A fitting Method for Headphones to compensate individual Hearing Impairments

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***Abstract*—Sensorineural hearing loss often results in altered loudness perception and a smaller perceivable dynamic range. To compensate sensorineural hearing loss it is necessary to fit an individual compressive gain. Fitting of hearing aids is usually done by professionals together with the hearing aid user. To give the users of TV-headsets the possibility to adjust not only the volume of the acoustic signal but also the dynamics, three different versions of interactive self-fitting systems were designed and evaluated. The user tests were performed with 11 subjects with mild to moderate hearing loss. The results were evaluated with regard to benefit in quailty, loudness perception and speech intelligibility.**

I. Introduction

Progress has been made in the last ten years by introducing full digital signal processing in hearing-aids, which allowed to develop and implement enhanced processing strategies including improved dynamic compression and noise reduction schemes. Consequently, available products are full of such features as multichannel amplifier designs with frequency dependent signal processing, algorithms for noise suppression that rely on automatic signal classification. In spite of these extensive technical advances, only 20% of the hearing impaired persons, who could profit by using a hearing aid, do not use one. Therefore one of the reasons is a stigmatisation of hardness of hearing [1]. To realise self-customisable headphones could help to conquer these prejudices. TV-headsets can help people with mild to moderate hearing loss to enjoy TV and music in satisfying loudness without disturbing their neighbours - or being disturbed - and offer a solution to bypass the disturbing free field influences. The users can choose their individual loudness level at the headset while the speakers of the TV device can be adjusted to a different level. It seems to be an optimal solution - but only in case of conductive hearing loss. Sensorineural hearing loss often causes an altered loudness perception and a reduced dynamic range. The headphones have to be adapted on the one hand to the individual hearing loss to assure audibility and on the other hand to the residual dynamic of the user to avoid too strong loudness. Therefore linear amplification of the input signals is not sufficient, compressive gain rules depending on frequencies are necessary. For hearing aids this fitting is usually done by professional audiologists. To give the users of TV-headsets the possibility to adjust not only the volume of the signal but also the dynamics, three different versions of interactive self-fitting systems were designed and evaluated. These interactive procedures for determining compressive gains enable the users to adapt the required settings by themselves in his home environment. For the first time, the recent “supportive audio signal processings (SASP)” strategies are developed to the specific application of supporting audio communication and speech intelligibility in radio and television.

This paper gives an overview of the state of the art fitting methods of hearing aids and on the specifics of broadcast material, before introducing the reader into the research and development of individual SASP strategies and their evaluations.

II. State of the Art

Sensorineural hearing impaired people often suffer from recruitment, means that the distance between hearing threshold and uncomfortable level is reduced and offers only a part of the dynamic of normal hearing persons. To take the best benefit of the remained dynamic range typically hearing aids amplify the input signal dependent on frequency and level by using compressors. Figure 1 shows an example of the loudness functions of a normal hearing and a hearing impaired person. Starting at 1 kHz the differences of the loudness functions become in evidence. The dynamic range above 2 kHz of the hearing impaired in this example is about 50 dB less than the dynamic range of the normal hearing person. To adapt the compressing gain rule in different frequency bands to individual hearing loss fitting methods are necessary. Usually this fitting is done with support of professional audiologists.

Technical progress in hearing aids needs also development of new fitting strategies and methods of configuring and fitting hearing instruments to individual user needs. A trend in fitting hearing aids is to involve the user's judgement in the fitting process at an early stage by using interactive procedures. As one of the first strategies “ScalAdapt” [3] is working

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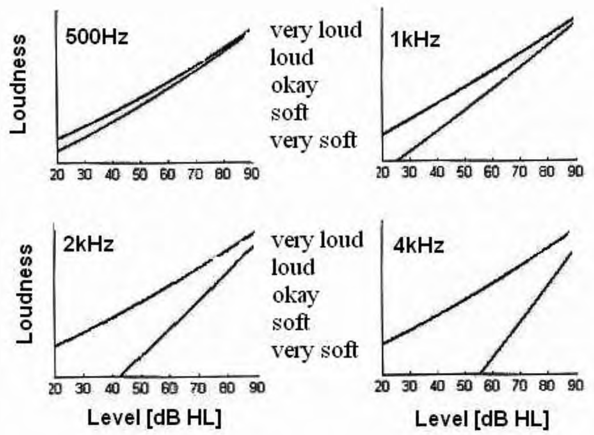

Fig. 1. Perceived loudness (from the bottom up: very soft, soft, OK, loud, very loud) of the frequencies 500 Hz, 1 kHz, 2 kHz and 4 kHz as functions of level. The bottom line shows the more rampant slope of loudness function caused by sensorineural hearing loss. Starting at 1 kHz the differences between the loudness functions become considerable. The dynamic range above 2 kHz of the hearing impaired in this example is 40 to 50 dB less than normal hearing capability [2].

with the loudness of narrow-band sounds. In 1998 Moore et al. developed the "CamAdapt"-procedure, which includes judgements on loudness and sound of speech [4]. Pastoors [5] combined these both approaches to "CascAdapt". In principle these methods are independent of hearing aid brands, but the more hearing aids get complex the fitting procedures get more complex. As well device-specific fittings are envolved, like the approach of Siemens Inc. with "Interactive Fitting" or the "Eartuner" of Microsound [6]. Both strategies includes additionally sound and loudness of the own speech and ambient noise. Interactive methods are applicable to achieve acceptable results and consumer acceptance in a relative short time. But it is still discussed controversially if it is useful to fit hearing aids only after hearing impaireds' fancy in the face of risk to fit the aid to the user's habit instead to his maximum benefit [7].

Also within the subject area broadcast there are different approaches for controlling dynamics at the receiver, taking into account diverse listening or reproduction conditions of different users. End of the 90's ideas of "Dynamic Range Control (DRC)" systems emerge based in many cases on transmitted control data [8], [9]. Other approaches like the so called Musicam-DRC system, proposed by IRT, are based on scale factor weighting in the MPEG-1-Layer2 source decoder [10]. But these systems are not able to take into account the diverse individual hearing losses of hearing impaired users.

III. Dynamics in Broadcast

System dynamic in audio signal processing is defined as the level difference between full scale or peak programme level and system inherent noise level. Digital broadcast offers about 20 dB more dynamic range than analog broadcast (fig. 2). Therefore, the provided dynamic range of digital radio can be reasonably used, e. g. to broadcast full dynamic range of excellent CD-recordings [11].

But the problems associated with programmes with a wide dynamic range are well known from experiences with watching DVD e.g. in a circle of friends, who boost the noise floor in the living room: the lower signals, especially the dialogues are hard to understand, but if people turn the volume up, they find the overall loudness much too loud. The transmission or recording dynamic range conflicts with the usable reproduction dynamic particularly in noisy environments. The available dynamic range decreases from about 40 dB in a quiet home environment to 25 dB in a noisy environment, whereas 15 dB are lost in the environmental floor [12].

The question is, how to achieve a satisfactory loudness balance as well between different television channels as betweeen the different formats and contents of the programms?

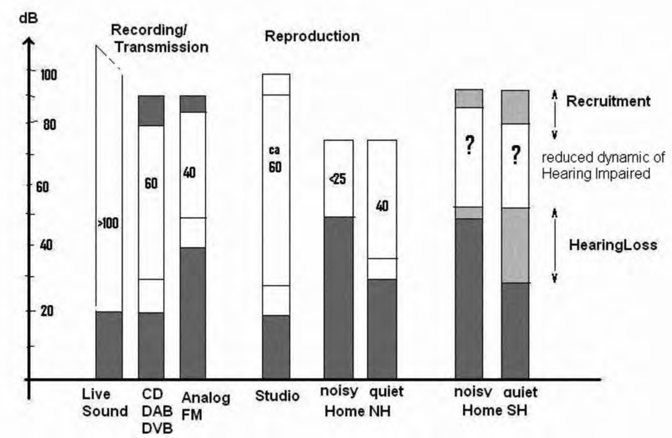

Fig. 2. Usable programm dynamics in audio systems: Dark grey coloured the environmantal noise floor, white the resulting dynamic range. For hearing impaired people not only noise floor, but also hearing loss and recruitment limits the dynamic range. Based on [12] [13].

The dynamic range of audio programme signals is in many cases larger than the dynamic range which is available at different home reproduction conditions or in a noisy environment. Fig. 2 shows relations between possible recording or transmission dynamics and the available consumer dynamic.

On the other hand the guidelines of German public service broadcast like ARD or the German-French station ARTE clearly define minimum and maximum levels and the dynamic ranges in between speech has to be arranged. Figure 3 shows the recording levels of ARTE programme [14]. In transmission systems there will always be the need to apply dynamic range compression dependent on the different types of programme material related to transmission guidelines.

Furthermore, the relation of loudness and peak levels of audio material is only marginal, e. g. at changes to commercials loudness is increasing rapidly, even though commercials follow the same production guidelines as all other broadcast material. For a fixed (root mean square-level (rms), compressed speech shows a higher long-term loudness than uncompressed speech. In particular multiband compression

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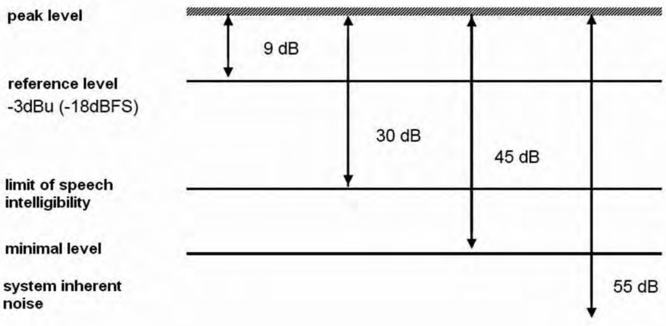

Fig. 3. Technical guidelines of the public broadcast station ARTE [14]. The reference level lies 9 dB under peak programme level and 18 dB under the clipping limit. The 9 dB above peak level are called headroom. Between limit of speech intelligibility and peak level there are about 30 dB dynamics for transmitting audible speech.

with short time constants makes the speech appearing louder than uncompressed speech of the same rms level [15]. Additionally compression allows to increase the rms level without increasing the peak level.

Considering the reduced dynamic range of sensorineural hearing impaired there has to be a frequency and level depending compression of the audio material in accordance with individual listeners' requirements. Particulary with a view on the steeper slope of the loudness functions of people with a sensorineural hearing loss, loudness leaps have to be avoided.

IV. Research and Development of individual SASP Strategies

The goal is the implementation and evaluation of individual supportive audio signal processing (SASP) strategies supporting audio communication and speech intelligibility in radio and television. The term "individual" means strategies which are dependent on the specific hearing loss of a single end user. It is obligatory that the developed fitting strategies do not require the knowledge or attendance of an audiologist. The underlying audio signal processing is based on a commercial product which is dedicated to the development of algorithms for hearing aids, the HOERTECH Master Hearing Aid (MHA) [17]. The graphical user interface (GUI) of the fitting-system is designed to be easily operated. The duration of the fitting should not take more than 10-20 minutes. This restricts the number of variables and the query of data, respectively. Three versions of fitting-systems for TV-headphones were realised. Of course users are not in the need of their audiogram data or any other technical or audiological background. The used fitting signals are taken out of broadcast news and movies. Low and high thresholds are determined by the use of the interactive GUI. These thresholds are applied within the system to provide appropriate compression parameters. Reference data of normal hearing people have been derived by measurements with ten normal hearing subjects.

A. Compressor Scheme

Loudness correction will be achieved by a multi-channel dynamic compression algorithm as used currently in hearing aids. In order to allow for a self-fitting of the algorithm, the number of adaptable parameters has to be limited. The used compression algorithm enables an independent processing in three compression bands (f_{center} = 250, 1000, 3000 Hz). The attack and release times describe the short periods of level metering (defined in standards like ANSI S3.22 and IEC 60118-2, [18]) and are fixed to 5 and 100 ms. An adaptive smoothing filter computes in each step the differences between current input to the filter and last output of the filter. Dependent on this differences the filter follows changes in level more or less immediately, with a smooth transition between the extremes. Every compression band is devided into three equaliser bands (f_{edge} = 75, 178, 354, 612, 866, 1225, 1730, 2450, 3460 and 4700 Hz). Fig. 4 shows the compressor and the center frequencies of compressor and equaliser bands.

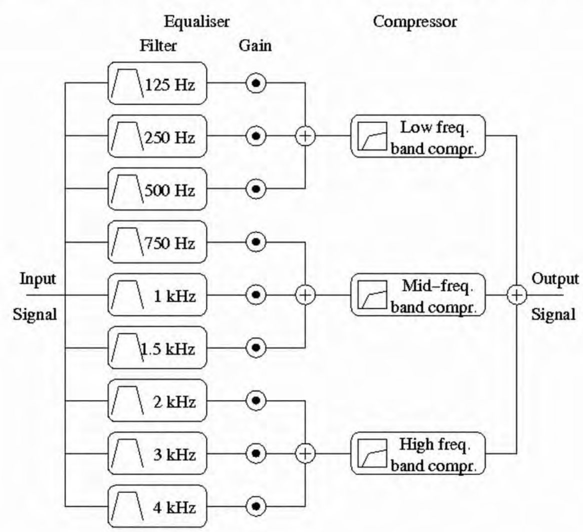

Fig. 4. The fitted compressor, with three equaliser bands in each of the three compression bands A (low frequencies), B (middle frequencies) and C (high frequencies).

The fitted parameters, compression ratio and linear gain, are computed individually on the basis of the metered Low- and High-leveled thresholds of the hearing impaired user related to reference data of normal hearing subjects.

B. Fitting Methods

The three different fitting approaches all have in common that the visual feedback of TV-signals supports the user to classify the respective hearing situation. The judgements on the loudness of the stimuli result from a realistic situation - watching TV. Usually the limits of a hearing loss are determined by audiologal sounds like sinus or band filtered noises. But TV signals consists mainly on speech, supported by ambient noises and athmospheric sounds, which are used to aid the imagination of the film reality.

To get an impression what is the difference between evaluating the Low- and High-levels with the help of speech

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(in combination with ambient noise) or with the help of only noise, two different metering concepts were followed (Vers.1 vs Vers. 2 & 3): The metering concept of Version 1 is to evaluate absolute thresholds with frequency dependent athmospheric sounds. In Version 2 and 3 the subjects have to compare the loudness of band filtered speech and to adapt the signals to the same loudness perception.

In principal the fittings are divided into two main blocks, one block of level metering task modules and after computing the compression parameters a block of fine tuning task modules. Every version is a composition of seven modules.

During the first block subjective Low- and High-Levels are determined within every compression band. These subjective levels are in combination with reference values the base of computing the compression parameters. To address possible asymmetrical hearing loss the measurements are conducted for both ears separately. This first block results in 12 thresholds: for both ear sides Low- and High-levels in three compression bands.

After this first block of level metering in a second block the user has the possibility to conduct some fine-tuning tasks to correct the balance and tone colour.

To get a further impression if it is possible to dispense with metering the High-level and to compute the compression ratios only on the basis of Low-levels, two different ways of computing the compression ratios were followed (Vers. 1 & 2 vs Vers. 3).

Version 1: Low- and High-Levels are measured based on frequency specific sounds of an animation film. Thereby the sounds are selected related to their naturally loudness: low sounds like wind or the noise of steps are used for metering Low-levels, naturally loud sounds for metering High-levels. The regulation of the levels is done by 6 dB steps.

Version 2: In a first step Low- and High-Levels are measured with news material filtered in the lowest compression band. To get the Low- and High-Levels in the upper compression bands the subjects have to adapt the levels of news material filtered in the second and third compression band to the loudness of the Low- and High-levels in the lowest compression band.

Version 3: In a third alternative the Low-Levels are determined as in Alternative 2, but the High-Levels are fixed in relation to technical maximum. The regulation of the 2nd and 3rd version are stepped by 3 dB.

C. Reference Data

Initially, a study with 10 normal hearing subjects (2 female, 8 male, age 21-56 years) has been performed to get reference values for computing the compression parameters. The normal hearing subjects did the fittings two times in test and retest. The reference data were middled (median) over all persons and both earsides. 5 shows the reference thresholds of the three versions.

Vers1	Low	High	Vers2	Low	High	Vers3	Low	High (fixed)
A	28	81	A	38	72	A	38	95
B	17	70	B	35	65	B	35	95
C	16	67	C	34	68	C	34	105

Fig. 5. Reference parameter sets: Low and High-levels of the three different approaches.

V. Usability Tests

Usability tests have been set up and performed in order to assess the applicability of the developed schemes. First the three different fitting procedures were performed with 11 hearing impaired subjects with mild to moderate hearing loss (5 female, 6 male, age 54-72 years). The compression parameters have been compared and evaluated in several measurements: The performance was assessed by questionnaires. A paired comparison test was used to show the preferences of the subjects and a "Channel-Hop-Test" was used to answer the question, whether a satisfactory loudness balance has been achieved in this way of compression. Finally the results of a speech intelligibility test are shortly discussed. All tests took place in an anechoic room in the HrTech laboratories in Oldenburg.

A. Fitted Compression Ratios and Linear Gains

Figures 6 to 8 show the different compression ratios as a result of the different approaches. The three alternative fitting methods induce clearly different parameter sets for the compressor.

Version 1 shows high linear gains increasing with frequency. The linear gains vary between 11 and 46 dB with compression bands dependent means (median) between 17 and 35 dB. The compression ratios (CR) also show a clear frequency relation: For higher freuqencies the compression ratio is increasing from 1.7 to 2.4 in the median. In the middle band (B-band) the median of CR is 2.2. Additionally the results for the CRs are very individual and the interquartiles range from 1.4 to 2.3 in the A-band, from 1.5 to 3 in the B-band and from 1.5 to 2.8 in the C-band (fig. 6).

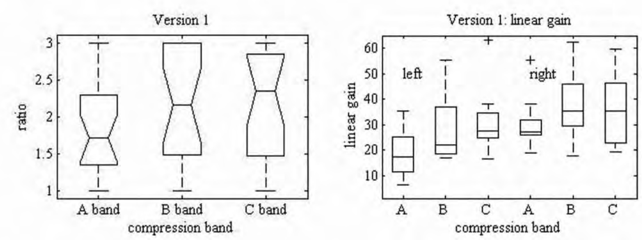

Fig. 6. Version 1: Low- and High-Levels are measured based on frequency specific sounds of an animation film. Regulation of loudness Level is done by 6 dB steps. The CR show a clear frequency relation: For higher freuencies the CR is increasing from 1.7 in the A-band to 2.4 in the C-band (median). Additionally the results are very individual and show a large scatter range.

Version 2 shows also wide spreading results for the CRs, but in the opposite order: For higher frequencies the medians of CR are decreasing from CR =2.1 for the A- and CR =2.0 for the B-band to 1.4 in the C-band. The interquartiles range

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here between 1.1 and 2.7 in the A-, from 1 to 2.9 in the B- and from 1 to 2.4 ind the C-band (fig. 7).

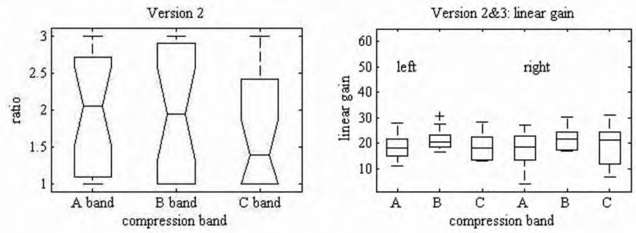

Fig. 7. Version 2: In a first step Low- and High-levels are measured with news material filtered in the lowest compression band. To get the Low- and High-Levels in the upper compression bands the subjects have to adapt the levels of news material filtered in the second and third compression band to the loudness of the Low- and High-levels in the lowest compression band. The so achieved CR decrease from low (CR = 2.1, median) and middle (CR = 2.0, median) to high frequencies (CR = 1.4, median).

Version 3 computes the lowest compression ratios with only a small range of spreading. The median values are 1.5 in the A- and B-band and 1.4 in the C-band. The interquartiles range only over plus/minus 0.1. The values are very close to the means in all frequency bands and the standard deviation is low (fig. 8).

The linear gains of Version 2 and 3 are the same and range from 12 to 25 dB with medians between 18 and 22 dB.

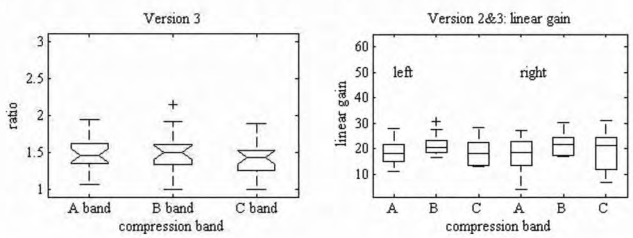

Fig. 8. Version 3: In a third alternative the Low-Levels are determined like in Version 2, but the High-levels are fixed in relation to technical maximum. The median CR values are 1.5 in the A- and B-band and 1.4 in the C-band and show only a narrow range of spreading.

The three parameter sets are exceeding different. Version 2 and 3 show explicit lower linear amplification than Version 1, unlike Version 1 computes higher compression ratios. These differences are caused on the one hand of the different fitting materials and and on the other hand of the different metering methods.

B. Questionnaire

During the fitting procedures the subjects had to compare and evaluate rather special, unusual sounds like frequency dependent sounds or band filtered speech. The subjects had to handle different task modules overall during the fitting. Seven of these tasks were selected for a questionnaire evaluation. One of the questions is (1) how comfortable or uncomfortable are the sounds which the subjects had to judge and (2) how applicable or not applicable are these sounds for executing the tasks. The subjects had to compare the loudness of different sounding stimuli. A further question was (3) how easy or difficult it was to do the several tasks. Additionally it was interesting (4) how intuitive the programme structure was and how clear the verbalization of the tasks was. For all questions an even scale from 1 (very good) to 6 (very bad) was used. Only the question (5) how optimal were the initial volume settings of the test signals were odd-scaled from 1 to 7 to offer the 4 for "optimal".

(1) Comfort of sounds (6 steps:"very comfortable" to "very uncomfortable"): The stimuli of Version 1 (band filtered movie sounds) were evaluated in median by 2.5 for the Low-stimuli, for the High-stimuli by 3. The band filtered speech of Version 2 and 3 was rated 3 on average for High- and Low-Level-Stimuli.
(2) Applicability of sounds (6 steps: "very applicable" to "very inapplicable"): The stimuli of Version 1 were evaluated by 2 for Low- and High-Stimuli. The band filtered speech of Version 2 and 3 got a median score of 2 for the task, in which the subjects had to adjust Low- and High-Levels of the lowest band. In the Loudness Compare-task they evaluated the applicability with 3.
(3) Difficulty of the tasks (6 steps: "very easy" to "very difficult"): Four of the seven requested task-modules were rated 2 (median), two modules 1 and only one task was rated 5 for "nearly very difficult".
(4) Usability of fittings (6 steps: "very intuitive" to "not intuitive at all"): For all fitting task-modules the usability and programme structure were rated 1 (median) for "very intuitive". Only two subjects gave a 4. In two modules one person had problems to understand the task description.
(5) Initial volume settings (7 steps: "too low" to "too loud"): Only one module was evaluated with 2 for "nearly too low", two task-moduls were evaluated by 3 for "bit too low" and four modules by 4 for "optimal" initial volumes.

C. Paired Comparison

The influence of compression on sound quality and the expected system accceptance of users has been checked by pair comparisons. Several TV-formats from different German broadcast stations were processed by the individual fitted compression parameter sets and their speech intelligibility and sound quality have been compared to quality of the original signal. The volume of the presented head phone signals has been set to a comfortable level with respect to the different processings.

The paired comparison evidences that the signal processings of all fitting-system are preferred by most of the subjects compared to the uncompressed signal as in terms of speech intelligibility as sound quality [16].

D. "Channel-Hop-Test"

Whether the settings of the self fitted compressor are appropriate and result in a satisfactory loudness perception of the TV signal had to be analysed. A "Channel-Hop-Test" was

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used to investigate that question. Therefore the Channel-Hop-Test simulates the type of loudness leaps of audio signals while watching television and changing stations or programmes. Fig. 9 shows a screen shot of the graphical user interface of the test.

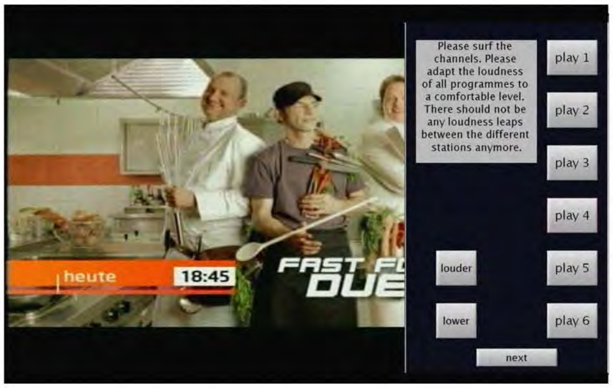

Fig. 9. User Interface of the "Channel-Hop" Test. Subjects can surf six programmes and adjust the levels to avoid loudness leaps between the diverse "stations".

The test includes six TV programmes which the subjects could toggle. The levels of the TV audio signals at the input of the compressor were prepared to vary in 5 dB steps between -10 dB and +15 dB relative to the subjects' most comfortable level (which was determined during the fittings). The order of the TV programmes and thus the different levels was randomized.

The subjects' task is to adjust the volume of all "channels" to a comfortable overall loudness in a way that there are no loudness differences between the six channels anymore. Therefore the subjects could change the volume settings for making louder or lower each single channel. The subjects were free to "zap" through the channels as often they liked. When all examples had reached the same subjective loudness the volume changes were stored. These volume changes were applied to the output signals of the compressor (fig. 10). The hypothesis is: If there is no compression the subjects will change the volume settings within the same range as the input levels differ, i.e. in a range of about 25 dB. For compressed signals the volume regulations carried out by the subjects will give the possibility to compare the subjective loudness differences with compression to the original, without compression. In this manner an effective broadband compression ratio can be computed.

To evaluate the users input pertaining to an estimated broadband compression ratio CR_{BB} the volume changes after compression are fitted by means of linear regression with a function $y(x) = x_0 + mx$ with the slope $m = y/x = -1/CR$ and x =5dB∗[-2 -1 0 1 2 3], whereas $y(x)$ are the volume changes in relation to the pre-compressor gain changes x_n for $N = 6$ different values. The fitted parameter $\hat{m}$ and $\hat{x}_0$ are computed with

$$\hat{m} = -1/CR = \frac{\sum_{n=1}^{N}(x_n y_n) - N\bar{x}\bar{y}}{\sum_{n=1}^{N}(x_n^2) - N\bar{x}^2}$$

and

$$\hat{x_0} = \bar{y} - \hat{m}\bar{x}$$

where $\bar{x}$ and $\bar{y}$ are the means of the vectors and $\hat{m}$ and $\hat{x_0}$ are the estimated values. Fig. 11 shows the so received straight line functions for the diverse processings of subject 8 as an example.

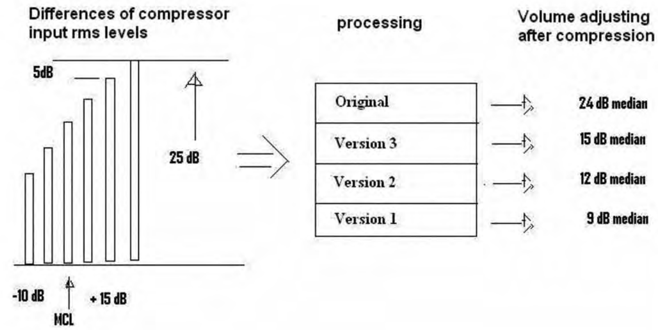

Fig. 10. Left: The differences of compression input signals in rms levels. Right: the mean values (median) of the subjective loudness differences, evaluated on adaptations of the output gain (volume) by the subjects related to the used compression data sets.

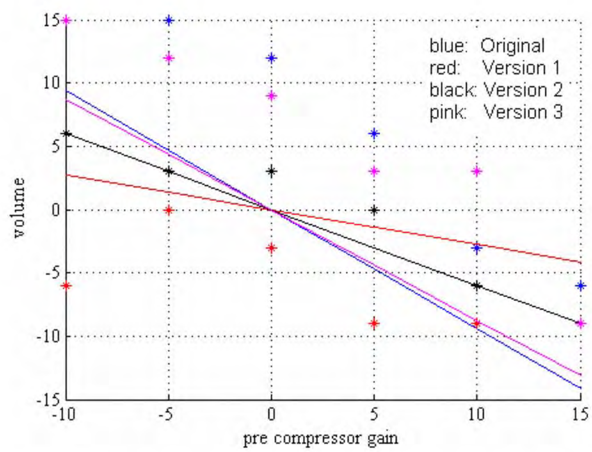

Fig. 11. Subjective volume adjustments after compression fitted by linear functions of level changes of the compressor input (blue: identity, red: Version1, black: Version2, pink: Version3). The straight line functions are shifted to reference point $R(0, f(0))$. Asterisks indicate measured values. In case of no compression (original version) subjective volume adjustments fully compensate the input level changes, thus resulting in a linear fit with a slope of about -1 -1 with $\hat{m} = -1/CR$

To assess the quality of the straight line fittings the coefficients of determination

$$B_{xy} = 1 - \frac{\sum_{n=1}^{N}(y_n - \hat{y}_n)^2}{\sum_{n=1}^{N}(y_n - \bar{y})^2}$$

was evaluated additionally. Fig. 12 shows the results: The coefficient of determination is strongly related to the way

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of fitting as well as to each single subject. Left in fig. 12 the coefficients of the four different versions of processing are displayed as boxplots. The loudness judgements on the non-compressed original material have the best coefficient (median 0.91). Thus, it seems to be reliable to be used as reference. Furthermore the median range of subjective volume adjustments after compression is 24 dB, i.e. only 1 dB less than the range of rms level differences of the compressor input signals.

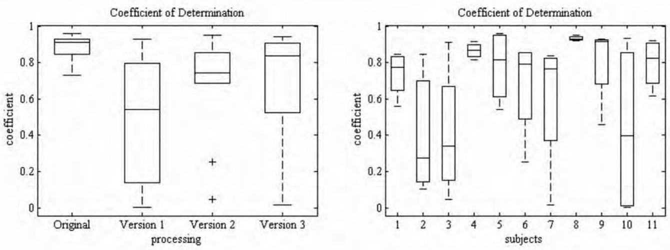

Fig. 12. Coefficients of Determination B_{xy}. Left: coefficients over all subjects according to the different processings (original and fitting Versions 1 to 3). Right: coefficients over all processings according to each single subject.

Version 1 shows not only the strongest compression (cmp. fig. 10) but also the highest uncertainty of the subjects' loudness (median = 0.54). The estimated broadband compression ratio of Version 1 is in five of eleven cases larger than 4, even though the maximal CR in the single compression bands is limited to $CR = 3$. Fig. 13 to 15 show the compression ratios per frequency band (A,B,C) and the estimated effective broadband (BB) compression ratio for each subject. Version 2 and 3 have higher coefficients of determination (Version 2 = 0.74, Version 3 = 0.84) (fig. 12, left), but also show less compression.

The volume adjusting after compression happens for Version 2 in a range of 12 dB (median), for Version 3 in a range of 15 dB (median). For Version 1 there is only a range of 9 dB (cmp. fig. 10). It seems to be a relation between strength of compression and decidedness in loudness judgements of the subjects. Having a look on the coefficients of determination plotted as a function of each single subject (fig. 12, right) the differences between the results are explicit. Subjects 2, 3 and 10 had serious problems in their loudness judgements, nearly independent of the used compression data set. Even though these subjects achieved high values for the coefficients in judging the loudness of the uncompressed signal (between 0.84 and 0.93), the coefficients of determination for the compressed material lies under 0.5. The assumption then is legitimate, that the compression parameter of these three subjects are bad fitted.

A view on the estimated broadband ratios in figure 13 to 15 affirms this assumption: Subjects 2, 3 and 10, they all hold very high estimated broadband ratios for the fitting Version 1 (fig. 13), subject 3 also holds an estimated ratio higher than 6 for fitting Version 2 (fig. 14) and for Version 3 subject 2 and 10 have estimated broadband ratios higher than 6, too (fig. 15). Furthermore Version 1 shows one more case of unusual high

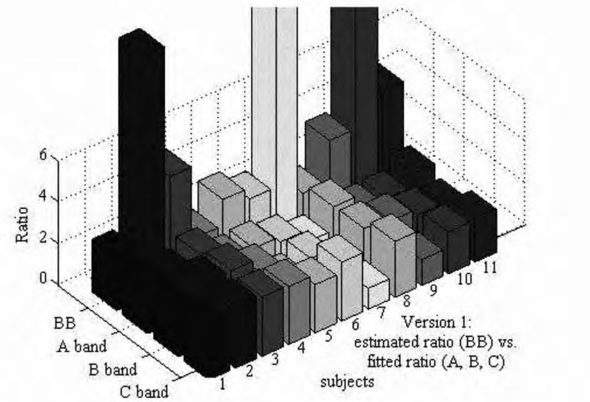

Fig. 13. Five of the eleven estimated broadband (BB) compression ratios are higher than 3, the maximal allowed CR in the fitting.

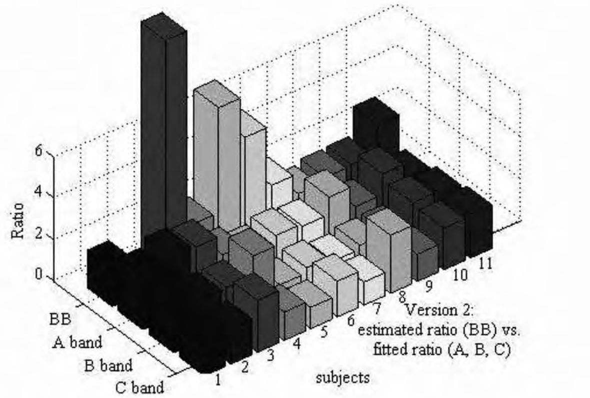

Fig. 14. Three of the eleven estimated broadband (BB) compression ratios are higher than the maximal allowed CR in the fitting.

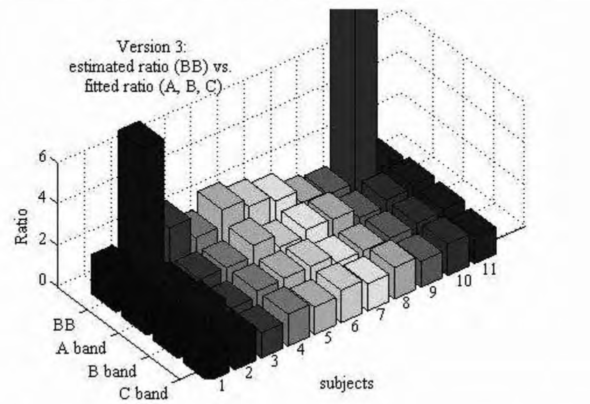

Fig. 15. Two of the eleven estimated broadband (BB) compression ratios are higher than the maximal allowed CR in the fitting.

estimated broadband compression ratio for subject 7, Version 2 relative high broadband ratios for the subjects 5 and 6.

E. Speech Intelligibility

To account the benefit in speech intelligibility the Oldenburg Sentencetest [19] was performed with the eleven subjects [16]. As expected there was no benefit in speech recognition threshold (SRT, signal-to-noise ratio for 50% speech intelligibility)

Digital Object Identifier: 10.4108/ICST.PERVASIVEHEALTH2009.5991
http://dx.doi.org/10.4108/ICST.PERVASIVEHEALTH2009.5991

for the subjects 2, 3 and 10.

Regarding the remainder of the subjects the obtained results are very positive. Six of the remaining eight subjects with coefficients of determination higher than 0.7 improved their speech recognition threshold with the help of the self-fitted compression related to free field transmission in anechoic room, whereas subject 7 does not profit by any fitting version at all. Subjects 5 and 6 - with the relative high broadband ratios in Version 2 - also achieve benefit in SRT for all fitting Versions. Estimated broadband ratios higher than 4 affect not necessarily the speech reception adversely.

Related to the uncompressed headphone signal the benefit of speech recognition depends on which version of fitting was used. For subject 11 there is no value measured, but for the other seven persons: By fitting Version 1 and 2, four of seven subjects profited, two subjects changed for the worse and one subject stayed equal in speech recognition threshold. From Version 3 all seven remaining subjects profited in speech recognition.

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VI. Conclusion

The three parameter sets are exceeding different. Version 2 and 3 show explicit lower linear amplification than Version 1, unlike Version 1 computes higher compression. The reasons for these different data sets have to lie in the applied metering methods as well as in the used audio material. Speech has the higher dynamic and an assumption for the higher scaled Low-level while using speech is that subjects try to understand the content, which could lead to a higher Low-level than even to recognise a sound. The different ways to compute the parameter sets results in a different behaviour related to the individual dynamic values of the hearing impaired. Computing the compression ratio only dependent on the Low-levels of the hearing impaired (Version 3) retains the influences of the individual thresholds small compared to the influence in Version 1 and 2.

The answers in the questionnaire causes the assumption that the used task modules are applicable and usable, but they show also a tendency to a certain grade of annoyance. Therefore the number of tasks have to be limited to a small set of the real important interactions. The paired comparison test and the OLSA - sentence test evaluates the fitting results, not the procedure. The paired comparison test shows a good acceptance of the processed material and an improvement of the subjective speech intelligibility. But also the objective intelligibility, the speech recognition threshold (L50) of many subjects get a better value than without processing.

Finally the "Channel-Hop-Test": Its results have very promise and the relation between the coefficients of determination and the quality of the fitted parameter set has to be proven in further evaluations. Implemented in an overall system a modified version of the Channel-Hop-Test could be used as an intelligent indicator of fitting quality and thereby of satisfaction of the users.

But there are still many improvements necessary to come up with an usable system. The principle goal of that approach is to enable users to fit their headphones with all broadcast material, only dependent of the actual programme. Version 2 and 3 converge more to that goal, cause the fitting material is just band filtered uncutted newscast, with clear speech, but also including reports or commentaries. In addition the combination of a user-side approach with a metadata approach to improve the signal noise ratio not only of the environment, but also of the broadcast audio mix should be brought forward and general SASP strategies have to be envolved.

Acknowledgment

The authors would like to thank EU project Hearing at Home FP6-2005-IST-6 for the financial support to carry out these studies. The authors would like to thank also all the consortium partners.

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Digital Object Identifier: 10.4108/ICST.PERVASIVEHEALTH2009.5991
http://dx.doi.org/10.4108/ICST.PERVASIVEHEALTH2009.5991