articulación con Amplificación; AT = tiempo de ataque; NAL-R = Laboratorio Nacional de Acústica-Revisado; REIG = Ganancia de Inserción en Oído Real; RMS = raíz media cuadrada; RT = tiempo de liberación; VCV = vocal-consonante-vocal; WDRC = compresión de rango dinámico amplio

Wide dynamic range compression (WDRC) circuits are used widely in the current generation of hearing aids. Their use allows greater audibility of lower-level sounds while not overamplifying higher-level sounds. This has particular advantages for improving the understanding of speech in quiet. However, the specific compression settings needed to insure each individual's optimal performance and acceptance remain unclear. A compressor's attack and release times (AT, RT) will have variable influences on speech and in some cases on a listener's preference. An understanding of the interaction of compression time constants and user gain preference has clinical relevance, especially when fitting hearing aids that have a manual volume control. That is to say, the benefits that certain compression settings have on speech might be overridden by a user seeking his or her preferred volume control setting.

The merits of shorter versus longer time constants in WDRC hearing aids have been studied in several investigations. Syllabic compression utilizes time constants (AT < 10 msec; RT < 50 msec) short enough to respond to moment-to-moment changes in amplitude within a word. It is intended to increase the audibility of lower-level, high-frequency consonants (Van Tasell, 1993; Dillon, 1996; Moore, 1996; Souza and Turner, 1996). Although the goal of improved audibility is of theoretical importance, the benefit of syllabic compression for improving speech understanding has not been demonstrated uniformly across listening conditions. Several investigations using a syllabic compressor that also incorporated a signal delay to minimize overshoots found significant improvements over linear amplification in speech recognition in quiet, particularly at a compression ratio of 2:1 (Verschuure et al, 1993, 1996, 1998). In noise, subjects with poor unaided speech discrimination performed better with the syllabic compressor; subjects with relatively better unaided speech discrimination improved with linear amplification. Stelmachowicz et al (1995) showed that syllabic compression, compared to linear amplification, improved audibility of speech sounds but only significantly improved speech recognition in one of their three subjects.

In contrast to syllabic compression, which affects the temporal amplitude-modulation envelope of a speech signal (Van Tasell, 1993), WDRC with longer time constants (AT > 100 msec; RT > 400 msec) maintains the intensity relationships between speech sounds (Kuk, 1998; Dillon, 2001). In a related article (Ellison et al, 2003), we discussed these effects in detail. A summary is reproduced here. Several investigators (Gordon-Salant, 1986, 1987; Freyman and Nerbonne, 1989; Preves et al, 1991; Balakrishnan et al, 1996; Souza and Turner, 1996; Hickson and Byrne, 1997; Sammeth Dorman and Stearns, 1999; Smith and Levitt, 1999; Souza, 2000) have demonstrated how the intelligibility of speech units (i.e., items no longer than a word) is affected by the amplitude envelope or consonant-to-vowel ratio of speech; however, results have been conflicting. The outcomes are less variable when longer duration stimuli are used. Souza and Kitch (2001a) determined that when compression ratios were increased from 1:1 to 2:1 and from 2:1 to 5:1, word recognition scores for sentence materials became poorer; there was a negative correlation between performance and the magnitude of amplitude envelope alteration. Van Tasell and Trine (1996) reported the degree to which the amplitude envelope contributes to recognition of vowel-consonant-vowel (VCV) disyllables and sentences in normal-hearing subjects. Subjects were able to classify disyllabic stimuli on the basis of compressed envelope cues after spectral and periodicity cues were removed. Alternatively, compression of envelope information and not the removal of periodicity information adversely affected subject performance for sentence recognition after spectral information had been removed. The authors argued that single-channel instruments with a high compression ratio and short attack and release times should not be fit to persons that have difficulty extracting spectral cues from speech.

At its essence, the issue of release time selection revolves around the question of which is more important for speech recognition, improved audibility (as with syllabic compression) or well-maintained temporal characteristics (as with WDRC with slower time constants)? However, patients' perceptions of the resulting speech signal are also important and influence their preferences. Bentler and Nelson (1997), comparing release times ranging from 20 to 500 msec, found that subjects did not have a significant preference for one release time condition over another. By contrast, studies comparing even longer release times did show a significant effect. Hansen (2002) found that subjects preferred a release time of 4 sec over two shorter release time conditions (400 msec and 40 msec) when listening through a 15-channel WDRC instrument and asked to rate quality and intelligibility. Neuman et al (1998) found that, with hearing-impaired individuals listening to continuous discourse, ratings of pleasantness increased while ratings of overall loudness and loudness of background noise decreased with increasing release times from 60 to 1000 msec.

Because changes in compression time constants can alter the temporal and spectral characteristics of speech and listeners' preferences, and because many hearing aids have a manual volume control (e.g., Kochkin, 2002 reported that 88% of hearing aids have volume controls, and that 48% of new hearing aid users would prefer to have a volume control), there is a need to investigate how these variables interact. That is to say, if a hearing aid's compression is set to shorter versus longer release times, will users adjust the volume control to compensate for their concomitant changes in perception, possibly negating the desired goal? Most studies that have evaluated the effects of compression have used smaller speech segments such as syllables or words in isolation. Compression circuits may respond differently during continuous speech than they do to isolated words separated by silence. Our study addressed two questions concerning the interaction of compression release times and user gain preferences for a continuous speech stimulus:

1. Do hearing-impaired subjects' overall gain preferences, when listening to continuous speech in quiet and noise, change as a function of changes in release time?

2. How do changes in preferred gain and release time affect the amplitude of selected elements of a continuous speech stimulus?

METHOD

Subjects

Eighteen subjects (ten women, eight men) were recruited from the University of Arizona Hearing Clinic and the Adobe Hearing Center in Tucson, Arizona, and were paid for their participation. All were adults, aged 36-88 years (mean = 68.4 years) who had worn a hearing aid in the test ear for at least one year before their entry into the study. An acoustic immittance screening was done on all subjects using a GSI-33 Middle Ear Analyzer, and the results were within normal limits for

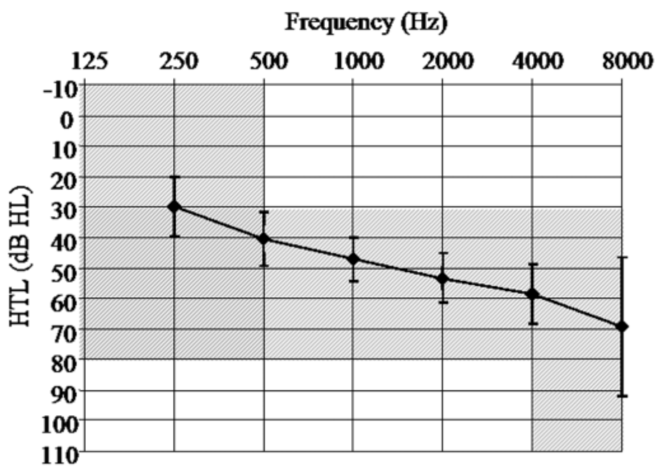

Figure 1. Average pure-tone thresholds of the test ears (n = 18). Error bars represent +/- 1 SD. The gray area represents subject inclusion criteria.

adults. Pure-tone air-conduction thresholds were obtained at 250, 500, 1000, 2000, 3000, 4000, 6000, and 8000 Hz using a Madsen Aurical Audiometer (calibrated to ANSI S3.6, 1996a). Only one ear was used for testing, resulting in 11 left and 7 right ears. Ear selection was based on the best fit to the inclusion criteria; if equal, ear selection was pseudorandom to counterbalance left and right ears. Figure 1 displays the mean audiometric configuration for the test ears of all subjects. The shaded area indicates the audiometric inclusion criteria. Bone-conduction testing confirmed sensorineural hearing loss for all subjects. Word-recognition ability was assessed using recorded BYU-25 word lists (Harris and Hilton, 1998). All subjects had a PB-Max greater than 60% in the test ear.

Hearing Aid

An Oticon Digifocus II Power behind-the-ear digital hearing aid was used because its release time can be modified. It contains two channels with three frequency bands in the low-frequency channel and four frequency bands in the high-frequency channel (cutoff frequency = 1500 Hz). Other than release time and gain, all parameters of the hearing aid remained fixed across measurement conditions. The hearing aid was fit to each subject using either his or her personal earmold or an EAR temporary earmold if a personal earmold was unavailable. Venting was adjusted to insure that there was no feedback at any point during data acquisition. The attack time and compression ratio were set to 5 msec and 3:1, respectively, and remained fixed in both channels in all conditions. The frequency response of the hearing aid was adjusted using a Madsen Aurical Real Ear probe microphone system set to a National Acoustic Laboratory—Revised (NAL-R) target (Byrne and Dillon, 1986) using a 65 dB SPL stimulus level. Variations in earmold acoustics were compensated for in this process using the available 7-band gain adjustment. A nonlinear formula was not chosen given our intention of maintaining a fixed compression ratio of 3:1 for all subjects. The output saturation sound pressure level was set to its highest possible setting and remained unchanged throughout the experiment. The compression release time was adjusted to either 40, 160, or 640 msec. The overall gain was adjusted in 2 dB steps by varying both the "gain" and "loud" settings in the Otiset V4.2 software. Quantitative changes in gain were verified prior to the experiment using the Hearing Instrument Test Module of the Madsen Aurical. This method of gain adjustment did not alter the shape of the frequency response of the hearing instrument.

Speech and Noise Stimuli

Running speech was selected to measure the effect of compression on a dynamic signal. The prerecorded stimulus consisted of three sentences:

(1) Peter will keep at the tall chilly peak.

(pitə˞ wɪl kip æt θʌ tɔl tʃɪlɪ pik)

(2) The blue spot is on the thin key again.

(θʌ blu spɑd ɪz ɑn ðʌ θɪn ki əgɛn)

(3) When he comes home we'll feed him fish.

(wɛn hi kʌms hom wɪl fid hɪm fɪʃ)

As we discussed in Ellison et al (2003), these sentences were chosen for three reasons. First, they include most phonemes found in American English, including all voiceless consonants and all vowels except diphthongs and /e, 3, U, 3^, a/. Second, vowels and voiceless consonants are represented in the initial, medial, and final position. Third, these sentences represent continuous speech that a listener might encounter in a real-world situation. The speech was presented at 65 dB SPL.

The noise stimulus was BYU recorded multitalker babble (Harris and Hilton, 1998) presented at 50 dB SPL (+15 SNR) at a 0 degree azimuth. This level was selected to coincide with the hearing instrument's compression threshold so that near-maximum gain would be attributed to the noise in the absence of a competing speech signal.

Instrumentation and Calibration

All data recordings were made with the subject seated in a double-walled sound booth at a 0 degree azimuth, 1 m from a Now Hear This (NHT) loudspeaker at head level. The loudspeaker has a 6.5 in woofer and 1 in tweeter, a response of 57 Hz to 25 kHz, a crossover of 2.2 kHz (6 dB/octave high-pass and 12 dB/octave low-pass), and a sensitivity of 2.83 V at 1 m. It was powered by a six-channel power amplifier (Rotel, model RB-976).

A 10 sec 1000 Hz calibration tone was created using Praat 3.9.22 software (Boersma and Weenink, 2001), to set the VU meter of the audiometer for delivery of the speech stimulus. The prerecorded calibration tone on the BYU CD was used to set the VU meter for the noise stimulus. The soundfield stimuli were calibrated using a Type I sound level meter (Brüel and Kjær [B&K], model 2204) and 1 in condenser microphone (B&K, model 4145). Calibration of this instrumentation was performed using a Quest Electronics (model CA-12B) and a B&K acoustical calibrator (model 4231).

Signal Delivery and Recording

Computer-driven (Hewlett Packard Pentium III) media software (Windows Media Player, version 6.01.05.0217) was used to present the speech stimulus. The noise was presented via a Sony CDB XE500 CD Player. An audiometer (Madsen Aurical) was used to control signal routing and presentation levels.

Recordings were obtained using a probe microphone (Etymotic Research, model ER-7C) coupled to a computer sound card (Sound Fusion Wave) and processed using speech recording and analysis software (Praat 3.8.69) at a monaural 44,100 Hz sample rate. A 94 dB SPL 1000 Hz tone generated by the B&K acoustical calibrator (model 4231) was recorded and functioned as a reference to convert relative phoneme segment level into absolute levels. The difference between the 94 dB SPL calibration tone and the recorded level observed in Praat was added to the root mean square (RMS) calculation for each phoneme. This was necessary because the levels displayed in Praat do not represent absolute levels.

Procedure

The subject was seated in the test position, and the instrument was fit to the NAL-R target as previously described. The subjects were given these instructions:

> The purpose of this test is to find and maintain a loudness at which sounds and words are most comfortable. You are going to hear three sentences repeatedly presented through the speaker. Please rate each presentation by telling us if it was "comfortable" or "too loud." We will then turn the volume of the hearing aid up or down, repeat the sentences, and ask you to rate them again.

To determine the preferred loudness level in each condition, the stimuli were presented, and the gain of the hearing instrument was adjusted using an adaptive procedure (Dirks and Kamm, 1976). Specifically, the gain of the hearing instrument was initially set to 4 dB above the prescribed gain; following the subject's response to the initial presentation, the gain was increased or decreased by 4 dB, and the procedure was repeated. This procedure continued until four reversals using a 4 dB step size were completed. The step size was then changed to 2 dB. The adaptive procedure was stopped after four identical, but not necessarily consecutive, reversals were obtained in the 2 dB step series. This yielded a gain setting consistently rated as "too loud" and a setting 2 dB lower consistently rated as "comfortable." This lower gain setting was noted and used for all subsequent real ear measures and recordings.

Preferred loudness levels, real ear measures, and recordings were obtained in each of six release time/noise conditions (i.e., 40, 160, or 640 msec in quiet and in noise). The noise was played continuously during the entire rating procedure for the noise trials, including the time between sentence presentations. The condition order varied and was counterbalanced among subjects to minimize any order effect. Test time was approximately two hours per subject, including audiometry, and was completed during one test day. There was a delay of approximately three minutes between conditions as the speech stimulus was recorded and the hearing aid was set to target for the next condition.

Real Ear Measures

Real ear insertion gain (REIG) measures were obtained at 65 dB SPL when setting the hearing instrument to the NAL-R target and for each condition at the selected comfortable setting for each subject. The REIG data were exported from the Madsen Aurical Real Ear Module to an Excel spreadsheet. The gain at all tested frequencies from 750 to 4000 Hz, 58 frequencies distributed logarithmically, was averaged to yield one value to represent gain for each condition.

Aided Real Ear Recordings of Speech Stimuli

Using the ER-7C probe microphone, real ear measurements of the three sentences were recorded for analysis for each condition at the gain level determined by the subject as "comfortable." All recordings were made in quiet following completion of the adaptive procedure, irrespective of the presence or absence of noise during the rating procedure.

Recording Analysis

The original speech recording and all test recordings were phonemically segmented into 68 units using the procedure described by Ellison et al (2003). All data sets were equivalently balanced, and 30 phonemic units were selected for analysis; data from the remaining 38 segments were not considered. These segments are indicated in bold in Figure 2.

pi**tə** wɪl ki**p** æ**/tθ/ʌ** tɔl **tʃ**ɪlɪ p**i**k.

θʌ **b**l**u** **s**pɑ**d** ɪ**z** ɑn **/ðʌ/** **θ**ɪn ki **əgɛ**n.

wɛ**n** **/hi/** **k**ʌms **hom** wɪ**l** fid **/hɪ/**m **f**ɪ**ʃ**.

Figure 2. Transcribed stimuli sentences; analyzed speech sounds in large, bold font. Phonemes grouped by // could not be separated due to coarticulation and were analyzed as one speech segment.

Amplified speech segment amplitudes vary considerably based on phonemic context. The influence of this variability was controlled by selecting only one incidence of each phoneme for analysis. Calculation of RMS amplitude for each selected segment was performed using a Matlab (V.6.1.0450R1.2) subroutine. Matlab is an integrated technical computing environment that allows for customized computational analysis of digitized sound samples.

Data Analysis

A multiple comparison ANOVA was performed for all measures. The influence of two variables (i.e., release time and noise) and their interaction upon preferred gain were assessed. An alpha of 0.02 was used to adjust for multiple comparisons. Also, the influence of release time on phoneme amplitude was assessed. A Tukey Studentized-Range (HSD) post hoc analysis was performed for all measures. Generalizability theory was applied to all measures to predict the portion of expected variance of each of the independent component variables on their respective dependent variables.

RESULTS

Preferred Gain

The preferred gain settings for each condition, collapsed across all 18 subjects, are illustrated in Figure 3. The results of an ANOVA indicated that the presence or absence of noise accounted for a significant portion of the variance; the release time setting did not. Tukey HSD post hoc tests ($\alpha = 0.02$) revealed a significant noise effect but no significant release time effect. On average, subjects preferred slightly less gain than prescribed by the NAL-R target in quiet and slightly more gain in noise. Averaged across all release time conditions, subjects preferred 2.86 dB more gain in noise than in quiet. The results of generalizability theory analysis indicated that when the noise, target, and noise/target interactions were considered,

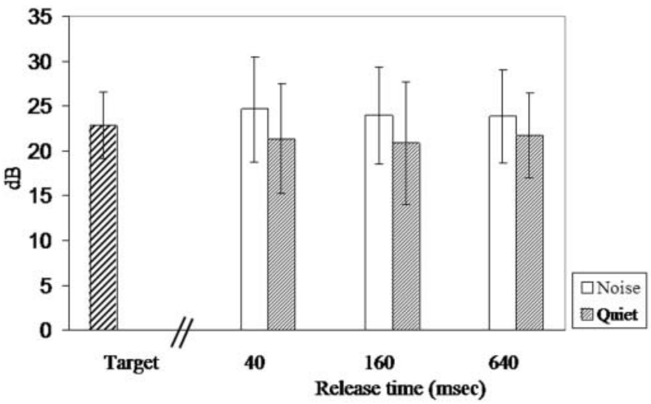

Figure 3. Average real ear gain at 58 frequencies from 750 to 4000 Hz. The striped bar represents mean target gain and the white and shaded bars mean gain selected by the subjects for each release time condition for noise and quiet, respectively. Error bars are +/- 1 SD.

the target accounted for 47% of the variance; the presence or absence of noise accounted for 35%. Release time did not account for a significant portion of the variance.

Phoneme Amplitude

The amplitudes of all 30 selected phonemes were calculated for each release time condition for presentations in both quiet and noise. Results of an ANOVA of these data indicated that RT accounted for a significant portion of the variance. The amplitudes in each release time condition were compared, and these differences are displayed in Table 1. Tukey HSD post hoc tests ($\alpha = 0.02$) revealed significant differences between 40 and 640 msec and between 160 and 640 msec. (It should be noted that there is a larger time difference between 160 and 640 msec than between 40 and 160 msec, possibly explaining why the former comparison was significant while the latter was not).

For the purpose of this paper, we will refer to the reduction in amplitude of speech segments with increasing release time as the "release time effect." The magnitude of this effect for each phoneme averaged for the 640 and 40 msec conditions (640–40 msec) is displayed in Figure 4. The graph is ordered from greatest negative to positive shifts. Some segments were dramatically affected, with a reduction of amplitude of approximately 6 dB; others had no shift or were even positively affected. An examination of the error bars indicates the extent of the variability for these differences.

Generalizability theory analysis was used to identify factors that might account for this

Table 1. Average Phoneme Amplitude Difference between Release Time Conditions

RT Comparison (msec)	Mean Difference (dB)
160–40	0.01
640–160	-1.18*
640–40	-1.16*

* Significant, $p < 0.02$

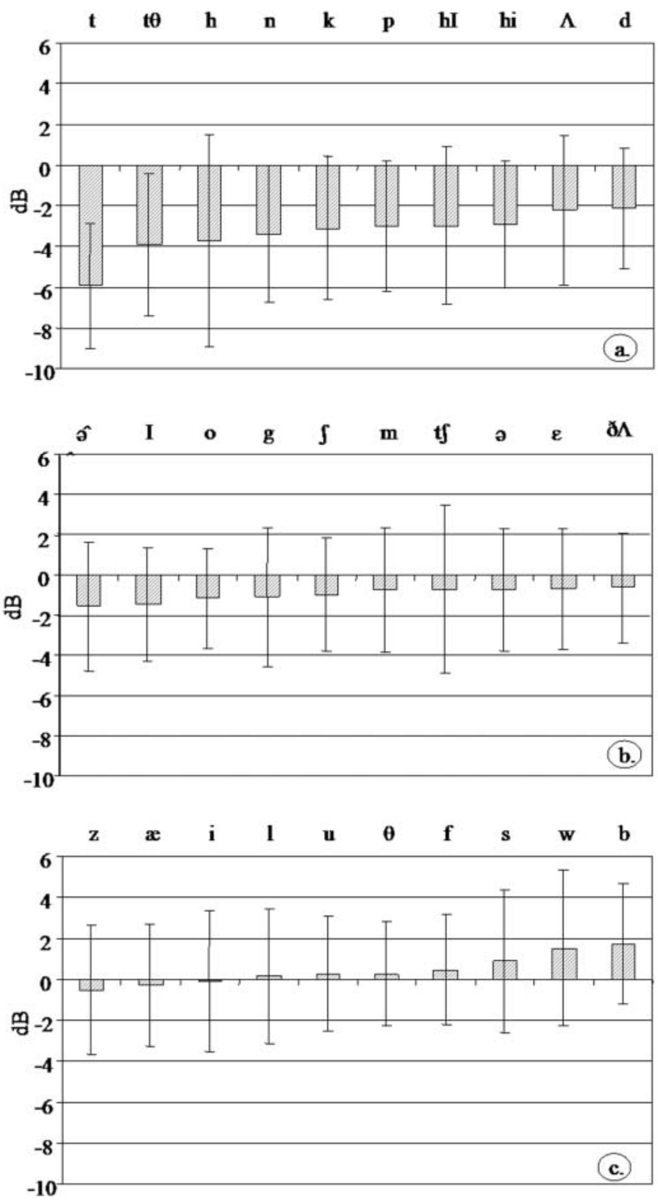

Figure 4. Differences in average segment amplitude (640-40 msec) as a function of release time for 30 phonemic units (see text). Panels *a*, *b*, and *c* have been separated arbitrarily for display purposes. Data are collapsed across noise and quiet conditions.

Table 2. Variance Accounted for by Speech Segment Amplitude (generalizability theory)

	Total Energy	Energy below 1500 Hz	Energy above 1500 Hz
Segment amplitude	11.56%	0%	0%
Preceding segment amplitude	1.19%	0%	20.06%
Interaction	0.5%	11.92%	0%
Error	87.25%	88.08%	79.94%
Total	100%	100%	100%

variability in effect. First, seven segments were identified (/æ/, /tð/, /ə/, /θ/, /g/, /ɛ/, /hi/, /k/, /ʌ/, /o/, /m/), each of which was selected because it was immediately preceded by another phoneme also included in the analysis. Two hypotheses were tested using this data set:

1. That the segment amplitude influenced the magnitude of the release time effect, with the lower-amplitude segments being affected the most, and
2. That the amplitude of the segment just prior to the segment of interest influenced the magnitude of the release time effect, with the sounds following high-amplitude segments affected most.

Because the hearing instrument's compression functioned in two channels, Matlab was used to divide the energy of each phoneme to that above and below the 1500 Hz crossover frequency. Changes in each segment's amplitude were compared to that segment's amplitude in the 40 msec condition, and to that of the segment just before it. Only the 640 to 40 msec change was considered. The percentage of the variance accounted for by the amplitude of the segment, the amplitude of the preceding segment, and the error (variables unaccounted for) is summarized in Table 2. The majority of the variance was not accounted for by the amplitude of either the targeted segment or the segment that preceded it.

SUMMARY AND DISCUSSION

The purposes of this study were twofold: First, to determine the impact of release time on listeners' preferred gain settings. Based on data from other investigators, we predicted that subjects would prefer less gain with shorter release time settings, possibly working counter to the goal of improved audibility with syllabic compression. Second, we sought to quantify the supposed advantage of fast time constants on speech transmission by assessing changes in phoneme amplitude associated with changes in release time as determined from real ear recordings. We discuss each of these in relation to our findings.

Gain Preferences

Listeners' gain preferences for stimuli in quiet and noise did not change with changes in release time of 40, 160, or 640 msec. This observation supports the notion that the highest amplitude elements of speech are primarily responsible for perception of overall loudness (Bakke et al, 1991; Zwicker and Zwicker, 1991). Changes in release time would be expected to alter primarily the low-amplitude speech elements. The highest amplitude elements (those in speech that result in the most compression) would cause roughly the same amount of compression in both slow and fast-acting systems. These amplitudes would be essentially unchanged with variations in release time. Therefore, subjects' perceptions of overall loudness, and thus their ratings of most comfortable loudness, would not be expected to change. This would also suggest that a listener would not be likely to adjust the volume control of the hearing aid when listening in quiet or noise on the basis of the RT alone. That a listener might make such adjustments for other reasons, such as to improve his or her understanding of speech, cannot be predicted from this study.

Cox and Alexander (1991) found that subjects wearing linear hearing aids preferred less gain in noise than in quiet. In contrast, our subjects, who were listening through a WDRC instrument, preferred 2.86 dB more gain in noise, irrespective of RT. This is consistent with Souza and Kitch (2001b), who also found that listeners preferred approximately 3 dB more gain in noise than in quiet when listening through WDRC instruments. One explanation for this finding might be that subjects would increase gain to compensate for a loss of gain caused by compression in response to the noise. To provide evidence for this assumption, we measured the output of the speech signal used in this study under identical test conditions but with the hearing aid coupled to a 2-cc coupler that was attached to a Larson-Davis 824 sound level meter. Two settings were evaluated with the hearing aid programmed to the frequency response and gain characteristics of a representative subject. The compression ratio was set to either 3:1 with a 40 msec release time or 1:1 (i.e., "linear"). In both the compression and linear conditions, visual inspection of the analysis of peak SPL values over three repetitions of the stimulus revealed that the occurrences of higher amplitude elements of the output signals were essentially unchanged in the signal plus babble condition. This implies that addition of babble at 50 dB SPL was not sufficient to activate the compression circuit, suggesting that our low level (50 dB SPL) babble was not significantly activating compression. This finding supports the notion that the subjects' preference for more gain in noise was due to some other aspect of their perception of the signal.

Phoneme Amplitude

Changing release time had a dramatic effect on the amplitude of some speech segments. For example, the /t/ (**pitə̂ wɪl kip**…) was reduced approximately 6 dB when RT was increased from 40 to 640 msec. However, the magnitude and even direction of the effect was difficult to predict on the basis of either speech segment or RT, and there was considerable variability. These differences could not be accounted for by either the amplitude of the targeted segments or of the preceding segment. When both of these variables were evaluated, 87.25% of the variance remained unaccounted for. We considered that in our two-channel system, signals below the crossover frequency (1500 Hz) would control low-frequency gain while those above the crossover would influence high-frequency gain. However, separate consideration of these two frequency ranges had poor predictive value. The effect of compression on continuous speech is obviously complicated. The data imply that the acoustic costs and benefits of compression change from moment to moment, from speech sample to speech sample, and from environment to environment. Lengthening RT can result in a substantial loss of amplitude for some phonemes, but it is very difficult to predict which ones or under what conditions this might occur.

We did not evaluate speech recognition. Therefore, it is not known whether the observed changes in phoneme amplitude would result in any change in a subject's ability to understand speech. It may be reasonable to speculate, however, that increased audibility would have a positive effect. Souza and Turner documented improved recognition of VCV syllables with syllabic compression over linear amplification, particularly at low input levels. An improvement in audibility with compression was calculated using the Aided Articulation Index (AAI), and this improved audibility was correlated to the observed improvement in speech recognition (Souza and Turner, 1998, 1999). In a similar study using nonsense syllables, Stelmachowicz et al (1995) also found that AAI-calculated audibility was a good predictor of subject performance. Even so, the loss of spectral amplitude that comes with increasing RT might be less important than the retention of temporal cues under such conditions.

CONCLUSIONS

Despite prior reports that listeners describe speech that has been amplified using short RT as being louder and noisier, they are not likely to reduce the gain of their hearing aid to compensate for this effect under shorter RT conditions. This finding supports the notion that formulas do not need to account for RT when prescribing optimal gain. However, listeners are likely to increase gain in a nonlinear hearing aid in the presence of noise, no matter how the RT of the hearing aid has been set, even if the noise is not intense enough to result in compression. The effects of changes in RT on the acoustic elements of speech, as measured in the ear canal, are obviously complicated and unpredictable. The conclusions drawn from the results of this study are necessarily confined to the specific conditions of the investigation, some of which may limit the generalizability of the findings. Specifically, the hearing aid that was selected satisfied the requirements for manipulation of the variables under test. However, other hearing aids may have parameters, such as their filter characteristics, that could influence signals differently. This was not evaluated in this study.

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