Model-based loudness compensation for broad- and narrow-band signals

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A fundamental problem when attempting to restore loudness perception in hearing-impaired listeners are differences in the loudness perception of narrow- and broad-band signals when compared to normal-hearing listeners. Here, a multi-channel dynamic compression algorithm is presented where the signal-to-masking ratio (SMR) is used to modify the channel gain function. Result 1: The evaluation of this approach using a loudness model showed that the loudness perception of hearing-impaired listeners can be restored to the loudness perception of a normal-hearing listener for signals with different bandwidths. Result 2: Inconsistencies between the individual measured loudness function using the categorical loudness scaling procedure and the model predictions were found. The available model parameters, being i) hearing threshold level, ii) outer, and iii) inner hair-cell loss, were not sufficient to fit the model to the individual narrow-band loudness perceptions.

INTRODUCTION

Loudness perception of hearing-impaired (HI) listeners differs from the loudness perception of normal-hearing (NH) listeners. Typically, HI listeners show increased hearing threshold levels (HTL) whereas uncomfortable loudness levels (UCL) remain at the same level as in NH listeners (Bentler and Cooley, 2001). Therefore, to restore loudness perception in HI listeners, a compression algorithm is required which applies the appropriate gain for signals with low amplitudes and reduces the gain for signals with high amplitudes. The individual narrow-band loudness perception can be measured using categorical loudness scaling (CLS; Brand and Hohmann, 2002). The result of the CLS procedure is a loudness function which maps the signal level to the perceived loudness category. Level-dependent gain functions for restoring narrow-band loudness perception with a compression algorithm can be derived when comparing the measured loudness function with the average NH loudness function for the same signal (compare Fig. 3c). It is known that the gain required to restore the narrow-band loudness perception in a multi-channel dynamic compression algorithm leads to overly high gains for broad-band signals (Latzel *et al.*, 2004), resulting in too high loudness impressions. Using both signal types in a current loudness model clarifies why different gain values for narrow- and broad-band signals are required.

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Figure 1a shows different channel gain functions required for restoring the specific loudness perception using the loudness model of Chen *et al.* (2011) and the model of Moore and Glasberg (1997). The specific loudness of Bark channel no. 8 (920-1080 Hz) was calculated for a NH listener and a HI listener having a 50-dB flat hearing loss with standard model parameter settings of 80% outer (OHC), and 20% inner hair-cell (IHC) loss. The channel gain in dB required to restore specific loudness in this channel to normal was calculated for a 1/3-octave, low-noise noise stimulus (LNN) centred at 1 kHz, and for a stationary speech-shaped noise (IFnoise) generated from the international speech test signal (ISTS; Holube *et al.*, 2010) as a function of input level. The estimated gain for restoring the narrow-band loudness function differs by more than 10 dB between both loudness models. The difference between the narrow- and broad-band gain function required to restore the specific loudness is shown in Fig. 1b. The model of Chen *et al.* (2011) estimates a gain reduction of up to 9 dB for the broad-band signal for medium signal levels. Using the model of Moore and Glasberg (1997), only 2-4 dB of gain reduction is predicted for high signal levels.

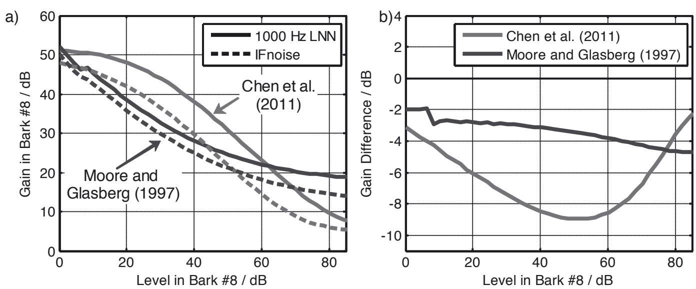

Fig. 1: a) Example of model calculations for the different channel gain functions required in a multi-channel dynamic compression algorithm to restore specific loudness at 1 kHz for a narrow-band noise and broad-band noise (IFnoise). b) Difference between the gain functions from a) showing that both models estimate different channel gain reductions to restore normal loudness perception in HI when a broad-band signal is presented.

It can be concluded from the model calculation that a multi-channel dynamic compression algorithm that analyses and processes the input signal independently in each frequency channel is not capable of applying the correct gain for restoring loudness of both narrow-band and broad-band sounds. To restore loudness using a multi-channel dynamic compression algorithm some measure of the actual signal's bandwidths is required to control the compressive gain functions depending on the signal's bandwidth. In this study, the signal-to-masking ratio (SMR) calculated in each processing channel as an estimator of the signal's bandwidth is introduced. The calculation of the SMR and its integration in a compression algorithm is described in the next section. To demonstrate the properties of the suggested approach, the loudness model of Chen *et al.* (2011) is used in the following.

DYNAMIC COMPRESSION ALGORITHM

A multi-channel compression algorithm was implemented in the frequency domain using an overlap-add (FFT, sampling rate 22 kHz, frame length 408 samples) processing scheme. The signal level is calculated for each of the 24 Bark-spaced channels formed by adding up the squared magnitudes of the corresponding FFT-bins. According to the excitation pattern calculation proposed by Moore and Glasberg (1997), the masking of each Bark-channel on the neighbouring Bark-channels is calculated. Instead of the quite complex calculation scheme proposed by Moore and Glasberg (1997) a more efficient approximate method to calculate the masking patterns was used. Based on the channel levels and the masking slopes, the SMR for each channel as an estimator of the signal's bandwidth was calculated. As shown in the left figure of Fig. 2, we define the SMR to be the difference in dB between the channel level and the maximum masking level caused by all other channels (dashed lines). In the left panel of Fig. 2 the SMR is about 10 dB, which corresponds to the value of the upward masking slopes (10 dB/Bark at medium overall level) for signals having the same level in each Bark band (i.e., uniform-exciting noise, UEN; Fastl and Zwicker, 2007).

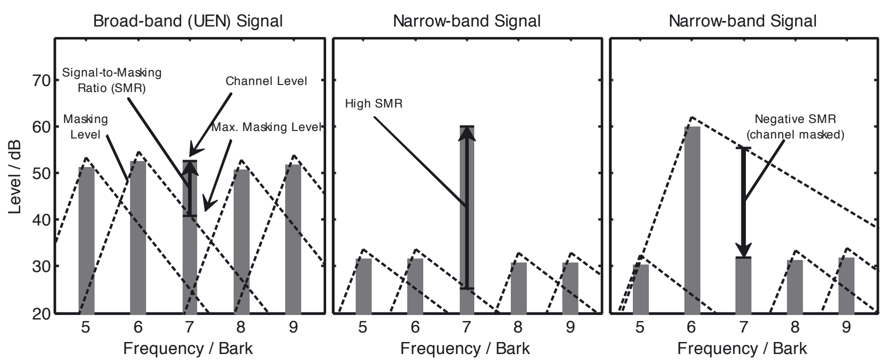

Fig. 2: Calculation of the SMR. The SMR is the difference in dB between the channel level and the maximum masking level. Left panel: A broad-band signal with equal channel levels produces a SMR of about 10 dB at medium input levels. Middle panel: A narrow-band signal produces a high on-frequency SMR value, and (right panel) negative SMR values in the neighbouring channels.

The middle and right panels of Fig. 2 show the calculation of the SMR for a narrow-band signal. Most of the signal energy falls into channel #7 and the masking levels of the other channels are well below the signal level. Accordingly, the SMR value in channel #7 is high. If the narrow-band signal falls into channel #6 as shown in the right panel, the masking level of channel #6 towards channel #7 is much higher than the channel level in channel #7. Hence, the SMR in channel #7 is negative, meaning that this channel will not be perceived and thus should not be amplified by the compressor. In summary, high SMR values indicate a narrow-band signal prominent in the respective Bark-channel, low SMR values indicate a broad-band signal, and negative SMR values indicate signal components that are not perceived.

In the next step the SMR was integrated in a dynamic compression scheme as a major control parameter to adapt the channel gain based on the signal's bandwidth parametrically. This is achieved by modifying the channel level which is used as the input level to the channel gain function. The channel's gain functions are initially set to restore the narrow-band loudness perception. This corresponds to the 'LoudFit' fitting rationale by Herzke and Hohmann (2005) which will serve in this work as a comparison fitting rationale. Figure 3a shows examples for the SMR-dependent modification of the channel level. For high SMR values (SMR > 20 dB) it is assumed that the signal is narrow-band and therefore no further modification to the channel level is applied. In contrast, broad-band signals lead to low SMR values and the estimated 'effective' channel level is increased with decreasing SMR. The 'effective' channel level is used as input to the channel gain function. Since the SMR-dependent modification is always positive, this modification leads to a reduced amount of gain due to the steeper loudness function of HI listeners. This is illustrated in Fig. 3c for the loudness functions derived from categorical loudness scaling. The amount of SMR-dependent level modification is adapted to the individual loudness perception.

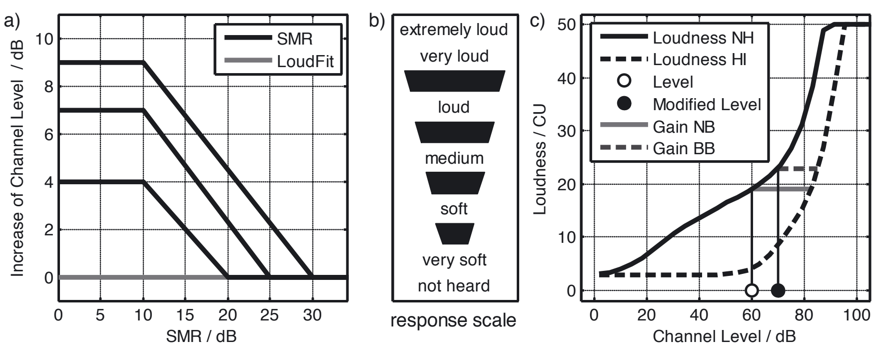

Fig. 3: a) Increase of channel level depending on the SMR. The amount of level increase can be individually adjusted (4, 7, and 10 dB in this example). b) Response scale of the CLS procedure. c) Simulated narrow-band loudness functions of a NH and a HI listener for CLS. According to the 'LoudFit' fitting procedure, the gain to restore the loudness of a 60 dB narrow-band signal ('Gain NB') is approximately 23 dB. For a broad-band signal the channel level is increased ('effective' level), hence the applied gain ('Gain BB') is reduced from approx. 23 to 15 dB for a broadband signal having the same channel level as a narrow-band signal.

EVALUATION OF THE SMR-BASED COMPRESSION ALGORITHM

To evaluate the basic properties of the proposed algorithm, the loudness model of Chen *et al.* (2011) was used and two hearing losses where simulated: a flat 50-dB hearing loss (FHL) and a sloping haring loss (SHL; audiograms are shown in the middle panel of Fig. 4 and 5). The model's standard parameters of 80% OHC loss and 20% IHC loss were used for the FHL und the SHL. To compare the results with results of the CLS procedure, the transformation from the model output in sone to the CLS response scale in categorical units (CU) according to Heeren *et al.* (2013) was used. Uniform-exciting noise (UEN), i.e., noise that produces the same channel level in each bark band (Fastl and Zwicker, 2007) was used as a test signal. The UEN was limited to bandwidths of 1, 5, and 15 Bark centred at 8.5 Bark (1000 Hz) and is accordingly referred to as UEN1, UEN5, and UEN15. The aim was to restore the NH loudness function for all test signals and hearing losses using the SMR algorithm.

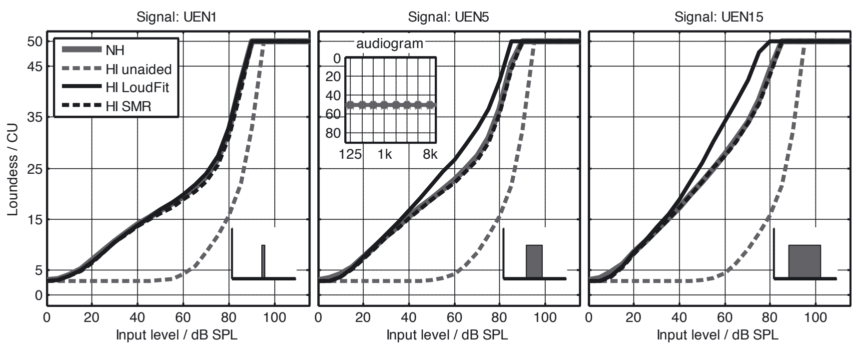

Fig. 4: Loudness functions for a NH and a HI listener with a flat 50-dB hearing loss for UEN with a bandwidth of 1, 5, and 15 Bark, left to right panels, respectively. The black solid line and the black dashed line show the aided loudness function using the fitting rationale 'LoudFit' and the proposed SMR approach. It can be observed that the LoudFit rationale leads to a too high loudness sensation for broad-band signals (i.e., the UEN5 and UEN15 signal shown in the middle and right panel, respectively), whereas the SMR algorithm is able restore the HI loudness perception to NH perception for all signals independent of bandwidth.

In Fig. 4 and 5 the unaided (grey dashed line) and aided loudness (black curves) functions for the modelled FHL and SHL are shown. Normal-hearing loudness perception (indicated by the solid grey curve) was the target for the two aided conditions tested: aided according to the 'LoudFit' rationale (black line) and aided with the SMR-algorithm (black dashed line). Figure 4 and 5 show that the 'LoudFit' fitting rationale as well as the SMR algorithm were able to compensate the loudness perception for the narrow-band signal UEN1 (left panels). For broader signals like UEN5 and UEN15 (middle and right panel, respectively) the 'LoudFit' procedure applied too much gain resulting in a higher loudness perception for the HI when compared to NH, especially at medium levels.

In contrast, the SMR algorithm was able to correctly reduce the gain for broad-band signals and in consequence to restore HI's loudness perception to normal independent of bandwidth. Very similar results were obtained in Fig. 5 for the SHL simulations. Again, too much gain was applied for broad-band signals when signals were processed according to the 'LoudFit' rationale, whereas the SMR algorithm again restored normal loudness for all signals. Here, the SMR-dependent level modification was adjusted to be 4 dB for signals having a low SMR value (lowest function in Fig. 3a for both hearing losses tested. However, depending on the individual hearing loss or preference other modifications might be required. To find those individual settings of the SMR modification, information about the individual perception of broad-band signals is required. This information can be derived from individualized loudness models (as it is done for two types of hearing-impaired here) or by CLS measurements of broad-band signals carried out in addition to the standard clinical procedure of CLS measures with narrow-band signals.

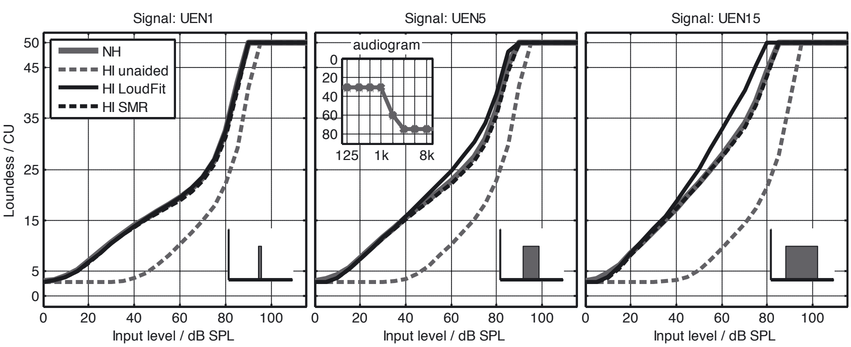

Fig. 5: same as Fig. 4 but for a listener with a sloping hearing loss (SHL).

INDIVIDULIZATION OF THE LOUDNESS MODEL

As pointed out above, the individual adjustment of the SMR algorithm requires information about the individual loudness perception of narrow-band signals over frequency to derive frequency-dependent gain functions as well as information about the perception of broad-band signals to adjust the SMR-based modification. While the CLS measurement with narrow-band signals becomes more and more clinical practice, information about the perception of broad-band sounds is typically not collected. To close this gap, a loudness model could be used, which can be individualized based on data from CLS measurements with narrow-band stimuli to predict the individual broad-band loudness perception. The loudness model of Chen *et al.* (2011) for NH and HI listeners provides the parameters IHC-related and OHC-related hearing loss to individualize the model predictions, whereas the sum of both losses equals approximately the total hearing loss for hearing losses below 60 dB HL (Chen and Hu, 2013).

Following the idea to adjust the loudness model to predict individual CLS data measured with narrow-band stimuli, the model of Chen *et al.* (2011) was adjusted using the IHC and OHC parameters. The result is shown in Fig. 6. Grey curves show variations of the IHC and OHC parameters, whereas thick curves show loudness scaling data measured in two HI listeners having the same hearing threshold of 45 dB HL at 2 kHz but different loudness perception above threshold. A systematic variation of OHC/IHC loss configurations under the constraint of a fixed audiometric threshold (grey lines in Fig. 6) showed that the loudness model was not able to predict the loudness perception of the two HI listeners at all input levels. Therefore, it does not seem meaningful to use the loudness model of Chen *et al.* (2011) to model individual loudness perception of broad-band signals as it is needed for individual adjustment of the SMR algorithm.

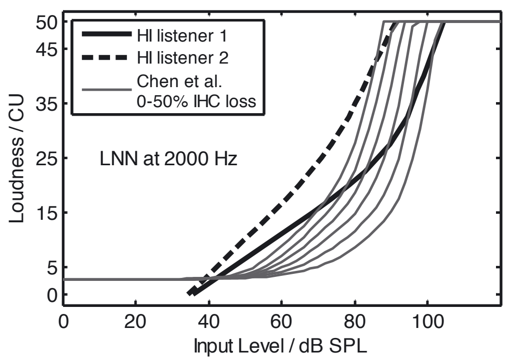

Fig. 6: Loudness functions of two HI listeners derived from CLS measurements with a narrow-band noise signal centred at 2 kHz. Both listeners had a hearing threshold of 45 dB HL. The modelled loudness perception (grey lines) does not match the measured loudness function independent of the possible parameter configuration.

SUMMARY AND CONCLUSION

A multi-channel dynamic compression algorithm for restoring the loudness perception of HI listeners was proposed, which is capable to restore normal loudness perception for narrow-band and broad-band stimuli. The algorithm uses the signal-to-masking ratio (SMR) to modify the level in each processing channel. This 'effective' channel level is then used to determine the channel gain. The evaluation of the algorithm with two simulated HI listeners (flat and sloping hearing loss) using the recent loudness model by Chen *et al.* (2011) showed that the SMR approach is able to restore loudness perception for narrow- and broad-band signals. Thereby, the SMR algorithm requires information about the individual loudness perception of broad-band signals. A first approach to gather this information from predictions of the loudness model of Chen *et al.* (2011) failed, because the model could not be adjusted to correctly predict loudness perception for narrow-band signals. The model predictions in the lower loudness domain (between 'very soft' and 'soft') for the HI listeners were too low for all possible model parameter configurations when compared to the measured loudness in CLS.

As a conclusion, the SMR-algorithm requires to be adjusted using additional CLS measurements with broad-band signals. Further evaluations of the SMR algorithm using CLS of different everyday signals will show if the individually-adjusted SMR-algorithm restores the loudness perception to normal for narrow- and broad-band signals. Further improvements of recent loudness models to predict the individual loudness perception of a single HI is required for future research and fitting of model-based algorithms which individually restore loudness for a variety of stimuli.

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