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SII and fit-to-target analysis of compression system performance as a function of number of compression channels

SII y análisis de adaptación a meta del desempeño de un sistema de compresión como función del numero de canales de compresión

Key Words
Hearing aid
Compression
Speech intelligibility index
Cambridge algorithm

Abbreviations
ANSI: American National Standards Institute
CR: Compression ratio
PB: Phonetically balanced
PTA: Pure-tone average
RAA: Residual auditory area
REAG: Real ear aided gain
RMS: Root-mean square
SII: Speech intelligibility index
ULCL: Upper limit of comfortable loudness
WDRC: Wide dynamic range compression
%C: Percent correct

Abstract
This work was undertaken to answer the question, 'How does the speech audibility/fit-to-gain-target provided by compression change with number of channels?' For each of 957 audiograms and a given number of compression channels, the channel crossover frequencies were set either to maximize the SII (speech intelligibility index) for low- and high-level speech spectra, or to optimize the fit-to-gain targets from the Cambridge method for loudness equalization (CAMEQ). The audiograms comprised all common configurations, and losses ranged from mild to severe. Use of these computational procedures allowed the predicted, channel-number-based performance to be determined separately from the effects of other compression parameters. From one to five channels were sufficient to yield predicted speech recognition performance within 5% of maximum for 90% of the 'mild' and 'moderate' audiograms. Three to nine channels were necessary for the same level of predicted performance for 90% of the 'severe' audiograms. Four channels or fewer were sufficient to produce less than 5 dB rms error in fit to CAMEQ targets for 90% of all audiograms.

Sumario
Este trabajo se llevó a cabo para responder a la pregunta "¿Como cambia con el numero de canales la audibilidad/ganancia meta de acuerdo a la compresión? Para cada uno de los 957 audiogramas y un número dado de canales de compresión, se fijaron las frecuencias de intersección para maximizar el índice de inteligibilidad del lenguaje para espectros de habla de alta y baja intensidad o para optimizar el blanco justo a la ganancia a partir del método Cambridge para ecualización de sensación subjetiva de intensidad (CAMEQ). Los audiogramas presentaban todas las configuraciones y las pérdidas variaban de leve a severas. El uso de procedimientos computacionales nos permitió predecir en forma separada el desempeño por número de canales del efecto de otros parámetros de compresión. De uno a cinco canales fue posible predecir el desempeño alrededor del 5% con un reconocimiento máximo de 90% para los audiogramas con pérdida leve a moderada. Fueron necesarios de tres a nueve canales para obtener el mismo nivel de desempeño en la predicción de. 90% de las pérdidas severas. Fue suficiente con cuatro canales o menos para producir errores de menos de 5dB rms en la adaptación de metas CAMEQ del 90% de todos los audiogramas.

The use of compression hearing aids is motivated mainly by the fact that amplitude compression (i.e. gain change as a function of stimulus level) is required to 'fit' the dynamic range of acoustic stimuli into the reduced auditory dynamic range of listeners with hearing impairments. This is generally accomplished by using more hearing-aid gain for lower-intensity than for higher-intensity stimuli. In this way, less intense sounds are made audible while more intense sounds are neither peak-clipped nor delivered at uncomfortable levels.

Because the severity of hearing loss usually varies with frequency, the appropriate compression characteristic will also vary with frequency. Even for a hearing loss that is the same at all frequencies the appropriate gain for speech recognition would vary across frequency for two reasons: 1) the spectrum of speech varies across frequency (Byrne et al, 1994); and 2) when sounds of differing spectra overlap in time (e.g. low-level, high-pitched birdsong in the presence of high-level, low-frequency sound from a bus) listeners could benefit from different gain applied in the frequency regions of the different sounds. The basic motivations for *multichannel* compression hearing aids therefore are (cf. Villchur, 1973; Moore, 1990; Keidser & Grant, 2001): 1) the dynamic range of impaired hearing varies with frequency; and 2) the intensity of input signals (especially speech) varies with frequency, sometimes in complex ways.

There is a limit, however, to the benefit of increasing the number of compression channels. With a large number of narrow channels, short time constants, and high compression ratios, one runs the risk of removing most spectral information from the speech signal (Plomp, 1988). What, then, is an appropriate number of channels? Although there are some evaluations of this issue (e.g. Plomp, 1988, 1989, 1994; Villchur, 1989; Yund & Buckles 1995; Crain & Yund, 1995), the equivocal results of those studies are due partly to the wide variation in possible configurations of compression parameters other than channel number. That is, in what were intended as evaluations of the effect of number of compression channels, other parameters

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(such as gain as a function of frequency) covaried with number of channels, making it difficult to determine the role of number of channels *per se*. Thus, an evaluation of the effects of number of channels is still needed. These considerations motivate this work's computational evaluation of the speech audibility and target-gain fit provided by multi-channel compression as a function of number of channels.

The computational evaluation in this study was based on either: 1) the calculated intelligibility index (ANSI, 1997) of speech in quiet for hearing-impaired persons listening to the compression systems, or 2) the ability of the compression system to produce channel gains matching the gain targets of a clinical non-linear fitting method. Of the many possible measures of hearing aid performance, these two were chosen because they correlate highly with, and directly represent, hearing aid outcome measures (i.e. aided speech intelligibility and ability to fit gain target, respectively) that are important in the fitting process (ASHA, 1998).

Toward this end, the work reported here is concerned only with isolating and describing the effects of the *number* of compression channels on the speech intelligibility index (SII, ANSI, 1997) and fit-to-target measures of performance for a comprehensive range of hearing loss configurations and degrees. Therefore, aided versus unaided listening effects are not considered, and compression parameters other than channel number were assumed fixed at values appropriate to the main purpose of the investigation. Although it has been argued that the SII may not accurately predict intelligibility in all conditions (e.g. Ching et al, 1998), it is the case that suggested methods for increasing the SII accuracy would in fact *decrease or leave unchanged* the number of channels required to achieve a given SII (this is considered briefly in the Discussion section). Thus, use of the SII as specified (ANSI, 1997) yields the most stringent evaluation possible with respect to practical predictions of intelligibility performance.

Methods

General

Maximum achievable SII (constrained such that speech peaks are never above the upper limit of comfortable loudness; cf. Rankovic, 1991, 1995), and best achievable fit-to-target-gain were computed for low- and high-level (45 and 82 dB SPL, respectively) speech as a function of number of compression channels, and for a wide variety of audiograms. Both SII and fit-to-target were maximized for a given number of channels by varying the compression channel crossover frequencies and gains, and choosing those crossovers and gains yielding the largest SII or best fit-to-target for each audiogram.

In performing the analysis, some simplifying assumptions were made:

1. *Slow-acting compression is used.* In the SII specification (ANSI, 1997) it is assumed that only energy in a 30-dB range ($\pm$15 dB re: RMS) in each 1/3-octave band contributes to intelligibility. Thus, the current assumption is required for application of the SII. Only compression systems with time constants much slower than the modulations of speech (referred to as 'AGC' systems) fit this assumption. Note that we used the analysis of Stone and Moore (1992) to provide a frame of reference for time constants. We determined the release time required to compress a sinusoidal envelope with a 30 dB dynamic range (the speech range) to a 29 dB dynamic range when the compression ratio was 3:1 and the attack time was 1 ms. We found that this occurred when the 'RC' release time was 0.3, 0.69, and 1.3 s for 1-, 2-, and 4-Hz modulation frequencies, respectively. The 1 Hz modulation was the lowest value investigated by Stone and Moore (1992), and 2 and 4 Hz represent the rate of words and syllables, respectively, in the modulation spectrum of speech. The 1-dB dynamic range change is the just-noticeably-different value approached for high modulation indices and 16 Hz modulation in the data of Ewert and Dau (2004, Figure 1). Multiplying the 'RC' release time by 4.0 will give the corresponding ANSI-S3.22 (2003) release time.

   The assumption is appropriate because in such systems the effects of number of compression channels are most apparent. This is because fast systems can quickly reconfigure gain for non-simultaneous different-frequency inputs within the same compression channel. Slower systems must depend on different compression channels to react appropriately to these inputs, and thus generally would require more channels to achieve a given performance level. Thus, the results here represent an upper limit on performance for slow-acting compression systems, but a lower limit on well-designed, faster-acting systems with the same channel configurations as those designed here. We note also that application of the aided audibility index (AAI; Stelmachowicz et al, 1994; Souza and Turner, 1999) instead of the SII is not necessary, since we are using the slow-system assumption to form the lower limit on performance (with respect to time constant value), and for such a system the SII and AAI would be equivalent.
2. *The compression ratio can be no greater than 3:1.* In order to be consistent with compression ratio constraints in current hearing aids, we limit the compression ratio to 3:1 or lower. This constraint has evolved because of the typical size of the dynamic range of the target population, and because of the lower preference or quality ratings given to WDRC

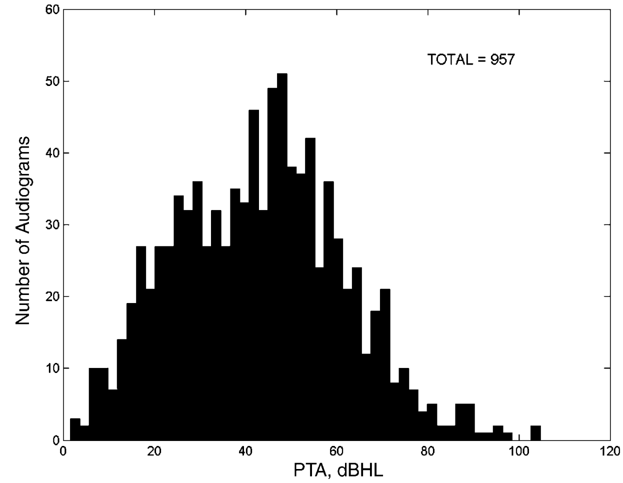

Figure 1. Histogram of pure-tone average (PTA, threshold averaged across 500, 1000, and 2000 Hz) for the audiograms used in this study.

Table 1. Center and crossover frequencies of the one-third octave band analysis. Center frequencies are taken from the SII standard (ANSI 1997), Table 3. The cut-off frequencies equal the center frequency multiplied by $\sqrt[6]{2}$. The lower cut-off on the first band is at 143 Hz (i.e. $160/\sqrt[6]{2}$)

Band #	*Center Frequency (Hz)*	*Upper Cut-off Frequency (Hz)*
1	160	179
2	200	224
3	250	281
4	315	355
5	400	447
6	500	561
7	630	710
8	800	894
9	1000	1118
10	1250	1414
11	1600	1789
12	2000	2236
13	2500	2806
14	3150	3550
15	4000	4472
16	5000	5612
17	6300	7099
18	8000	8980

systems with high compression ratios (e.g. Neuman et al, 1994; Woods et al, 1999; Hansen, 2002). Computations with and without this constraint indicate that it limits the ability of a given N-channel system to maximize performance (i.e. compression systems without this constraint will require fewer channels for similar results). Our results with the constrained computations are thus conservative.

3. *The compression channels are rectangular* and always comprise one or more contiguous analysis bands of the 18-band, 1/3-octave SII procedure. This assumption simplifies the calculations without significant loss of applicability of the results. Table 1 provides frequency information for these bands.

Because the SII 1/3-octave band frequency analysis underlies all compression channel configurations, all crossover frequency sets used here (see Table 1) comprised the SII analysis crossover frequencies, and all compression channels comprised combinations of contiguous SII 1/3-octave analysis bands. For the SII-based computation, the 'best' N-channel combination of bands is the combination that allows the highest SII. For the fit-to-target computation, the 'best' combination is the combination that yields the lowest root-mean-square (RMS) difference (in dB) between compression-system gain and the target gain specified by the Cambridge method for loudness equalization (currently called CAMEQ; Moore et al, 1999a). Note that the Cambridge nonlinear algorithm was used because the available software allowed automation of the target gain calculations. Results are not expected to change significantly if other target gains are used.

All computations were performed in floating point arithmetic with custom programs on a digital computer.

Audiograms

In order to evaluate the compression systems over a range of audiological requirements, a large number (957) of audiograms were obtained from the Vanderbilt Bill Wilkerson Center Audiology Clinic, accumulated over a three-year period of everyday operation of the clinic. Figure 1 shows a histogram of the pure-tone average (PTA – threshold averaged across 500, 1000, and 2000 Hz) in dB HL (ANSI, 1996) of the 957 audiograms. They encompass a range of hearing losses from mild (PTA $\leq$40; 41%), to moderate (40 $<$PTA $\leq$70; 51%), to severe-to-profound (PTA $>$70; 8%). Breakdown of the audiogram configurations following the nomenclature of Tables 1 and 2 of Kaplan et al. (1993) is as follows: 65% falling, 31% flat, 3% rising, and 1% not fitting in any of these categories. Average audiograms are shown in Figure 2.

Target gain determination: Maximum SII

For SII calculations, compression target gains were pre-determined in each of the SII 1/3-octave bands. The compression target gains are the real ear aided gain (REAG) required to achieve presumed levels at the eardrum for low- and high-level speech. For high-level speech (the 'loud' category—at 82.3 dB SPL overall—from ANSI 1997) the level at the eardrum was such that the speech peaks were at the upper level of comfortable loudness (ULCL, calculated from audiogram data following Seewald et al, 1996). For low-level speech (the 'quiet' category minus 17 dB—yielding 45 dB SPL overall—from ANSI 1997) the target level at the eardrum was such that the speech minima were at threshold, *or* the speech peaks were at ULCL, whichever level required the lower gain. Speech peaks and minima were assumed to be at +15 and -15 dB re: the long-term RMS level in each band, respectively. Further details of this REAG computation are provided in Appendix 1.

Channel configuration iteration: Maximum SII

An exhaustive search for the channel configuration (i.e. a set of channel crossover frequencies) yielding the maximum SII (averaged across low and high speech levels) for a given audiogram was performed for each possible number of channels N. That is, for a given number of channels $1 \leq N \leq 18$, the SII for each possible N-channel contiguous combination of the eighteen 1/3-octave bands was computed (using the above-mentioned REAG target), and the combination yielding the highest SII was recorded. Note that for each value of N, this resulted in $\frac{2 \cdot 17!}{(17-(N-1))! \cdot (N-1)}$ SII computations per audiogram (the factor of 2 arises from the computations of SII for both low and high speech levels). When summed over all values of N, this resulted in 2^{18} SII computations per audiogram.

For a given channel configuration, the gain for compression channel k and speech level L (G_k^L) was found from the pre-computed band REAGs as follows. First, for high-level speech the channel gain was set to the minimum high-level target REAG across bands in the channel (guaranteeing that all speech peaks were at or below ULCL in each band in the channel). Next, the low-level speech gain was set to the maximum low-level speech REAG across bands in the channel. Low-level speech peak levels were then computed (applying the channel gain to the speech peak level in each SII band within the

Table 2. Minimum number of bands N meeting various SII performance criteria, 'easy' speech case. Values in parentheses in the loss specification cells are number of audiograms in that loss category. For those categories with three or fewer audiograms, the 90th percentile actually includes all audiograms. To use the table, select a criterion (column) and loss category (row), and the value at the row/column intersection gives the number of channels required to meet that criterion for that category. For example, the first column shows the minimum number of bands N guaranteeing that the average (across audiograms in the given category) increase in low-level easy-speech SII when going from N to 18 channels would be ≤0.05. As another example, the eighth column shows the minimum number of channels N guaranteeing that the increase in predicted low-level percent correct when going from N to 18 channels would be ≤2% for 90% of the audiograms in the given category

		Metric	*Low-level SII(18)−SII(N)*				*Low-level %C(18)−%C(N)*			
		Criterion	*0.05*	*0.05*	*0.02*	*0.02*	*5.0*	*5.0*	*2.0*	*2.0*
		Percentile	*Avg.*	*90%*	*Avg.*	*90%*	*Avg.*	*90%*	*Avg.*	*90%*
Loss	Mild	Flat (62)	2	3	3	4	1	1	1	1
		Falling/Rising (332)	3	4	4	6	1	1	1	1
		Other (3)	2	3	3	3	1	1	1	1
	Moderate	Flat (203)	4	5	7	9	1	1	1	1
		Falling/Rising (284)	4	5	6	8	1	1	1	1
		Other (1)	4	4	6	6	1	1	1	1
	Severe	Flat (35)	4	5	7	10	3	8	6	11
		Falling/Rising (34)	3	6	6	9	1	3	3	4
		Other (3)	4	5	8	9	3	3	4	5

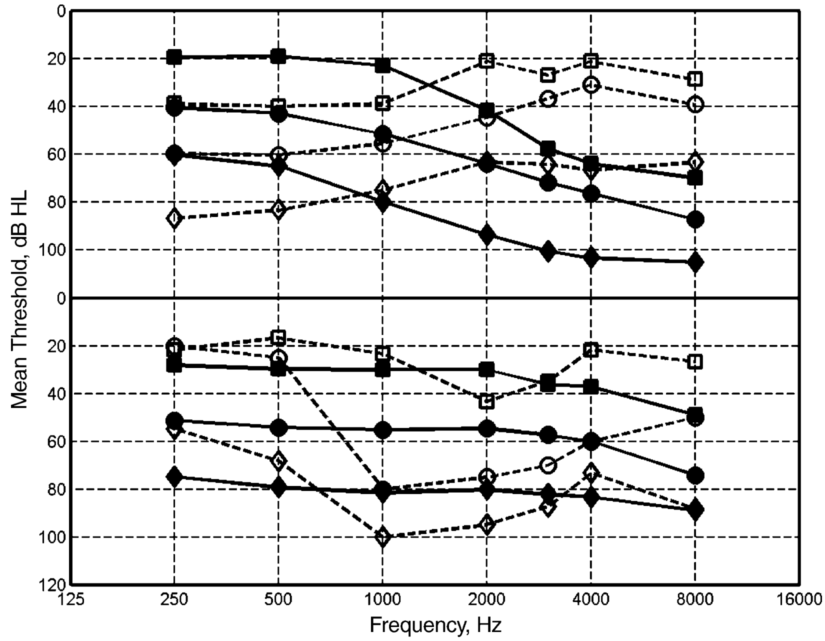

Figure 2. Mean audiograms in each group. The upper panel shows data for falling (solid lines) and rising audiograms (dashed lines), and the lower panel data for flat (solid) and all other audiograms (dashed). In each panel data for mild, moderate, and severe groups are indicated by squares, circles, and diamonds, respectively.

channel), and if any exceeded ULCL the low-level gain was reduced until the highest low-level speech peak in the channel was at ULCL (again guaranteeing that all speech peaks were at or below ULCL in each band in the channel). The compression ratio in the channel was then computed using these low- and high-level gain values and the overall levels in the channel (computed via power summation across the bands within the channel) for the two speech levels. If the compression ratio was greater than 3:1 the low-level-speech channel gain was lowered to the value generating a compression ratio of 3:1; otherwise the channel gain was not changed. After doing this for each channel, the SII was calculated using the determined channel gains, and

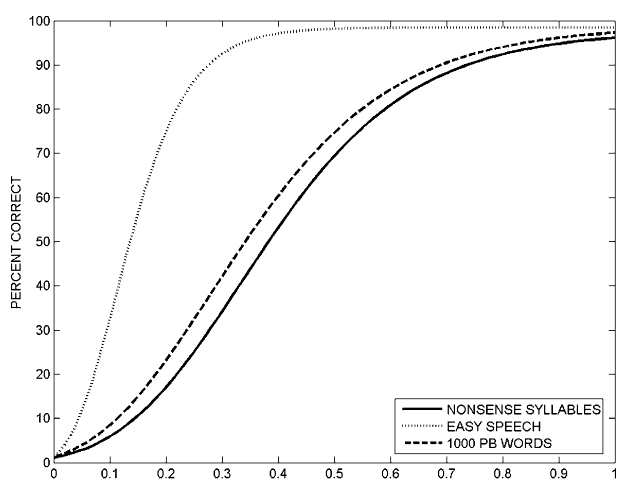

Figure 3. Transfer functions from SII to percent correct. The lines are plots of the polynomials relating SII to percent correct specified in Sherbecoe & Studebaker (1990).

averaged across the two speech levels (details of this calculation can also be found in Appendix 1). To examine how predicted performance varied for different speech types, calculations were done using importance functions (ANSI, 1997) and SII-to-percent-correct functions (Sherbecoe & Studebaker, 1990) for nonsense syllables, 1000 phonetically balanced words, and 'easy' speech (i.e. the 'Familiar sentences' category of Sherbecoe & Studebaker's Table 1). The transfer functions are shown in Figure 3.

For each audiogram and for each possible number of channels, the channel configuration yielding the highest SII averaged across the two presentation levels was recorded. No SII correction factors were used since their use would either leave optimum gain targets unchanged, or create targets that are easier to match with a given channel distribution than non-corrected gains (see the Discussion section for further remarks on this issue).

Target gain determination: Cambridge algorithm

For fit-to-target evaluation, real-ear aided gain was determined using CAMEQ (Moore et al, 1999a). Because the minimum edge frequency allowed by the Cambridge algorithm is 200 Hz, gain targets were those produced by the Cambridge algorithm for a 16-channel system with center frequencies at the 16 highest of the 1/3-octave band procedure of the SII (i.e. 250 Hz and higher). These gain targets were obtained using the Cambridge group's DOS-based software. Specifically, the software-determined values for the *REAG65* (i.e. the REAG for a 65 dB sinusoid in the free field) and the compression ratio (*CR*) for a given channel were used to compute the $REAG_L$ for a given speech level L as follows:

$$REAG_L = \Delta L + REAG65 - \Delta L/CR \qquad (8)$$

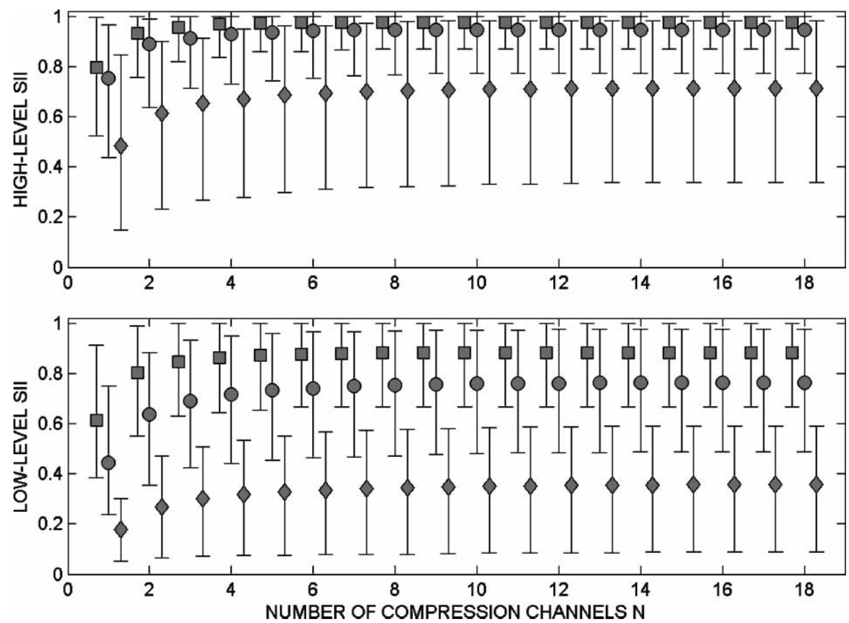

Figure 4. Maximum nonsense syllable SII as a function of channel number N. Symbols represent the mean value (across audiograms) of the maximum SII in mild (squares), moderate (circles), and severe (diamonds) loss groups. Error bars enclose the central 90% of the results across subjects in the respective group. Top and bottom panels are for high- and low-level speech, respectively. Symbols are shifted horizontally for ease of viewing.

$$\Delta L = 65 - L \tag{9}$$

where L is the free-field level in dB of the speech in a given channel determined by power summation of free-field band levels in the channel. The 'Aid' input parameter for the software was 'ITE' and reference microphone position selected to be 'close to the tragus'.

Channel configuration iteration: Best fit to target gain
As with the maximum SII evaluation, an exhaustive search for the channel configuration yielding the best fit to the Cambridge target gains was performed for each possible number of channels. The search was over all possible contiguous combinations of the 16, 1/3-octave crossover frequencies, but in this case the 'best' configuration was that which produced the lowest root-mean-square difference (RMS, computed in dB) between target and channel gains. For a given configuration, setting each channel gain to the mean target gain (averaged in dB) across all bands within the channel yields the lowest mean-square difference.

Results

General results: SII
The nonsense-syllable maximum-SII results for all audiograms are shown in Figure 4. Results for other speech types have a similarly shaped dependence on channel number but are at a higher absolute performance level. (We return to data for other speech types in the Discussion section.) The top and bottom panels show results for high- and low-level speech, respectively. The squares, circles, and diamonds show mean performance across audiograms for the normal-to-mild (PTA $\leq$40), moderate (40 $<$PTA $\leq$70), and severe-to-profound (PTA $>$70) losses, respectively. The ends of the bars enclose the central 90% median $+/-$ 45%) of the results across audiograms for the given group. Symbols are shifted horizontally for ease of viewing.

As expected, maximum achievable SII increases or is approximately constant as the number of channels increases. SII is in general higher for the high-level speech than for the low-level speech. This result is mainly due to the fact that the gain for the low-level speech is reduced for many subjects from its ideal value due to the compression ratio limit. When this limit is removed (not shown) there is less difference between SII for the two speech levels. A remaining, slight, difference is due to the fact that, in general, the gain required for the high-level speech peaks to match ULCL has less variance as a function of frequency (is 'smoother') than that required for the low-level speech minima to reach threshold. This lower variance is in turn due to the fact that ULCL is in general smoother than threshold and the slope of the high-frequency half of the speech spectrum is relatively constant (approximately -10 dB/octave) with overall effort.

The wide range in performance across audiograms reflects the fact that a wide range of PTAs has been used. For a given number of channels, mean SII and mean PTA (in each loss group) vary inversely across loss severity. This is due mainly to the fact that audiograms with higher PTAs have a smaller residual auditory area (RAA, the area between threshold and ULCL) in which to fit the speech dynamic range and thus in general yield smaller SIIs. The limit on maximum compression ratio may also contribute to low-level speech not receiving full audibility.

General results: Fit-to-target
The best-fit-to-target-gain results for all audiograms are shown in Figure 5. The top and bottom panels show results for high- and low-level speech, respectively. The squares, circles, and

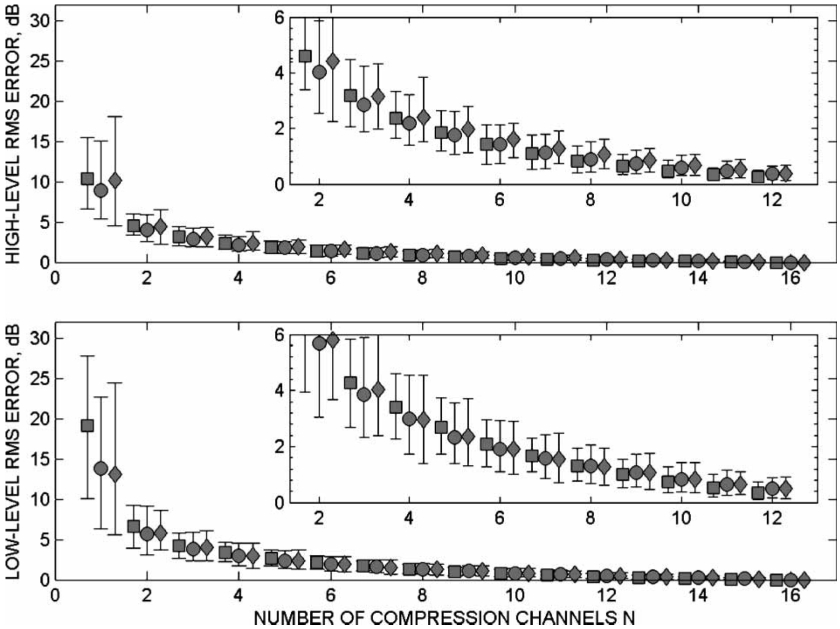

Figure 5. Minimum RMS error as a function of channel number N. Symbols represent the mean value (across audiograms) of the minimum RMS error in mild (squares), moderate (circles), and severe (diamonds) loss groups. Error bars enclose the central 90% of the results across subjects in the respective group. Top and bottom panels are for high- and low-level speech, respectively. Insets show an expanded view of some of the data. Symbols are shifted horizontally for ease of viewing.

diamonds show mean performance across audiograms for the normal-to-mild (PTA $\leq$40), moderate (40 $<$PTA $\leq$70), and severe-to-profound (PTA $>$70) losses, respectively. The ends of the bars enclose the central 90% (median +/ −45%) of the results across audiograms for the given group. Symbols are shifted horizontally for ease of viewing. The insets in Figure 5 give an expanded view of a portion of the data in the given panel.

As expected, minimum achievable RMS error decreases as the number of channels increases. RMS error is lower for the high-level speech than for the low-level speech. Comparisons of Cambridge target gains for the two speech levels show smoothness trends that are similar to those seen in the SII target gains, underlying the lower errors for the higher speech level.

There is an inverse relationship between RMS error and PTA for low-level speech at very low channel numbers due to smoother target gains for higher PTAs at this speech level. For the high-level speech there is not much difference in average RMS error across loss groups because target gains are relatively smooth for this speech level compared to the low-level speech level. Also, target gains differ across loss groups mainly in overall gain, and the optimization procedure is insensitive to overall gain.

Range of SII performance and benefit of additional channels
Although there is a wide range in group-mean SII found in Figure 4, the range of ‘benefit’ of increasing channel number is far smaller, as shown for all audiograms in Figure 6 (as before, squares, circles, and diamonds represent mean values across audiograms showing normal-to-mild, moderate, and severe-to-profound losses, respectively). Benefit is defined as the difference in SII between an 18-channel and an N-channel system, indicating the extent to which performance can be improved by going from an N- to an 18-channel system. The top and bottom panels show benefit for high- and low-level speech, respectively, and the insets show an expanded view of some of the data. The smaller range of benefit is exemplified by, for instance, the fact that for the central 90% of the 957 audiograms, benefit is at or below 0.05 SII by N $=$6 channels. In addition, mean benefit for each loss group is less than 0.05 SII by four channels. Benefit drops off rapidly, with the benefit at N $=$1 being two to three times as large as that at N $=$2, and subsequent ratios slightly smaller.

The pattern of average benefit as a function of loss group for low-level speech seen in Figure 6 (circles at or above squares; diamonds showing slower decrease in benefit as a function of number of channels) is, like absolute performance, driven primarily by PTA via the RAA. For lower PTAs (squares) the RAA is large and the whole speech dynamic range can be fitted in without much shaping required (i.e. with few channels). Thus, benefit is high for low channel numbers and rapidly decreases. For moderate PTAs (circles), the speech dynamic range can also fit the RAA, but it requires more shaping (i.e. more channels) than for lower PTAs. Thus, the benefit starts high, as for lower PTAs, but does not decrease as quickly. For the highest PTAs (diamonds), the RAA can be much smaller than the speech dynamic range, and even the 18-channel SII will be lower than for the other PTA groups. Thus, benefit starts low, and each additional level of shaping (i.e. each additional channel) will fit another, albeit small, bit of the speech dynamic range into the RAA, yielding a slow decrease in benefit with channel number. Target gain smoothness and compression ratio constraints also play a role in the benefit pattern, but more in degree than kind. For example, removing the compression ratio constraint (not shown) allows relatively higher eighteen-channel SIIs for the highest PTAs, but still leaves it difficult to fit the speech dynamic range into that RAA (since it has been

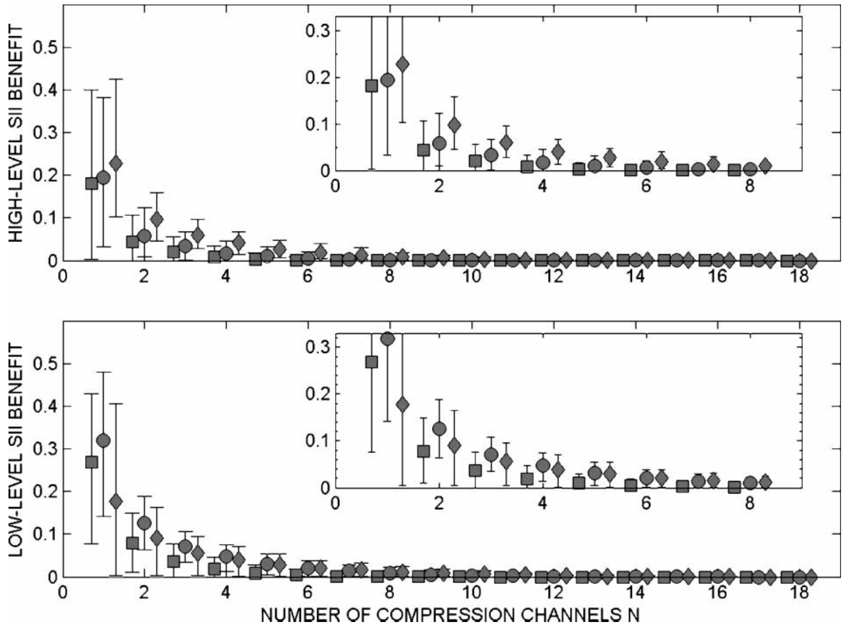

Figure 6. Nonsense syllable SII benefit [SII(18)-SII(N)] as a function of channel number N. Symbols represent the mean value (across audiograms) for the mild (squares), moderate (circles), and severe (diamonds) loss groups. Top and bottom panels are for high- and low-level speech, respectively. Insets show an expanded view of some of the data. Symbols are shifted horizontally for ease of viewing.

assumed that the compression does not change the dynamic range of the speech signal). This causes the benefit for low-level speech without a compression ratio constraint to look like that for high-level speech with a constraint: similar in pattern to that for low-level speech with a constraint, but different in amount of benefit.

Channel SII benefit as a function of degree and configuration of loss

The pattern of benefit as a function of number of channels and PTA is similar, on average, across the different configuration types. This is demonstrated in Figure 7. Each panel shows low-level-speech benefit as a function of PTA for a single channel number, and for all audiograms with a given configuration or group of configurations. The left column is for flat losses, the middle column for rising and falling losses, and the right column for all other loss configurations. Data are shown for N = 1, 2, 4, and 8 in the top, second, third, and bottom rows, respectively. The general pattern of benefit as a function of PTA and channel number is the same across configurations (i.e. across columns): low and high PTAs show less benefit than moderate PTAs, and benefit is almost negligible by eight channels. When plotted in a similar fashion, the same is true for the high-level speech benefit.

Channel percent-correct benefit as a function of degree of loss

The SII can be converted to percent correct (%C), and benefit computed for this measure as well. This was done using the formulae described by Sherbecoe & Studebaker (1990). Figure 8 shows benefit (for nonsense syllables) in %C as a function of number of channels for low-level (bottom panel) and high-level speech (top panel) using the same conventions as in Figure 6. The pattern for the severe-loss group (diamonds) is similar to that for SII benefit, but for groups with mild or moderate loss %C benefit decreases much more quickly than does SII benefit. For example, mean benefit for both speech levels at five channels is less than 6, 3, and 1% for severe, moderate, and mild loss groups, respectively. This is due both to the relatively high SII values (>0.7) at low channel numbers, and the relatively low SII benefits at higher channel numbers, for the latter two groups. These factors combine with the fact that the nonsense syllable %C is high (80%) for an SII of 0.7 and the %C as a function of SII curve is flattening out at high SIIs to produce quickly diminishing average %C benefit (a ceiling effect) as a function of channel number for the mild and moderate groups. The rapid decrease in %C benefit with channel number is consistent with the results of Crain and Yund (1995) in their SHAPED condition, in which no significant effect on %C of having two through 31 channels was found for their moderately-impaired subjects. In our high-level (82 dB SPL) condition (comparable to Crain & Yund's 75 dB SPL level) mean %C benefit for moderately-impaired listeners is already less than 2% by two channels. Thus any significant change in %C for more than two channels would be difficult to find. This was also found for most subjects with the other speech types. That is, benefit decreased rapidly for these other speech types, since lower SII values are sufficient for high %C.

Two studies have measured percent correct over a wide range of channel number (with 'syllabic' compression systems) and found results that differ from each other and from what might be expected based on our audibility computations. Plomp (1994, Figure 7) found that for his moderately impaired subjects, sentence percent correct *decreased* with increasing channel number, even for the relatively low compression ratio of 2.0. Yund and Buckles (1995) found that for their mild-to-moderately impaired subjects, average nonsense syllable percent correct for a nominal presentation level of 85 dB SPL *increased* 3 to 4% when going from four to 16 channels,

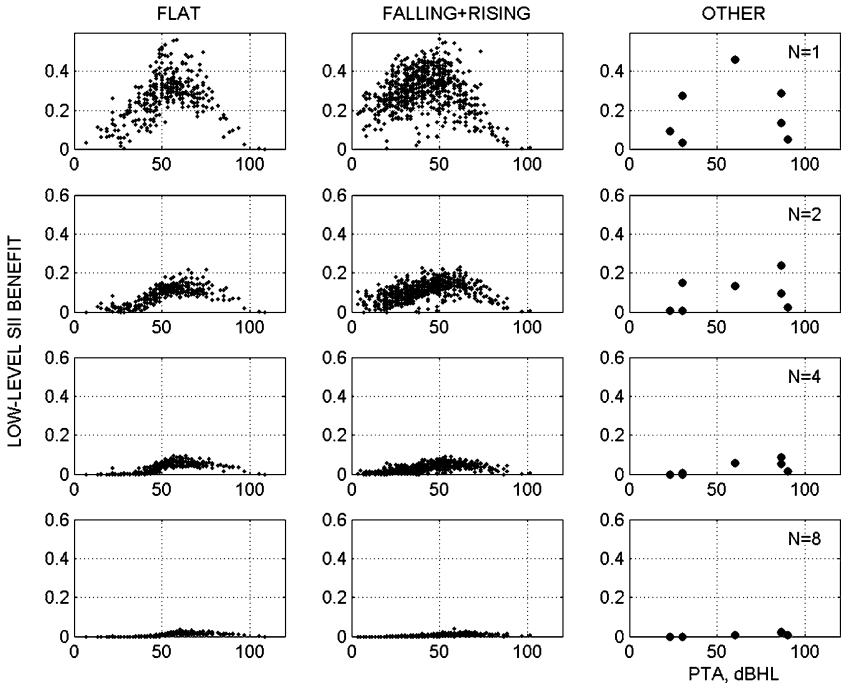

Figure 7. Low-level nonsense syllable SII benefit [SII(18)-SII(N)] for all audiograms as a function of loss degree, loss configuration, and channel number N. Each point is the result for a single audiogram. Results are shown for flat (leftmost column), falling and rising (center column), and all other configurations (rightmost column), and for N = 1 (top row), N = 2 (second row), N = 4 (third row), and N = 8 (bottom row).

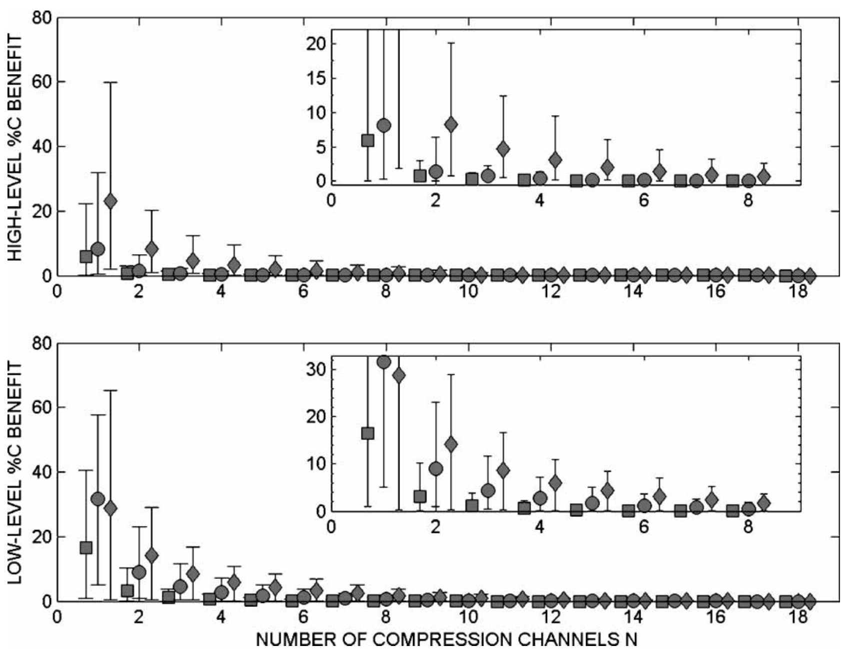

Figure 8. Nonsense syllable percent correct benefit [%C(18)–%C(N)] as a function of channel number N. Symbols represent the mean value (across audiograms) of the benefit in mild (squares), moderate (circles), and severe (diamonds) loss groups. Error bars enclose the central 90% of the results across subjects in the respective group. Top and bottom panels are for high- and low-level speech, respectively. Insets show an expanded view of some of the data. Symbols are shifted horizontally for ease of viewing.

Table 3. As in Table 2, but for 1000 phonetically balanced words

	Metric	Low-level SII(18)−SII(N)				ow-level %C(18)−%C(N)			
	Criterion	*0.05*	*0.05*	*0.02*	*0.02*	*5.0*	*5.0*	*2.0*	*2.0*
	Percentile	*Avg.*	*90%*	*Avg.*	*90%*	*Avg.*	*90%*	*Avg.*	*90%*
Loss Mild	Flat (62)	2	3	3	5	1	1	1	2
	Falling/Rising (332)	3	4	4	6	2	2	2	3
	Other (3)	2	3	3	4	1	2	2	2
Moderate	Flat (203)	4	6	7	11	2	3	4	7
	Falling/Rising (284)	4	6	7	9	3	4	4	6
	Other (1)	5	5	6	6	4	4	5	5
Severe	Flat (35)	4	6	8	11	5	9	9	14
	Falling/Rising (34)	3	6	6	10	5	7	9	13
	Other (3)	4	5	7	8	6	6	9	9

regardless of signal-to-noise ratio (varying from −5 to +15 dB). When run with Yund and Buckles' subjects' audiograms, the maximum SII routine used here predicted (not shown) an average benefit of less than 0.5% when going from four to 18 channels for high-level speech. One major difference between these three studies is the way in which the gain was set in each channel. Plomp used a different pre-emphasis filter (i.e. a filter applied before applying compression) for each subject. This filter placed the speech spectrum halfway between threshold and ULCL for each subject. Similarly, the current analysis optimized the placement of speech for each channel configuration by varying crossover frequencies and channel gain to maximize the SII for each number of channels. Yund and Buckles used a suboptimal method, comprising a single fixed set of channel crossover frequencies (per channel number N) for all subjects and channel gain that was based on threshold at only the center frequency of each channel. This may have caused lower performance than possible at lower channel numbers, and thus a need for more channels to generate the best performance possible. Though the investigations differ in other details as well, this only emphasizes the point that what is ostensibly a comparison of the effect of number of channels, only, is in reality a comparison of the combined effects of channel number, gain prescription, and other factors. Avoiding these combined effects is one of the main advantages of the procedure used here.

Crossover frequencies

In order to understand the implications of this work for hearing aid filter-bank design, we evaluated histograms of the crossover frequencies as a function of number of channels N. These histograms were often bimodal, with the two modes often separated widely in frequency (e.g., ten SII bands apart), and could change greatly depending upon the number of channels, and upon which channel was being examined. It is thus difficult to describe crossover-frequency distribution in terms of 'typical' values. In order to cover the range of crossover frequency configurations found here, then, a filter bank configurable to the crossover configurations comprising the 18 one-third-octave bands of the SII specification would need to be implemented. This should not be difficult given current technological capabilities.

Discussion

Application to hearing aid design and evaluation

As an aid to applying the results of the present work, Tables 2–5 show the minimum number of channels meeting different example criteria for 'easy' speech, 1000 phonetically balanced (PB) words, nonsense syllables, and RMS error, respectively, broken down by hearing loss type and degree. The data are shown for the low-level speech condition only, since this required more channels than did the high-level speech condition for any given criterion. The tables show the wide variability one obtains using different criteria. For instance, because of the rapid rise in %C with SII for 'easy' speech (Table 2), there is little benefit beyond one channel for all but the severe audiograms for the %C criteria. For the same loss groups and criteria, minimum channel number ranges from 1–6 for the PB word case (Table 3) and from 1–8 channels for the nonsense syllable case (Table 4). As

Table 4. As in Table 2, but for nonsense syllables

		Metric	*Low-level SII(18)−SII(N)*				*ow-level %C(18)−%C(N)*			
		Criterion	*0.05*	*0.05*	*0.02*	*0.02*	*5.0*	*5.0*	*2.0*	*2.0*
		Percentile	*Avg.*	*90%*	*Avg.*	*90%*	*Avg.*	*90%*	*Avg.*	*90%*
Loss	Mild	Flat (62)	2	3	3	4	1	2	2	2
		Falling/Rising (332)	3	4	5	6	2	3	3	4
		Other (3)	3	3	3	4	1	2	2	3
	Moderate	Flat (203)	4	6	7	9	3	5	5	7
		Falling/Rising (284)	4	5	7	8	4	5	6	8
		Other (1)	5	5	7	7	4	4	6	6
	Severe	Flat (35)	4	6	7	9	5	7	8	10
		Falling/Rising (34)	3	5	6	8	5	6	7	9
		Other (3)	5	6	8	9	6	8	9	11

Table 5. As in Table 2, but for RMS-error performance criteria

		Metric	*RMS Error(N) (dB)*			
		Criterion	*5.0*	*5.0*	*2.5*	*2.5*
		Percentile	*Avg.*	*90%*	*Avg.*	*90%*
Loss	Mild	Flat (62)	2	3	5	6
		Falling/Rising (332)	3	4	6	7
		Other (3)	3	4	5	7
	Moderate	Flat (203)	2	3	4	5
		Falling/Rising (284)	3	4	6	7
		Other (1)	4	4	7	7
	Severe	Flat (35)	2	3	4	6
		Falling/Rising (34)	3	4	6	7
		Other (3)	4	4	7	8

discussed earlier, audiograms with higher PTAs generally require more channels to meet a given criterion.

The application of these results to the design of compression systems, as well as to the evaluation of existing compression hearing aids, is limited by certain aspects of the design. The most important are:

1. *Speech recognition criterion*. It has long been known that the hearing aid frequency response that maximizes intelligibility is not necessarily the one that listeners find most agreeable (Punch & Beck, 1986; Rankovic, 1991, 1995; Skinner, 1980) This same distinction might apply to compression channel number, although the limited published data available generally support the conclusion that perceived quality does not increase with number of compression channels (e.g. Keidser & Grant, 2001).
2. *Rectangular analysis bands and compression channels*. In practice, the analysis bands and compression channels in actual hearing aids are non-rectangular, and may even have, by design, very gradual filter slopes. The more gradual the filter slopes of the bands and channels, the more correlated the compression will be across channels. In these instances, a compression system with a large nominal number of channels may be functioning more similarly to one of the low-channel-number systems described in the current work.
3. *Evaluation based on predicted performance in quiet*. It is possible that the benefit as a function of number of compression channels might be different—and possibly greater—in noisy situations. For example, Moore et al (1999b) found that an 8-channel system yielded on average 1 dB lower speech reception thresholds in noise than did a 4-channel system when the noise was temporally modulated and contained spectral gaps. In addition, a many-channel system would be more likely to lead to increased intelligibility by reducing the gain in the frequency region of an intense narrowband background sound. (e.g. van Dijkhuizen et al, 1991; Rankovic et al, 1992; Rankovic, 1995). This is due to the fact that the many-channel system would be better able to attenuate in the required narrow band without affecting speech outside the band. Expectation of such benefits in the field, however, needs to consider whether or not such conditions occur naturally. In addition, given the peculiar conditions generating such benefits, they are easily identified and acted upon by signal processing operating in parallel with the compression. For the case of steady-state, broadband noise, as long as audibility is controlled by absolute threshold or within-band noise (i.e. there is no excessive spread of masking) after gain application, the maximum-SII gain function determined here would also yield the maximum SII in the noise (although the maximum SII value would be lower whenever the noise spectrum is above threshold). This is because the quiet-derived gain function yields the maximum possible amount of audible speech, and the addition of the noise simply reduces that maximum possible amount. If conditions are such that the presence of the noise affects the loudness constraint (e.g. if the speech alone were not too loud, but the combination of speech and noise were), then performance as a function of number of channels would depend on the details of how loudness is determined in these conditions, and might or might not require more channels for a given level of performance than found here.
4. *Time constants not evaluated*. Because the SII and target analyses performed in the current work do not take dynamic changes in gain into account, the effects of time constant are not reflected in the analyses. As noted previously, a compression system with fast time constants might be better at reconfiguring gain for changes within an analysis band. Thus, for fast compression systems maintaining as high or higher sound quality as obtained with very slow systems, it is expected that the benefit of increasing number of channels would decrease even more rapidly than reported here.
5. *Only compression evaluated*. The current work looked only at the relation between channel number and certain measures of compression performance. Compression may not be the only channel-based process in a hearing aid. To the degree that other processes depend on the same frequency analysis as the compression system, the benefit as a function of channel number of the other processes should obviously be considered in the design or evaluation of the hearing aid.

Also, a major assumption of the present work is that the performance based on SII predictions is directly related to actual performance. Although published evaluations of the accuracy of predictions of the SII and related indices are equivocal, the pattern of the majority of errors is such that unmodified SII computations in fact do represent a limit on performance for an overwhelming number of audiogram types. This is because the suggested modifications to the SII (to correct for, e.g. erroneous use of measured thresholds; 'rollover' due to high presentation levels; inability to make use of audible speech information, such as in a 'dead region'; or improperly represented masking or spread of masking) actually produce target gains that are as smooth or smoother than those produced with the unmodified SII. The smoother a target gain is, the better will be the fit with a given number of channels. Thus, the computations performed here represent the strictest of SII-based tests.

Summary

The computations performed here isolated the effects of compression channel number on predicted speech intelligibility and fit-to-gain-target measures from other compression evaluation parameters. These computations represent an upper limit on predicted speech intelligibility for slow-acting compression, in that performance is optimized for each channel number and audiogram, and most common predictors of speech intelligibility will predict either similar or diminished performance and benefit compared to the predictor used here. Use of this computational method allowed the evaluation of performance as a function of number of channels for a large number of audiograms. Tables 1–4 summarize the minimum number of channels meeting certain example criteria broken down by loss degree and configuration.

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Appendix 1. SII and maximum-SII target REAG computations:

This appendix describes in detail the computations of SII and real-ear aided gain REAG for the maximum-SII target gain condition. If E is the band RMS speech level at the eardrum, and M is the band RMS speech level at a reference microphone, the REAG in each band for the high and low speech levels is computed as follows:

$$REAG = E - M \tag{1}$$

$$E = \begin{cases} ULCL - 15, & loud \\ \theta + 15 & quiet \end{cases} \tag{2}$$

$$M = \begin{cases} FF^{l} + FF2MIC, & loud \\ FF^{q} + FF2MIC, & quiet \end{cases} \tag{3}$$

$$REAG = \begin{cases} ULCL - 15 - FF^{l} - FF2MIC, & loud \\ \theta + 15 - FF^{q} - FF2MIC, & quiet \end{cases} \tag{4}$$

where all quantities are expressed in dB and at the frequencies of the ANSI (1997) specification; $ULCL$ and θ are the ULCL and the hearing threshold expressed as a noise spectrum level at the eardrum, respectively; FF^{l} and FF^{q} are the free-field levels of the high- and low-level speech, respectively; $FF2MIC$ is the free-field-to-reference-microphone transfer function; and the speech peaks and speech minima are assumed to be 15 dB above and below the speech RMS level in the band, respectively (ANSI, 1997).
θ at the eardrum is found using

$$\theta = \theta_{dBHL} + RINS + FF2ED, \tag{6}$$

where θ_{dBHL} is the absolute threshold in dB HL, and $RINS$ and $FF2ED$ are the reference internal noise spectrum level in the free-field and free-field-to-eardrum transfer function, respectively.
The SII SII^{L} for a given speech level was computed as:

$$SII^{L} = \sum_{k=1}^{N} \sum_{i=FLk}^{FHk} I_i \cdot \min(30,\ \max(0,\ FF_i^{L} + FF2MIC_i + G_k^{L} + 15 - \theta_i))/30 \tag{7}$$

where k indexes the compression channels; i indexes the bands within a channel; G_k^L is the gain in channel k for speech level L; FHk and FLk are the index values of the highest and lowest bands in the compression channel; FF_i^L, $FF2MIC_i$, and θ_i are the i^{th} band free-field speech level, free-field-to-microphone transfer function value, and threshold noise spectrum level at the eardrum, respectively; and I_i are the nonsense syllable 1/3-octave importance values from the SII specification.

Reference data for these calculations were obtained as follows. For the SII calculations the speech levels were based on the 1/3-octave band data of the SII specification (ANSI 1997, Table 3). Low-level speech levels equaled the SII spectrum for 'normal' vocal effort minus 17.35 (to reduce the 62.35 dB overall level to 45 dB), and high speech levels equaled the SII 'loud' vocal effort spectrum. *RINS* and *FF2ED* were also taken from Table 3 of the SII specification. *FF2MIC* is the free-field-to-ITE-microphone data from Bentler & Pavlovic (1989). Whenever reference data were not available at the 1/3-octave frequencies, linear interpolation and extrapolation were performed on a dB/log-frequency basis.

To find $ULCL$, the audiogram data were first limited to a maximum of 110 dB HL and interpolated and extrapolated to the 1/3-octave frequencies, and then a ULCL table (Table A1, specifying the ULCL in eardrum tonal dB SPL as a function of threshold in dB HL and frequency in Hz) was 2D linearly interpolated onto the 1/3-octave frequencies internal to the table. After linear extrapolation from the resulting ULCL values onto the remaining 1/3-octave frequencies, a correction factor was subtracted to convert ULCL into dB HL. Finally, the spectrum level of a noise in each band representing ULCL at the eardrum was found by adding the ANSI reference internal noise spectrum level and FF2ED in dB to ULCL in dB HL. The data in Table A1 are taken from the 'Auditory Area' window of the DSL® software (Seewald et al, 1996) in response to entering an audiogram with the given loss θ in dB HL for each audiogram frequency. The correction factor converting ULCL from eardrum dB SPL to dB HL was taken from the same DSL window using data showing threshold in eardrum dB SPL for an audiogram that was 0 dB HL at all frequencies.

Table A1. Predicted ULCL in eardrum SPL given threshold θ in dB HL at the associated frequency in Hz

	Frequency, Hz								
θ, dB HL	*250*	*500*	*750*	*1000*	*1500*	*2000*	*3000*	*4000*	*6000*
0	94	105	101	99	99	101	100	99	97
5	95	102	101	100	100	102	101	100	98
10	95	102	102	101	100	103	102	101	98
15	96	103	102	101	101	104	104	103	99
20	96	104	103	103	103	105	105	104	101
25	97	105	104	104	104	107	107	106	102
30	98	106	106	106	106	108	109	107	104
35	100	108	107	107	108	110	111	109	106
40	102	109	109	109	109	112	113	111	107
45	104	111	111	111	111	114	115	113	110
50	106	113	113	113	113	116	117	115	112
55	108	115	115	115	116	118	119	117	114
60	111	117	117	117	118	120	121	119	116
65	111	117	117	117	118	120	121	119	116
70	117	121	121	121	122	124	125	124	120
75	120	123	123	123	124	126	127	126	122
80	123	126	125	125	126	128	129	128	124
85	126	128	127	127	128	130	131	130	126
90	129	130	129	129	130	131	133	131	128
95	131	131	131	130	131	133	134	133	129
100	132	133	132	132	133	134	135	135	130
105	133	135	134	133	134	135	136	136	131
110	133	136	135	134	135	136	137	137	132