Multi-Channel Compression and Speech Intelligibility in Industrial Noise

This study examined how various forms of compression affect speech intelligibility in moderately high levels of industrial noise. Thirteen listeners with high-frequency sensorineural hearing loss were fit with a digital hearing aid programmed to operate in three modes of compression: compression limiting, compression of low frequencies only (BILL), and compression of high frequencies only (TILL). For each of these conditions, listeners attempted to recognize CID W–22 monosyllabic words delivered in a background of recorded industrial noise. Word recognition was also measured in quiet for the compression-limiting condition. The highest word identification scores were obtained in quiet (80.9%). Of the three conditions in which background noise was present, the mean score was highest for TILL compression (52.4%), and lowest for BILL (39.7%). Representative time-weighted average (TWA) exposures for each of the aided noise conditions were determined by means of a procedure outlined previously by the first author. The projected 8-hour amplified TWA with the hearing aid programmed to the TILL configuration was 94 dBA, whereas TWAs for the linear and BILL configurations were each 104 dBA. Thus, the highest intelligibility and lowest noise exposures were obtained with the aid in the TILL mode.

KEY WORDS: hearing aids, compression, industrial noise, hearing loss, intelligibility

One of the most common problems in hearing aid use is difficulty in understanding speech in background noise. Over the years, hearing-aid manufacturers have spent millions of dollars on the development of technology to alleviate this problem. Technical solutions have included analogue and digital filtering, compression, directional microphones, and the application of various signal-processing algorithms. These developments have been accompanied by a great deal of research aimed at determining how effectively the various technologies mitigate the interfering effect of background noise on the intelligibility of speech (Dillon, 1996; Hickson, 1994). However, the vast majority of this work has focused on the effects of low or moderate levels of ambient noise in avocational settings, such as restaurants, classrooms, and cocktail parties. Relatively little research has addressed problems of the smaller, but growing, population of hearing aid users who work in noisy industrial settings.

Many workers with hearing impairment use hearing aids on the job in order to communicate with other personnel directly or via radio, to monitor equipment sounds, or to listen for warning signals, such as sirens and back-up alarms (Hétu, 1994). In some instances, failure to hear
warning sounds or speech can place the individual in danger of physical injury (Hétu, Getty, & Tran Quoc, 1995). Consequently, workers with hearing loss often feel that they cannot perform their jobs safely without using amplification.

Industrial workers who wear hearing aids on the job are also subject to hearing loss due to noise exposure. That is, wearing a hearing aid can place the worker at risk of exacerbating his or her hearing loss, because the already high ambient noise may be amplified to even higher levels. Dolan and Maurer (1996) demonstrated that amplified exposures to otherwise safe levels of industrial noise resulted in 8-hour time-weighted averages (TWAs) that exceeded the Occupational Safety and Health Administration’s limit of 90 dBA (OSHA, 1983). In that study, a worker wore two dosimeters while operating a turret lathe in a manufacturing plant. One of the dosimeters logged ambient noise levels while the other simultaneously measured the output of one of three hearing aids, each with a different amount of gain. Dolan and Maurer found that ambient TWAs ranging from 82.6 to 84.1 dBA were raised to 91.8 dBA by a mild-gain hearing aid, 104.6 dBA by a moderate-gain instrument, and 115.4 dBA by a high-gain hearing aid. Similar results were obtained in laboratory measurements using noise recorded in other industrial settings.

The issue facing audiologists is whether workers can be protected from auditory damage due to amplified industrial noise, while maintaining the benefit of hearing aids for work-related tasks. One approach to resolving this issue would be to lower the level of the ambient noise through engineering controls (such as installing a sound barrier around a noisy machine), or by reducing the amount of time a worker spends in noisy areas. However, even industries with noise levels well above 90 dBA are reluctant to pursue such solutions because they are expensive, and in many cases, impossible to implement. Companies with ambient noise levels already below the OSHA maximum would be even less motivated to apply these measures.

A more realistic approach might be to control amplified exposures by manipulating the worker’s hearing aid. The most obvious means would be through compression, or automatically reducing the gain of the hearing instrument when input levels exceed certain values. Indeed, some data suggest that occupational noise exposure among hearing-aid users can be significantly reduced through the use of commercially available compression systems. Dolan and Maurer (1994) demonstrated that 8-hour amplified exposures to moderate levels of industrial noise could be lowered to values near the OSHA maximum of 90 dBA by manipulating compression thresholds in a multi-channel programmable hearing aid. Dolan and Maurer (1996) also demonstrated that amplified exposure for a high-gain hearing aid was reduced substantially when single-channel compression was employed. Specifically, the TWA in industrial noise was reduced from 115 dBA, when the aid operated in a linear mode, to 104.3 dBA when compression was set to its maximum value.

The question remains, however, whether a worker’s noise dose can be reduced through compression without seriously compromising his or her understanding of speech. This question is complicated by the fact that there are many compression systems presently available, and it is unclear which best preserves the intelligibility of speech in occupational noise. The present study focused on three compression systems currently in widespread use: compression limiting, compression of low or bass frequencies, and compression of high or treble frequencies. We will refer to the latter two systems as BILL (bass increase at low levels) and TILL (treble increase at low levels), following the classification system of Killion, Staab, and Preves, 1990. Each of these signal-processing schemes is based on a different rationale for dealing with background noise (Dillon, 1996; Sammeth & Ochs, 1991). In compression limiting, the hearing aid amplifies linearly for low to moderate input levels, but the overall gain of the instrument is reduced automatically when input levels are high. This system is employed mainly in single-channel instruments with the goal of diminishing or eliminating distortion due to amplifier saturation at high input levels. It is also intended to reduce loudness discomfort in noise (Dillon, 1996). In BILL processing, gain in the low frequencies is reduced automatically as the input level is raised, whereas gain in the high frequencies is essentially held constant. This system was developed under the assumption that the acoustic energy of environmental noise tends to be concentrated in the low frequencies, and that the interfering effect of this noise can be reduced by making gain in the low frequencies dependent upon input level (Ono, Kanzaki, & Mizoi, 1983). On the other hand, TILL systems automatically reduce gain in the high frequencies as input level is raised, or conversely, automatically increase gain as input level is lowered. This type of compression was popularized with the introduction of the K-amp in 1990 (Killion, 1990). The rationale for this system was that for individuals with high-frequency hearing loss, gain in the high frequencies is needed mainly to amplify weak consonants when input levels are low; less gain is required at higher input levels where sounds are annoying (Killion et al., 1990). TILL processing is now employed commonly with multi-channel instruments, in part with the goal of restoring normal loudness growth for persons with high-frequency hearing loss (Kuk, 1999).

The purpose of the present study was to determine...
which of these signal-processing schemes best preserves the intelligibility of speech in industrial noise. Listeners with high-frequency sensorineural hearing loss were fit with a digital hearing aid programmed to compress in three modes: compression limiting, BILL, and TILL. For each of these conditions, the listeners attempted to recognize CID W–22 monosyllabic words delivered in a background of recorded industrial noise. Word recognition was also measured in quiet for the compression-limiting condition in order to assess the effect of the noise on speech intelligibility.

**Method**

Eight men and 5 women, who ranged in age from 28 to 71 years, served as listeners. All had sensorineural hearing loss with no history of conductive or retrocochlear involvement. Figure 1 shows the mean pure-tone thresholds and their ranges for the 13 test ears. Although the severity of hearing loss varied by up to 70 dB across ears at some frequencies, each listener's audiogram had the same sloping configuration, with thresholds being 40 to 75 dB poorer at 8000 Hz than at 250 Hz. Each listener had a minimum of 1 year of experience as a hearing-aid user.

Industrial noise was recorded on the shop floor of a metal heat-treatment plant by means of a TEAC DA-P20 digital audio tape (DAT) recorder and a calibrated Larson-Davis 2560 half-inch condenser microphone. A 2-minute segment of the recording was used for the noise conditions. The sample consisted of the sounds of hammering and bursts of compressed air against a background of steady-state noise produced by an overhead crane. Figure 2 shows the power spectrum of the noise measured acoustically in the sound chamber where the participants were tested. This was done by means of a Rockland 5840A FFT analyzer and a Larson-Davis 2560 half-inch microphone placed at a position that would be occupied by the listener's head (listener absent). The 2-minute segment of noise was presented via the same loudspeakers used in the test conditions, and the FFT analyzer was set to “peak averaging” mode. Thus, the spectrum depicted in Figure 2 gives the highest amplitudes that occurred at each frequency over the 2-minute period. The plot shows a peak at 125 Hz, and a drop in level of about 20 dB from 125 to 1000 Hz. The energy was distributed relatively evenly across frequencies above 1000 Hz.

Speech materials consisted of the Brigham Young University (1991) compact disk recording of CID W–22 monosyllabic words. In the BYU recording, each of the original CID W–22 lists of 50 monosyllabic words are divided into four lists of 25 words labeled “least difficult” and four lists labeled “most difficult.” The abbreviated lists within each category are balanced for homogeneity of audibility (Cromwell, 1995). The four “least difficult” lists were employed in this study. These were copied to the hard disk of a desktop computer by means of a Yamaha OPL3-SA sound card. The 2-minute sample of industrial noise was also transferred to hard disk, and the noise and speech waveforms were manipulated by means of a sound editing program (Johnston, 1996).

**Figure 1.** Mean audiogram of the 13 listeners (squares connected by solid lines). Curves above and below the mean audiogram denote threshold ranges.

**Figure 2.** Power spectrum of the recorded industrial noise measured acoustically at the output of the loudspeakers. The ordinate is peak level of the noise in decibels (arbitrary scale) measured over the entire 2-minute noise sample.
For each word list, the speech and the noise were arranged separately on two channels. The peak amplitudes of the speech and noise were adjusted to be equal (+1.2 dB), and the onset times of the words were adjusted such that word presentations from list to list occurred at the same points (+331 ms), relative to the noise waveform. Each word list and the noise were then dubbed onto separate channels of one of four audio cassette tapes. Using a different tape for each list facilitated the manipulation of their order of presentation.

A Siemens Prisma behind-the-ear hearing aid was used for all listeners. The Prisma is a four-channel, digital instrument in which compression settings can be programmed independently in each channel. This hearing aid also has a front-end input compressor with a high compression threshold that limits the microphone output prior to digital processing. The hearing aid was programmed by means of a computer equipped with a Pentium II processor, a Hi-Pro interface, and the manufacturer’s software (Connexx, Version 3.0). Advanced features, such as the voice activity detection and dual microphone processing, were disabled for this study. Crossover frequencies between the adjacent frequency channels were 400 Hz, 600 Hz, and 2000 Hz. The following paragraphs describe the three compression conditions.

**Compression Limiting**

For this condition, the compression characteristic in the Connexx software was set to “linear gain.” Figure 3 shows an input-output function of the hearing aid set to this configuration. To obtain these data, the gain of the hearing aid was set to approximate the Berger, Hagberg, and Rane (1989) prescribed coupler values for the mean audiogram depicted in Figure 1. The measurements were made with a Fonix 6500 hearing aid test system using the Frye HA-2 coupler and composite speech-weighted input signal. The function is linear for inputs up to about 75 dB SPL. As the input level was raised beyond 75 dB SPL, the slope of the curve progressively decreases, indicating that the hearing aid was compressing. Compression ratios determined from this plot ranged from 1.4:1 between inputs of 75 and 80 dB SPL, to 12.5:1 between 85 and 90 dB SPL. Figure 4a further illustrates the response characteristics of the instrument programmed to this configuration by showing a series of plots of its gain as a function of frequency. The curves show that gain was reduced markedly at frequencies below 1500 Hz when the level of the input signal was raised, whereas the portions of the curves above 2000 Hz remain almost identical to their counterparts in the compression-limiting condition (Figure 4a). It is also noteworthy that with 50 dB SPL input, the gain in the low frequencies was up to 9 dB below what it was in the compression-limiting condition for this input level. This indicates that, even with 50 dB SPL input; compression was occurring in the low frequencies. Under Dillon’s (1996) classification system, this mode of compression would be categorized as “wide dynamic range compression for bass frequencies,” or WDR-B.

**TILL**

In this condition, the three channels below 2000 Hz were set to “curvilinear compression” with the compression threshold (CT) nominally set to 34 dB SPL. The high-frequency channel was set to “linear gain.” Figure 4b gives gain curves obtained with the composite input signal at levels of 50 to 80 dB SPL. The curves show that gain was reduced markedly at frequencies below 1500 Hz when the level of the input signal was raised, whereas the portions of the curves above 2000 Hz remain almost identical to their counterparts in the compression-limiting condition (Figure 4a). It is also noteworthy that with 50 dB SPL input, the gain in the low frequencies was up to 9 dB below what it was in the compression-limiting condition for this input level. This indicates that, even with 50 dB SPL input; compression was occurring in the low frequencies. Under Dillon’s (1996) classification system, this mode of compression would be categorized as “wide dynamic range compression for bass frequencies,” or WDR-C-B.
Figure 4. Gain of the hearing aid in dB as a function of frequency in Hz with the hearing aid programmed to (a) compression limiting, (b) BILL, and (c) TILL. The parameter in each graph is the level of a composite speech-weighted input signal in dB SPL. Measurements were made using the Fonix 6500 Hearing Aid Test System with a 2 cm$^3$ coupler.
lower frequencies was essentially unchanged relative to the compression-limiting condition (Figure 4a). As in the BILL configuration, however, the hearing aid was compressing at the lowest input level. This is evidenced by the fact that the high-frequency portion of the 50 dB SPL curve is up to 6.6 dB lower than in the compression-limiting condition. Under Dillon's (1996) system this processing scheme would be classified as "wide dynamic range compression for treble frequencies," or WDRC-T.

For each participant, the hearing aid was fit on the ear with the lowest (best) thresholds. The gain in each channel of the aid was adjusted to approximate the Berger et al. (1989) prescribed values for the test ear when the compression parameters were set to achieve "compression limiting" as described above. The programmed hearing aid was then coupled to the listener's ear by means of a foam insert earmold, and the gain was verified by conducting real-ear measurements with a Rastronics 3000 hearing-aid analyzer. With the aid still set to "compression limiting," the overall gain was then adjusted to a value at which the listener judged the experimenter's voice to be comfortable when he spoke at a normal conversational level. Once the final gain settings for compression limiting were achieved, the aid was programmed as described above for the BILL and TILL conditions.

The noise and speech stimuli were presented to the listeners by means of a Proton 740 stereo cassette deck. The left and right channels of the cassette deck were routed to the left and right channels of a Grason Stadler 16 audiometer, the outputs of which were led to a pair of Grason Stadler loudspeakers. Playback level for each channel was set in reference to 1000 Hz calibration tones, which preceded the speech and noise on each stimulus tape. Listeners were seated in a double-walled sound chamber with the two loudspeakers positioned 30 inches from the test ear at azimuths of 45 and 135 degrees for right ear fittings, and 225 and 315 degrees for left ear fittings. The unaided ear was not occluded. The industrial noise and the speech stimuli were presented through both loudspeakers simultaneously at levels of 75 and 80 dBA, respectively. A quiet condition was also included in which word lists were presented at 80 dBA with Kemar in the same position as the listeners. The gain and compression settings of the hearing aid were the same as for Figure 4.

Results

The projected 8-hour time-weighted average with gain set to prescribed values for the mean audiogram (Figure 1) was 94 dBA when the aid was programmed to TILL compression. TWAs for compression limiting and BILL compression were each 104 dBA.

The bar graph in Figure 5 gives the mean percentage of monosyllabic words correctly recognized (hatched bars) by the 13 listeners in each condition and the respective standard errors of the means (filled bars). As would be expected, the highest scores were obtained in the absence of background noise (80.9%). In this condition the hearing aid was programmed to compression limiting. When the industrial noise was mixed with the speech in the compression-limiting condition, the mean score dropped to 47.4%. Of the three conditions in which background noise was present, the mean score was highest for TILL (52.4%), and lowest for BILL (39.7%).

A one-way repeated-measures analysis of variance revealed a significant main effect (p < .001), indicating that performance was significantly different across the four listening conditions. Paired one-tailed t tests indicated that scores in the quiet condition were significantly higher than for each of the noise conditions (p < .001). Performance with the aid programmed to compress high frequencies (TILL) was significantly better.
denote the standard error of the mean for each condition.

Word-recognition scores in the TILL condition were not significantly different from those in the compression-limiting in noise condition. Thus, gain in the high frequencies could be reduced substantially without having a negative effect on the intelligibility of speech in the background noise. Previous studies of the benefits of TILL compression on the intelligibility of speech in noise have yielded equivocal results (Dillon, 1996). Moore, Johnson, Clark, and Pluvinage (1992) found that their subjects’ speech-reception thresholds in babble noise tended to be lower when their dual-channel hearing aids were configured to compress high frequencies more than low frequencies. This result was consistent with our finding of higher word-recognition scores for TILL processing. However, our results differ from those of Horwitz et al. (1991). These authors compared synthetic syllable identification by four hearing-impaired listeners under conditions of TILL processing, BILL processing, and linear amplification. No differences were found among the three conditions when the speech was delivered in quiet or in a background of Gaussian noise. The discrepancy with the present findings could be related to any of a number of differences in the methodologies of the two studies. For example, their stimuli consisted of a small, closed set of speech sounds (six consonants and one vowel) delivered via earphones, whereas stimuli in the present study consisted of an open set of monosyllabic words delivered in sound field to listeners wearing hearing aids.

The TILL configuration was also advantageous in terms of the amount of noise exposure to the aided ear. In the TILL condition, the projected 8-hour time-weighted average was 4 dB above the OSHA maximum of 90 dBA, whereas for the BILL and compression-limiting configurations, it exceeded the maximum by 14 dB. Dolan and Maurer (1994) also found TILL processing to be more effective than BILL in reducing aided...
noise exposure for persons with high-frequency hearing loss. In that study, a TWA of 101 dBA was obtained when noise near 75 dBA was amplified by a hearing aid programmed to a BILL configuration. With the same aid configured to TILL, a TWA of 90.7 dBA was obtained. The effect is not surprising, given that for sloping losses the prescribed gain is greatest at frequencies above 2 kHz. Automatically reducing the gain in this frequency region would be expected to result in a lower overall output level than for gain reduction only at frequencies below 2 kHz. The higher TWA for compression limiting can be explained by the higher compression threshold in this condition. That is, little reduction in gain would be expected for this condition given that the 75 dBA noise was close to the CT, and the compression ratio was relatively low for inputs up to 80 dB SPL.

The fact that the projected TWA exceeded 90 dBA in each of the compression conditions raises the question of whether it would be appropriate to use amplification of this sort in comparable occupational noise. Both regulatory and safety issues must be considered here. The OSHA (1983) regulations do not address the issue of hearing aid use in occupational environments, nor has OSHA issued any interpretations or compliance directions on this matter. The OSHA maximum 8-hour TWA of 90 dBA applies only to ambient levels of noise in the workplace as measured in sound field with dosimeters or sound level meters. It does not directly apply to levels at the tympanic membrane, or in the case of a hearing aid user, to the amplified exposure. Thus, even though a worker wearing a hearing aid might have daily exposures exceeding 90 dBA TWA, his or her employer technically would be in compliance with OSHA (1983) regulations as long as the ambient noise were 90 dBA or less, and he or she did not exhibit a standard threshold shift.

OSHA's 8-hour exposure limit of 90 dBA was developed in consideration of the risk of damage to normal ears. If the same criterion were applied to ears fit with hearing aids, most hearing aid users would probably be considered at risk for hearing loss due to noise exposure. This is so because even in non-occupational settings, many hearing aid users are likely to have amplified TWAs above 90 dBA (Dolan & Maurer, 1996). However, studies that have monitored the thresholds of large numbers of hearing aid users tend to show no significant exacerbations of existing hearing losses in most subjects (Bellefleur & Van Dyke, 1968; Gelfand, Silman, & Ross, 1987; Markides, 1976; Naunton, 1957; Titche, Windrem, & Starmmer, 1977). The use of properly prescribed amplification is thus generally considered to be safe. Also, a number of studies suggest that the risk of temporary or permanent threshold shift diminishes with the amount of pre-existing hearing loss (Humes & Bess, 1981; Humes & Jesteadt, 1991; Kraak, 1981; Melnick, 1991). Aided exposures above 90 dBA TWA are thus not necessarily damaging to the ear. How then might we determine whether an amplified exposure for a particular worker is safe? The most expedient method would be to closely monitor the worker's pure-tone thresholds to ensure that he or she does not experience a temporary threshold shift (TTS). The presence of TTS would indicate that the amplified noise dose was excessive, and that intervention was necessary. Intervention could take the form of changing the response characteristics of the hearing aid, reducing the ambient noise levels through engineering or scheduling controls (OSHA, 1983), or simply having the worker abstain from using a hearing aid in that environment. For cases in which the threshold shift were sufficient to be classified as a standard threshold shift (defined as an average change in threshold of 10 dB or more at 2, 3, and 4 kHz relative to the baseline audiogram), federal regulations would require that the individual use hearing protection devices.

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