Introduction to Sound Quality

BA 7609-13, 1

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Abstract

This lecture is a basic introduction to Sound Quality.

"Sound Taylor" is Brüel & Kjær's concept name for Sound Quality solutions.

It will in short cover all the steps involved in a Sound Quality optimisation process with emphasis on all the major points and considerations leaving the details to other presentations.

LECTURE NOTE

English BA 7609-13

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Contents

- What is Sound Quality?
- Why improve the Quality of Sound?
- Work with Sound Quality
- Sound Recording
- Sound Editing
- Subjective Tests
- Objective Tests
- Trouble-shooting
- Product Engineering

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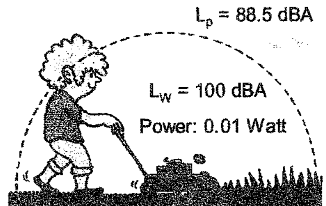
This list of contents reflects a guided tour through steps needed to make one Sound Quality optimisation in the same order as will be used in real life.

Noise versus Sound

Noise

Measured:
- to avoid hearing damage
- *to fulfil regulations*

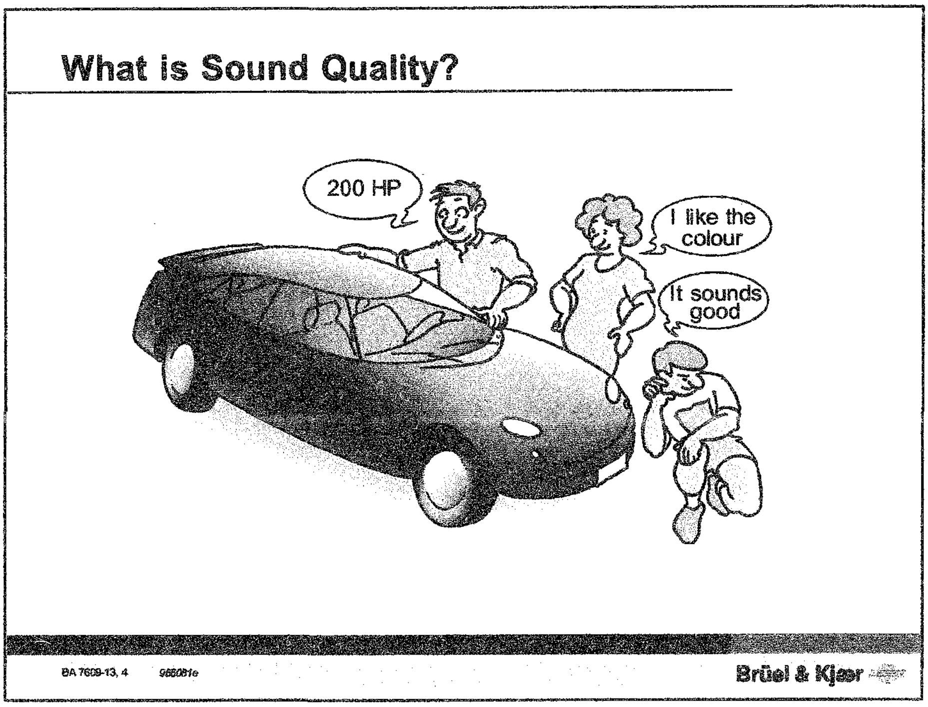
Measured as:
- dBA
- Sound Power



Sound

Perceived as:
- pleasant sound
- unpleasant sound
- informative sound

Measured as:
- "Sound Quality"



The sound of a product is now a product parameter that needs the same attention as its physical design, horsepower, color, weight, price etc..

It all started in the automotive industry more than 10 years ago and it is still in that area the most advanced Sound Quality developments take place. One reason is the very heavy competition between car manufacturers and the fact that most cars are of high quality and have the same performance in relation to what they are supposed to do. Sound Quality as a relatively new product parameter has then become an important item to compete on. If your car has better Sound Quality than the competition you are closer to win the sale.

In recent years the focus on Sound Quality has spread to almost all other industries producing products that makes noise. The household appliance industries are good examples.

Sound Quality as a product parameter is most developed in USA, Europe and Japan. In other countries is expected to grow rapidly as more and more products become not sellable unless their Sound Quality parameters has been attended to.

As product sound is directly communicating with the users senses - the ears - the knowledge of how we perceive sound has got increased focus. This discipline is called Pscychoacoustics and is important in the education of design and development engineers.

Why improve the Quality of Sound

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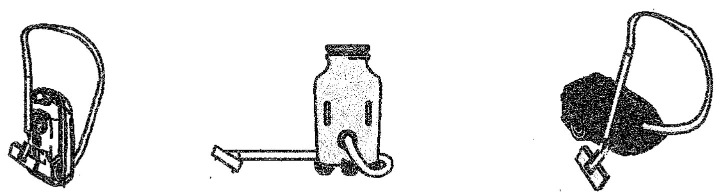

- Good Sound Quality signals proper operation of the product
- Sound Quality is a parameter that sells the product
- A-weighted noise levels and sound power are fine to characterise noise, but useless for sound
- Sound Quality is not static, but changes over time like fashion

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The noise from a product is a part of the communication between the product and its user. Therefore it has to be changed into sound that is pleasing to the user and give him all the information of the function and life time of the product he needs - no more no less.

The pleasing aspect of product sound is perceived subjectively and depend on the individual user. This situation leaves the designer of the product in a very challenging position.

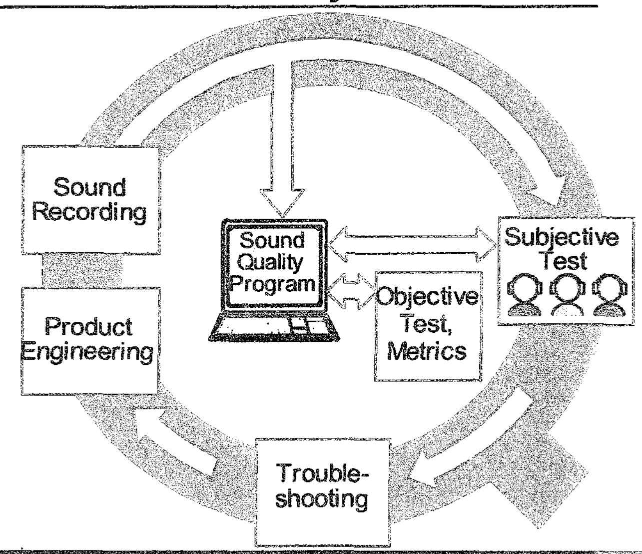
He has to optimise the product sound to the target customer group that has a taste that is not uniform and will change over time as fashion. Pleasing sounds get worn and need replacement by new exciting sounds.

He has to skip old design tools and learn new ones. For example the widely used A-weighting is fine for noise but useless for sound. Three different vacuum cleaners may have the same A-weighted noise level but can have very different sounds.

No noise may be a target for a noise control, but in relation to Sound Quality no sound is unacceptable.

Products should always signal proper operation to the user as well as a warning signal when the electric drill is overloaded or the car is hitting rough road surface an the driver should reduce speed.

Work with Sound Quality



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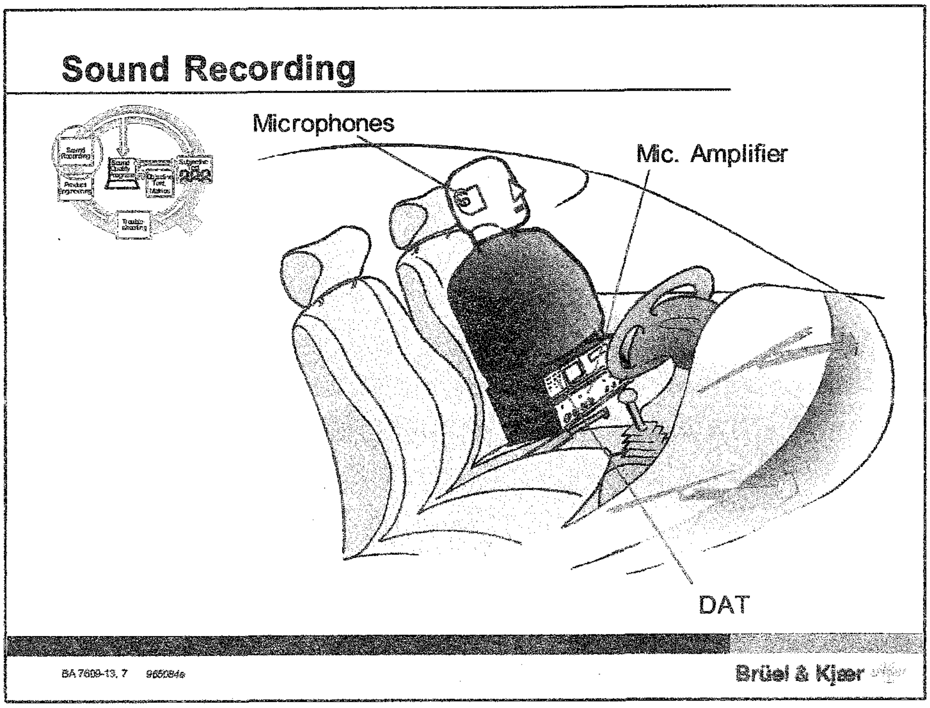
Working with Sound Quality is an iterative process. Often you start with prototypes of a product which has to be optimise in Sound Quality. You make recordings - preferably using a Head and Torso - of the sound from your prototypes and you may also have competitor products included in the test. Then you get the first evaluation from a listening test with a jury representing the final users of the product. If your prototype wins the listening test and is perfect you have finished the job.

If your prototype fails you can direct the sounds to the Sound Quality program for detailed analysis. In that you may find some spectral components which you expect responsible for the poor sound. With the edit function in the program you can then simulate a removal of the unwanted components. If a new listening test approve the modification the next step is to do some trouble shooting to identify where the unwanted components come from. Then some product engineering is needed to modify the prototype. Then a new sound recording and listening test is needed.

If your prototype still fails you must go to the analysis again and try other edits to modify the sound. In order to qualify your progress a number of objective tests - Metrics - are available. They give a single number to characterise specific properties of the sound for example how rough the sound is. If you know that e.g.. an increase in the value corresponds to improved Sound Quality you can use this metric to optimise the simulations of product changes and save time consuming listening tests.



Recording sound samples are typically done off line without the Sound Quality program, for example in a car. The calibrated sound signals are stored on a DAT tape for later analysis and listening tests.

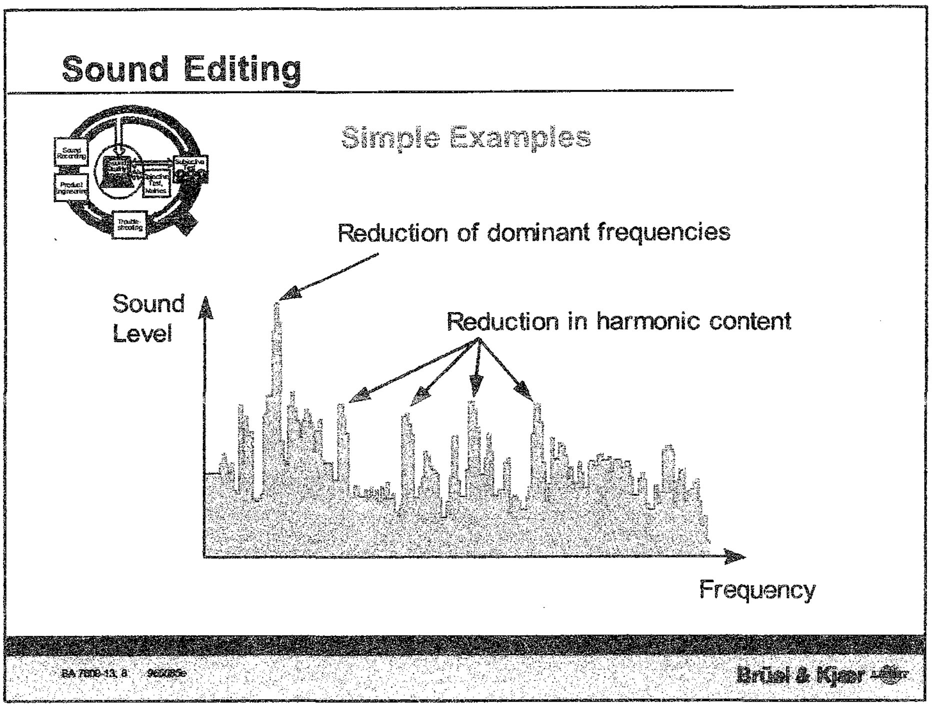
Only sound recordings using a Head and Torso gives signals that have all the directional clues from the sound field and they can be reproduced faithfully using headphones.

The sound recording system must be portable (light weight) and battery driven.

It is very important to have all recordings properly calibrated - a DAT tape with un-calibrated data is useless in later analysis or listening tests. Normally, calibration is only done acoustically using a Sound Level Calibrator. In addition Brüel & Kjær has developed the CIC - Charge Injection Calibration technique using electrical signals only. By pressing a button you can record test signals that can verify the calibration status of the complete system.

The CIC check facility is a reliable, convenient and time saving technique.

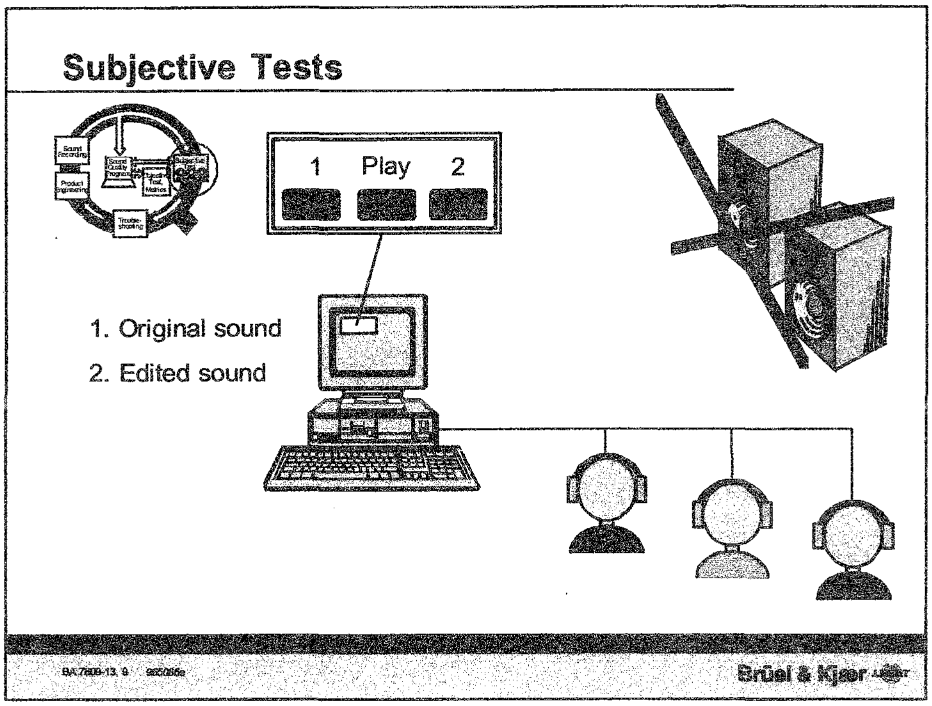
In situations where directional characteristics are unimportant or if cost and complexity of the recording package is a consideration, a single microphone can be used. However, a later listening test can never be completely faithful and is based on assumptions that may not be accurate. The play back is only in mono.



Sound editing takes place both in the time and frequency domain. The purpose is to change a given signal from a product using many different tools. These range from reducing or adding single frequencies or harmonic structures over shifting frequencies to another position, demodulate signal, limit their time responses, ...to mix it with another signal.

The trained user of a Sound Quality editing system will soon learn what type of edits that are useful to improve the Sound Quality of a given sound. But his job is not finished before the modified signal is preferred over the original in the following subjective listening test.

A number of display formats, including waterfall and contour plots as well as zoom and choice of resolution in the frequency displays, helps the user to select his edits.



he final evaluation of the Sound Quality of a product is always done by the person who consider to buy the product.

Subjective listening tests therefore plays a very important role in the design of the product. They must be conducted throughout the entire design process with the greatest care and accuracy.

A listening test can typically involve 16 persons. They have to be carefully selected for their demographic position, economy status and the probability that they are potential customers to the product under test.

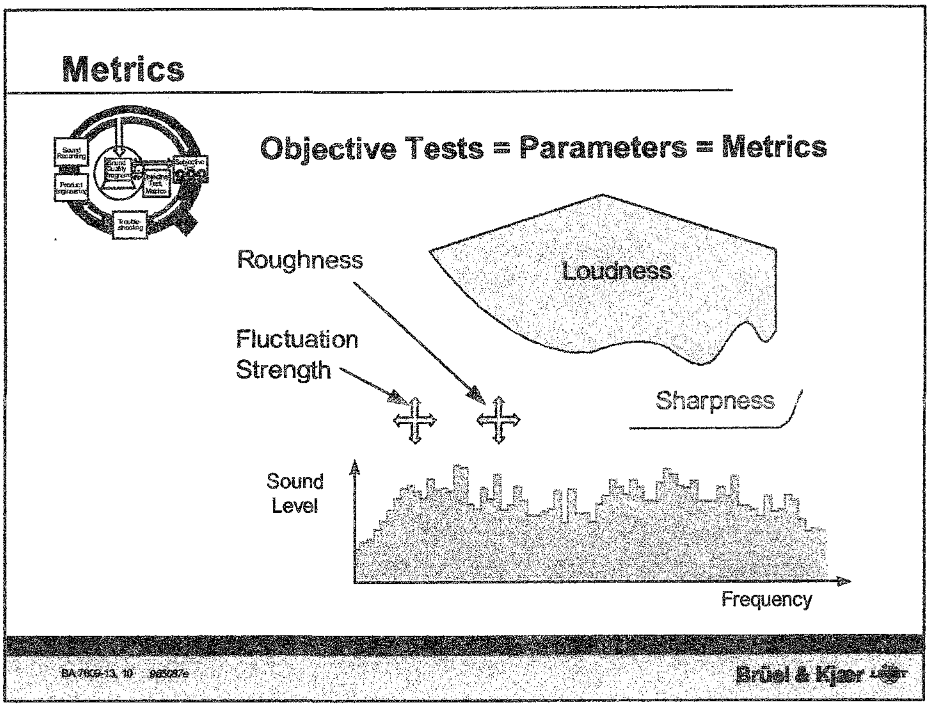
In almost all cases the listening is performed using headphones. That ensures that all the listeners have the same sound exposure and that they are reasonable protected against unwanted disturbing sounds.

Furthermore, sound reproduction using headphones is the only way to ensure faithful reproduction of sounds recorded with a Head and Torso.

There are two test methods commonly used:

Paired Comparison technique lets the listener be presented with a sequence of pairs of sounds A and B. For each pair the listener has to decide which is the preferred one.

Semantic differential technique lets the listener evaluate properties of the sound. For example the signal property smooth or harsh is judged on a 7-point scale, where the middle value 4 means neither smooth or harsh.



The most important Parameters or Metrics used in Sound Quality are those based on Zwicker Loudness calculations. They are reflecting most of the pscychoacoustic properties of the human perception of sound. They have the advantage that they put a single figure on characteristic properties of the sound. Three of the important ones are mentioned here:

•Fluctuation Strength is a measure low frequency - around 4 Hz - frequency and amplitude modulation in the time sample and is based on a non-stationary loudness calculation.

•Roughness is similar to Fluctuation Strength apart from measuring the modulation around 70 Hz.

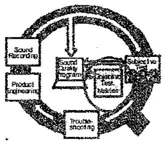
•Sharpness is a measure of the amount of high frequency content in the signals frequency spectrum. It can be calculated based on both a stationary and a non-stationary loudness calculation.

Only Zwicker Loudness calculations for stationary signals are standardised. And most real life signals are non-stationary.

Although there exists descriptions and formulas for Roughness, Fluctuation strength and Sharpness they are not very precise. That means that implementations of these Metrics from different manufacturers of Sound Quality analysis equipment will vary and can give different results.

Efforts regarding standardisation are going on both in ANSI and DIN.

Other Metrics



- Pleasantness
- Annoyance
- Tone-to-Noise Ratio
- Prominence Ratio
- Tonality, Pitch Strength
- Speech Interference Levels (SIL)
- Intelligibility, Articulation Index (AI)
- Speech Transmission Index (STI)
- Kurtosis
- etc.

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Here are some examples of other objective measurements - Metrics -often used in Sound Quality evaluations.

Pleasantness and Annoyance are combination Metrics based on a weighted sum of Zwicker Loudness, Fluctuation Strength, Roughness and Sharpness.

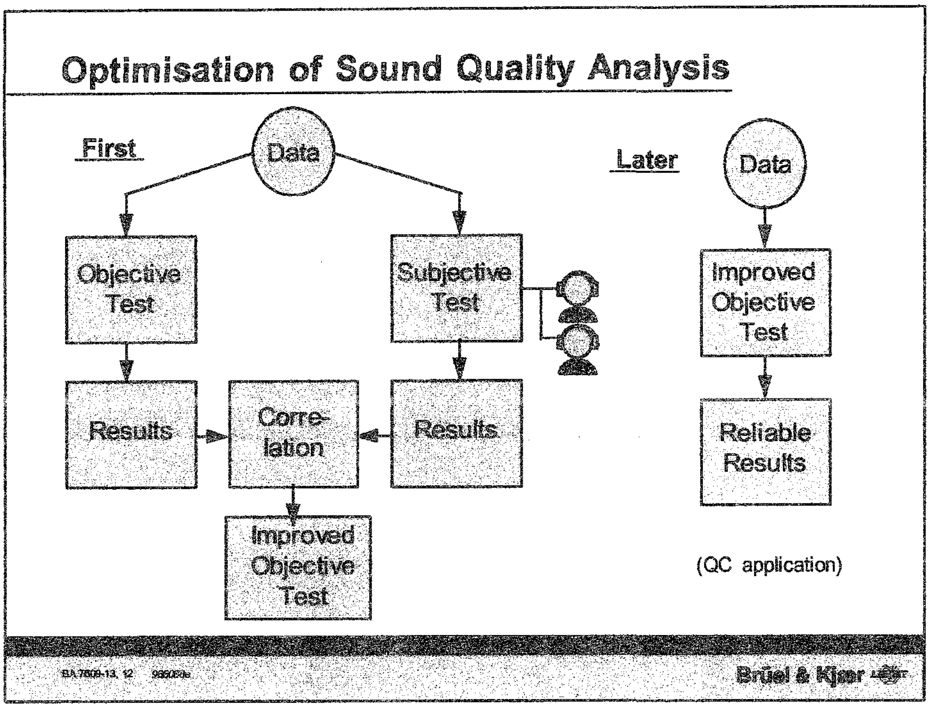
Tone to Noise Ratio is a measure describing the amount of pure tones in the signal

Prominence ratio is a description of the amount of noise in a critical band in relation to the noise i the adjacent bands.

Tonality or Pitch is a measure of how strong the sensation of "frequency" is in a complex signal.

Speech Interference Level, Articulation Index and Speech Transmission index are all measures related to the quality of a speech transmission channel. They also find uses in some Sound Quality applications.

Kurtosis is a measure of impulsiveness of the time signal. Basically it sums up all time samples level difference from the signals mean value and raised to the power of 4 and then normalised. The method exaggerates the impulses in the sound and a high kurtosis value normally reflects poor Sound Quality.



In optimising product Sound Quality you must never forget that the human being is the final judge on how well you succeed. That is, it has to pass the subjective listening tests flawlessly. Listening tests are both time consuming, costly and off line in relation to the Sound Quality editing and simulation process. Therefore the objective metrics based on pscychoacoustic research are very attractive as a complement to the subjective tests. They are cheap, fast and on line, but only a real substitute to listening tests if they can give matching results.

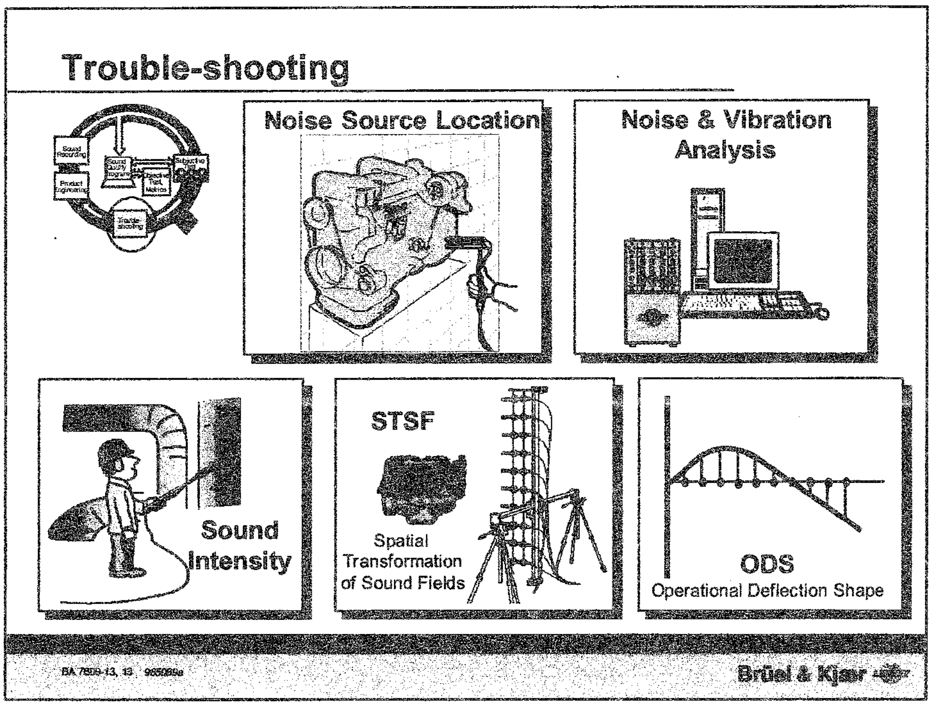
The big challenge is to design a set of metrics e.g.. as a combination of several metrics with individual weighting, that as a single number can give precise and reliable correlation to the subjective tests and preferences

Most manufacturers working with Sound Quality deal seriously with this problem. They regard the results as company secrets, so they seldom publish their findings. On the other hand the results are often so product specific that they hardly are of any direct use for other manufacturers.

It is believed that a metric never will be good enough to replace the subjective test completely they will remain complementary partners.

Another use of metrics is:

As sub suppliers now also have to meet Sound Quality requirements, they are obliged to perform QC on there products. Naturally, they can not rely on subjective tests for that purpose, but have to develop a good correlating metric.

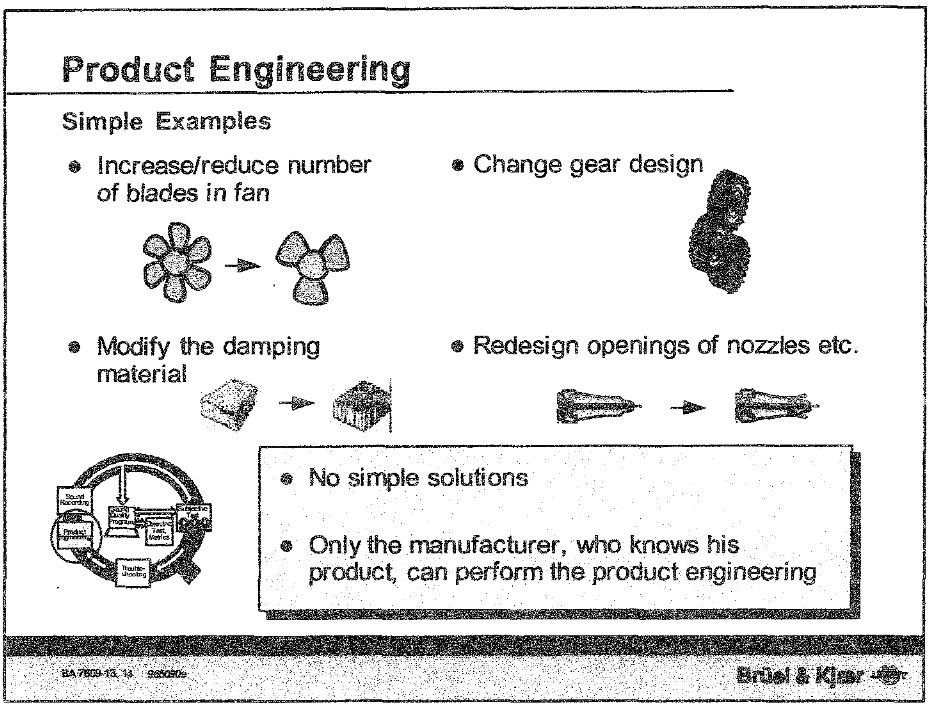


When the correct sound has been identified and verified in the listening test, you have to redesign the product.

In the Sound Quality editing and simulation that gave the right sound there is a clear documentation of the signal changes needed.

Now it is time to identify which part of the product is responsible for an eventual excessive frequency component that has to be removed. Or you can find sounds which comes from a wrong and unwanted direction a wrong direction.

To do that all Brüel & Kjær's acoustic and vibration analysis tools come in handy. From basic Noise & Vibration Analysis, Locating sources with intensity mapping, identify the Operational Deflection Shapes to complete modelling of the sound field using STSF - Spatial Transformation of Sound Fields.



Knowing what the product should sound like and where the unwanted sounds originate from, leaves a challenging part to be solved - re-engineering of the product.

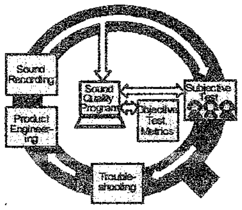
This can be a relative easy process if the product is simple and with few mechanical parts.

Or it can bee very complex if the structure is large with many interacting parts. In some cases the necessary redesign can be so demanding that it is close to starting the design from scratch in order also to satisfy the Sound Quality requirements.

Therefore when Sound Quality is an important product parameter it is obvious to include it in the design specifications for new product developments and include it in the engineering design process. Doing this you can save later troublesome adjustments and cost.

The required sound specification can be synthesised and approved by a listening test using a Sound Quality analysis system so it is ready when the product design starts.

Conclusion



- Sound Quality is an important *product parameter* and also subject to Quality Control
- Sound Quality optimisation requires *precise sound recording* and reliable subjective evaluation
- Efficient tools available to edit sound samples to *simulate* effect of improvements
- Customised objective measurements, Metrics, help to *cut time* in the sound design process
- *Complete range of tools* for trouble-shooting and engineering necessary to implement the desired sound quality

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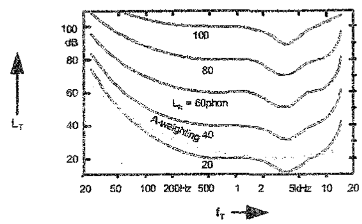
This short lecture has hopefully given you some idea of all the steps involved in a Sound Quality optimisation process.

All the presentations here has been held in qualitative terms leaving the detailed theory and mathematical terminology to other presentations.

Psychoacoustics

– a qualitative description –



Abstract

In this lecture we will highlight the most important aspects of the human hearing system that affect the perception of sound and its quality.

As will be seen and demonstrated, human sound perception is a complex matter with a lot of non-linear elements. These need to be understood and taken care of in order to make Sound Quality evaluations that match the subjective experience of the user of the sound.

This lecture will only give *qualitative* explanations of the psychoacoustic parameters and leave a proper *quantitative* description to another lecture.

Useful reference:

E. Zwicker & H. Fastel: Psychoacoustics, Facts and Models, Springer Verlag Berlin 1990

LECTURE NOTE

English BA 7615-13

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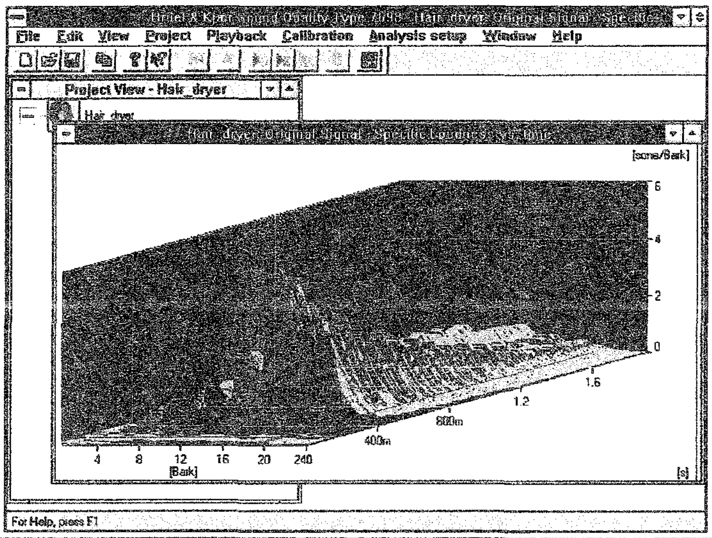
Loudness Spectra an Example



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This 3-D display of Loudness is an practical example of a measurement that takes most of the psychoacoustic parameters into account as they are explained in this lecture.

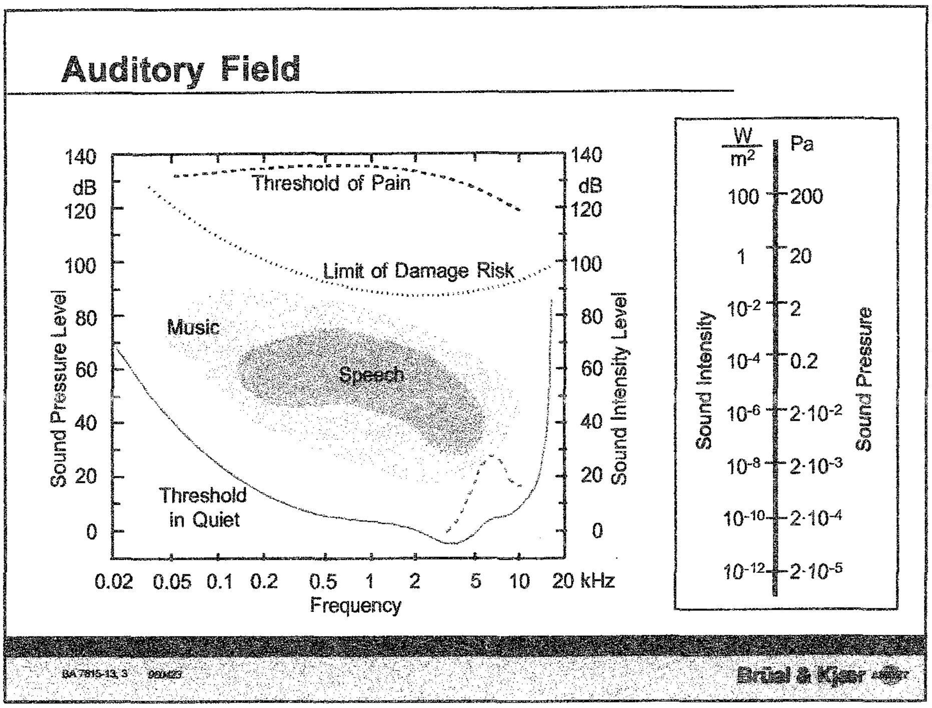
It is noted that the frequency scale is replaced by a Bark scale. This much better reflects the human ear's ability to discriminate frequencies and the perception of sound in bands one Bark wide.

The level axis normally seen in dB is replaced by a sone scale. This has the advantage that every time the perceived sound level is doubled the loudness value in sone is also doubled.

The 3-D display illustrates that sound signals are rarely stationary but vary over time. When the loudness spectra are sampled with short intervals, it is possible to see level changes - modulation - at frequencies up to around 70 Hz. If a high level of these are registered, a large level of Roughness can be measured. If the level modulation is at lower frequencies around 4 Hz, a high level of Fluctuation Strength can be expected.

Each loudness analysis includes compensation for the ear's frequency masking.

Also temporal masking is taken care of in the analysis when two closely spaced loudness spectra have large level changes.



This display of the auditory field illustrates the limits of the human auditory system.

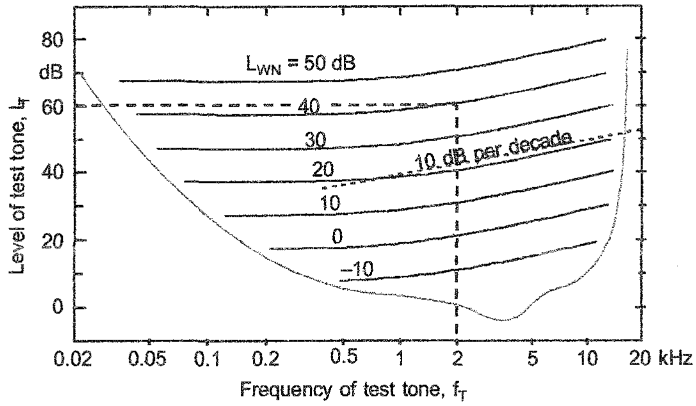
The solid line denotes, as a lower limit, the threshold in quiet for a pure tone to be just audible.

The upper dashed line represents the threshold of pain. However if the Limit of Damage Risk is exceeded for a longer time, permanent hearing loss may occur. This could lead to an increase in the threshold of hearing as illustrated by the dashed curve in the lower right-hand corner.

Normal speech and music have levels in the shaded areas, while higher levels require electronic amplification.

Human hearing is extremely sensitive. An acoustic power intensity of only 1 mW per square metre may already exceed the limit of damage risk.

Masking Patterns for White Noise



Masking patterns for white noise at different spectral density levels

Masking represents one of the most basic effects in psychoacoustics. This is normally determined as the audibility of pure tones in the presence of masking sounds.

This figure gives an example with white noise as a masker. The level of the just audible sound is given as a function of frequency.

The lowest curve represents the threshold in quiet of the audibility of test tones without masker. The other curves represent masking patterns of white noise at different spectral density levels.

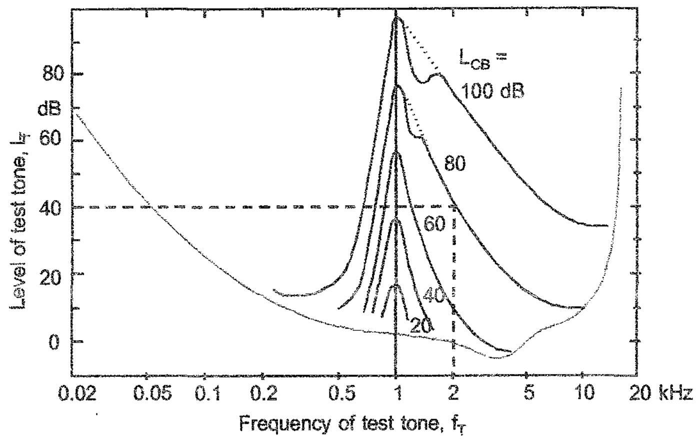
If, for example, the level of a test tone L_T at 2 kHz is 60 dB or below, it will be masked if the white noise has a level L_{WN} of 40 dB.

With increasing masking level, the masking patterns of white noise are shifted in parallel towards higher test tone level.

Up to a test tone frequency of about 500 Hz, the masking patterns are horizontal, at higher frequencies an increase with a slope of about 10 dB per decade shows up.

Since white noise has a spectral density level independent of frequency, the shape of the masking pattern is somewhat unexpected. However, it can be explained on the basis of critical bands described later in this lecture.

Masking Patterns of Narrow-band Noise



Masking patterns of narrow-band noise centred at 1 kHz with a bandwidth of 160 Hz at different levels L_{CB}

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This figure shows the masking patterns of a narrow-band noise centred at 1 kHz with a bandwidth of 160 Hz.

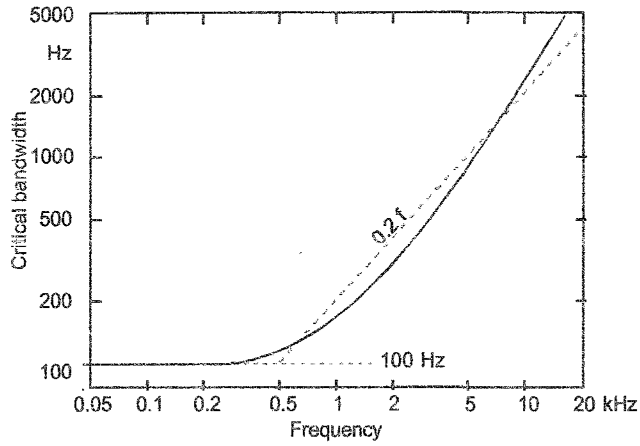
The lowest curve represents the threshold in quiet. The other curves illustrate masking patterns for different levels of the narrow-band noise.

For example, a test tone f_T at 2 kHz with a level L_T of 40 dB and below is masked if the noise level L_{CB} is above 80 dB.

At low levels of the narrow band masker, the masking pattern has a symmetrical shape. However, when increasing the masker level above 40 dB, the lower level is shifted in parallel, whereas the upper slope gets flatter and flatter.

This effect is called the "non-linear upward spread of masking".

Critical Bandwidth as a Function of Frequency



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A basic feature of psychoacoustics is the concept of critical bands.

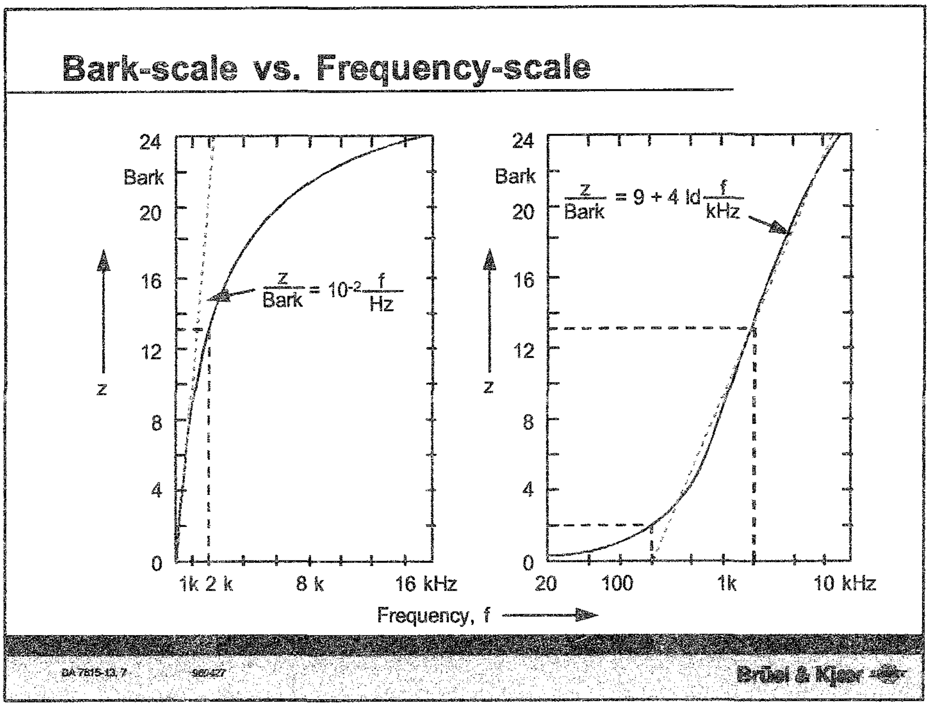
It is based on the assumption that the sound is analysed in the human hearing system by a bank of filters.

In the figure the bandwidth of these filters (critical bandwidth) is shown as a function of frequency - the solid line.

The dashed lines illustrate useful approximations:

- At frequencies up to 500 Hz the bandwidth is constant at 100 Hz
- At higher frequencies the bandwidth is relative - about 20%

That means that at frequencies above 500 Hz, the critical bands can be compared with 1/3 octave-band filters, which have a relative bandwidth of 23%.



In this figure we have compared the Bark scale with the frequency scale.

In the left panel, the frequency scale is linear, in the right panel it is logarithmic.

The solid curves describe the relation between the Bark scale and the frequency scale. The dashed curves shows the useful approximations. These are valid up to 500 Hz left panel, and above 500 Hz right panel.

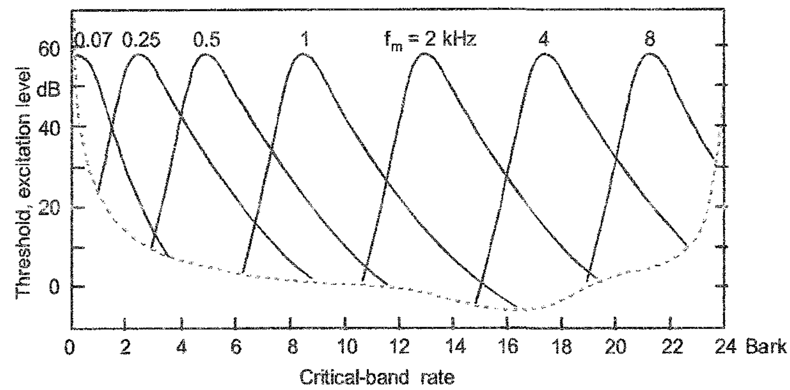
Examples of relation between Bark and frequency values:

- A frequency of 200 Hz corresponds to 2 Bark
- A frequency of 2 kHz corresponds to 13 Bark
- Bark band 1 covers the frequency range from 0 - 100 Hz
- Bark band 24 covers the frequency range from 12000 - 15500 Hz

The name "Bark" is chosen in honour of the late famous acoustician Barkhausen from Dresden.

Masking Patterns

Patterns of Narrow-band Noises Centred at Different Frequencies f_m



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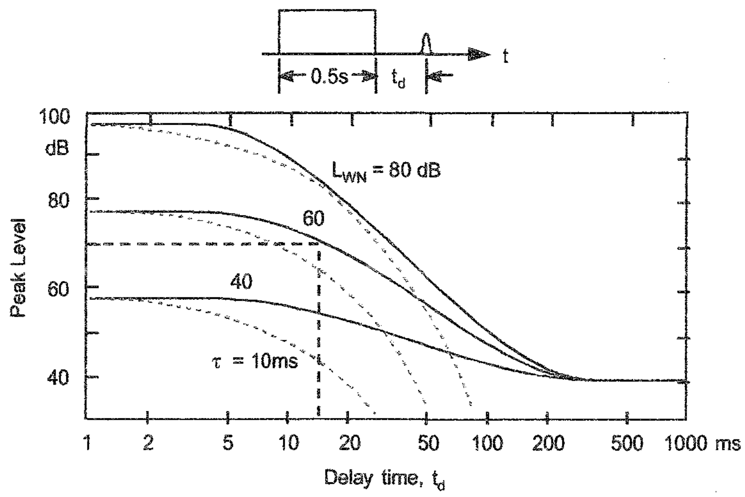
Here, one of the many advantages of the Bark scale is shown.

The masking patterns of narrow-band noises 1 Bark wide, centred at different frequencies are plotted as solid curves. The dashed curve is the threshold in quiet.

Plotted on the Bark scale they all have the same shape independent of the frequency and can be regarded as filter characteristics installed in the human hearing system.

Post-masking at Different Masker Levels



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Temporal masking is the other important masking property of the human hearing system.

When a masker pulse is switched of, decay effects show up in the human hearing system. This is called post-masking.

The figure illustrates post-masking effects for different masker levels. In this case white noise L_{WN} 0.5 seconds long at levels of 40, 60 and 80 dB.

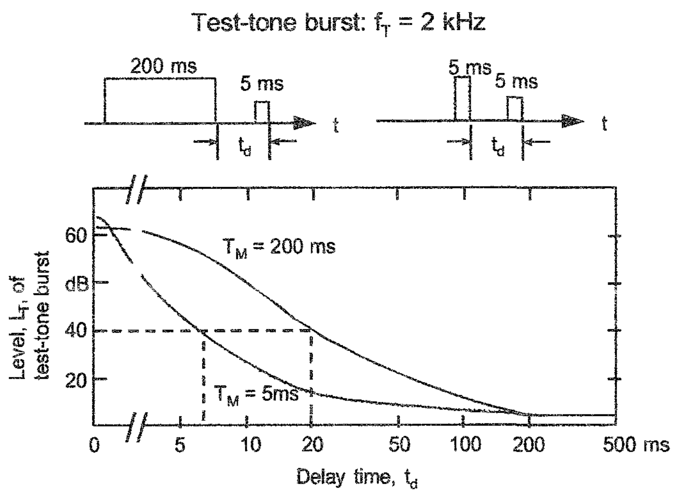
The level of a just audible short impulse, presented with a delay time t_d after the end of the masker, is given as a function of the delay time.

It is seen, for example, that a short pulse with a level of 70 dB presented 15 ms after a masker with a level of 60 dB is masked.

It is seen that masking curves have shapes that are strongly dependent on level. Compared to an exponential decay with a time constant of 10 ms, shown with dashed lines, only the shape of an 80 dB masker is close to being similar. At lower masker levels differences are substantial.

In all masking cases the decay and masking effects have finished after a time delay of about 200 ms.

Post-masking for Different Masker Duration



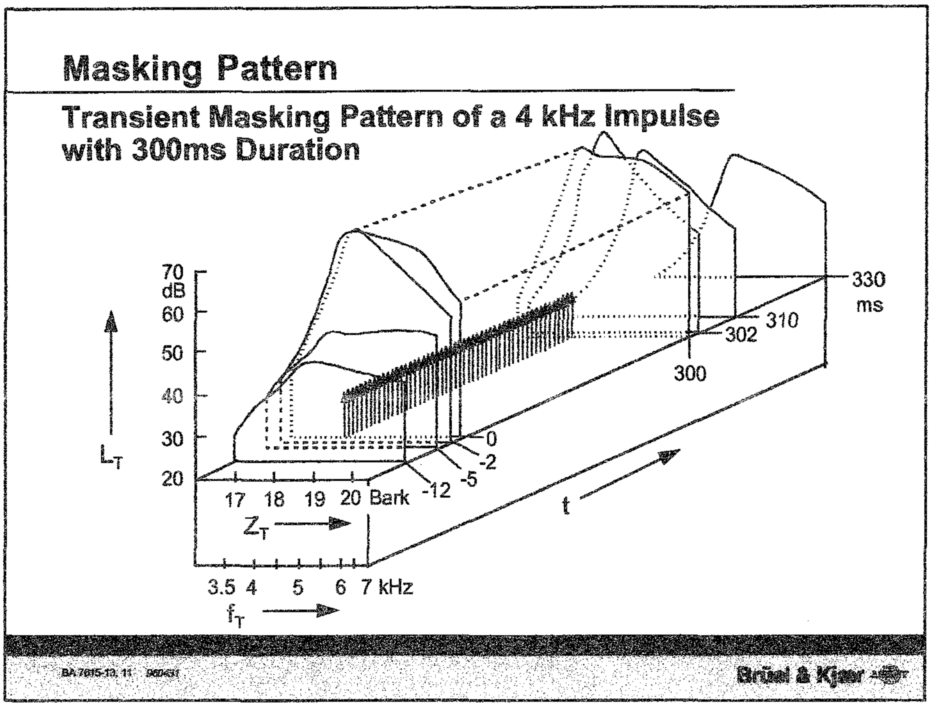
In this figure another non-linear post-masking in the human hearing system is illustrated.

The level of the just audible test tone-burst is given as function of the delay time in relation to two masker durations, T_M , of 200 ms and 5 ms.

It is clear to see that after a longer masker (200 ms) the decay takes a longer time than for the short (5 ms) masker.

These results again reveal that the decay processes in the human hearing system are highly non-linear.

As an example, a test-tone burst with a level L_T of 40 dB is only masked up to a delay of about 7 ms when the masker has a duration of 5 ms, while the masking from a 200 ms tone masker is effective up to about 20 ms.



This 3-D display is an illustration that summarises all the masking processes of the human hearing system.

The test tone f_T is at 4 kHz and has a duration of 300 ms.

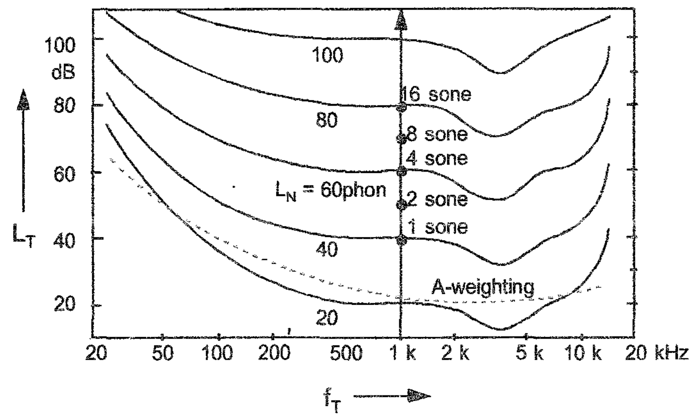
The “roof” of the “building” when the test tone is on represents the frequency masking.

When the test tone has finished after 300 ms, time masking is shown for the first 30 ms.

It is interesting to note a pre-masking phenomenon before the 4 kHz test tone starts. At first it looks acausal that you can have auditory impression before the event that causes it appears!

The explanation is that there are delays in the human hearing system and that the perception of the 4 kHz tone takes longer than the perception of the pre-masker.

Equal Loudness Contours and A-weighting



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The hearing sensation of loudness represents a dominant feature for sound quality evaluation.

The solid curves in the the graph are called "equal loudness contours".

They demonstrate that the hearing system is most sensitive for frequencies around 4 kHz and shows reduced sensitivity at lower and higher frequencies. In particular at low frequencies the equal loudness contours are not shifted in parallel, but show a level dependence.

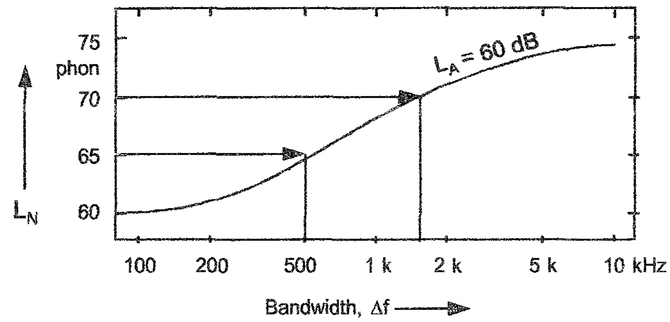
The contours are labelled in phon. A 60 phon contour represents the level in dB needed to give equal sensation of signal loudness versus frequency. At 1 kHz the level in dB and phon have the same value.

Another measure of loudness is sone. It has a reference in a 1 kHz level of 40 phon or 40 dB which is equal to 1 sone. A doubling of the sone value represents a doubling of the perceived loudness of a sound. It takes an increase in level from 40 phon to 50 phon to reach 2 sone. And another increase in level from 50 phon to 60 phon will give 4 sone. In short, it is necessary to increase the loudness value by 10 phon to give the sensation of a doubling of the loudness.

The dashed curve in the graph shows the well-known A-weighting. For very low sounds there is a good agreement with the 20 phon curve. At higher levels, e.g., 80 phon - typical for everyday sounds - it underestimates the loudness of their low frequency components.

Loudness Level vs. A-weighting

Loudness level of noises
with different bandwidth
around 1 kHz with the same
A-weighted level, $L_A = 60$ dB



Brüel & Kjær

Another example of the shortcomings of A-weighting is shown in this graph.

Here, noise of different bandwidths ,all adjusted to the same A-weighted level $L_A = 60$ dB, are shown with their different perceived loudness L_N.

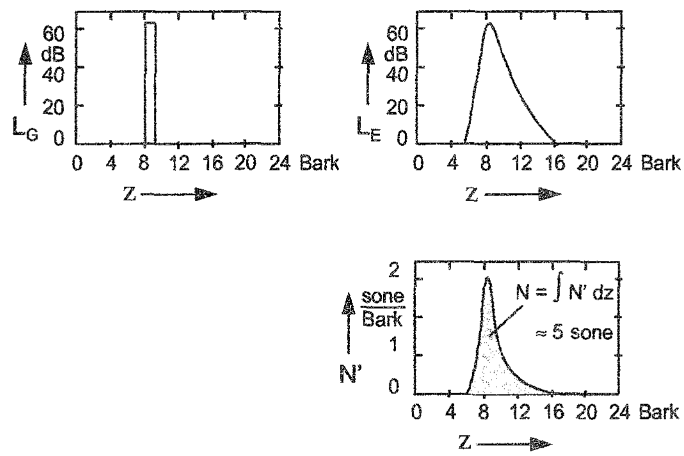
For example, an A-weighted noise of 60 dB is perceived to have a loudness of 65 phon when the bandwidth is 500 Hz and a loudness of 70 phon when the bandwidth is around 1.5 kHz.

In summary, these results clearly indicate that when using A-weighted levels, the loudness of broad band sounds is systematically underestimated.

A-weighted level measurements are inadequate in the context of evaluating sound quality.

Zwicker's Loudness Model

Schematic Illustration



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This figure illustrates the fundamentals in the Zwicker loudness model.

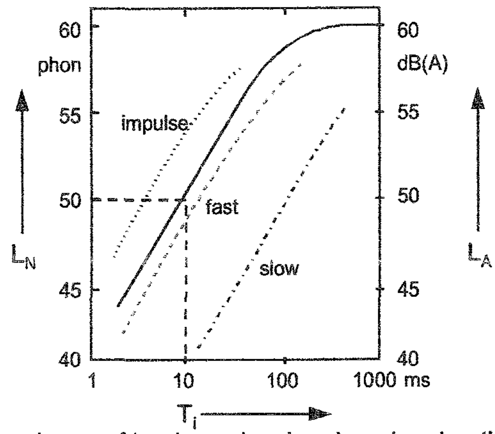
The left panel shows what a narrow band of noise, centred around 1 kHz corresponding to 8.5 Bark, will look like on an ordinary analyzer.

The upper right panel shows how it is perceived by the human including its masking pattern.

The lower right panel shows the specific loudness pattern. That is, the loudness value per critical Bark band measured in sone. The most important feature of this Zwicker loudness model is that the area under the specific loudness curve N' (shaded area) is directly proportional to the perceived loudness.

This direct relationship is the great advantage of loudness patterns in comparison to alternative spectral representations like FFT spectra or 1/3 octave-band spectra.

Loudness Level vs. Impulse Duration



Dependence of loudness level on impulse duration
in comparison to measurements

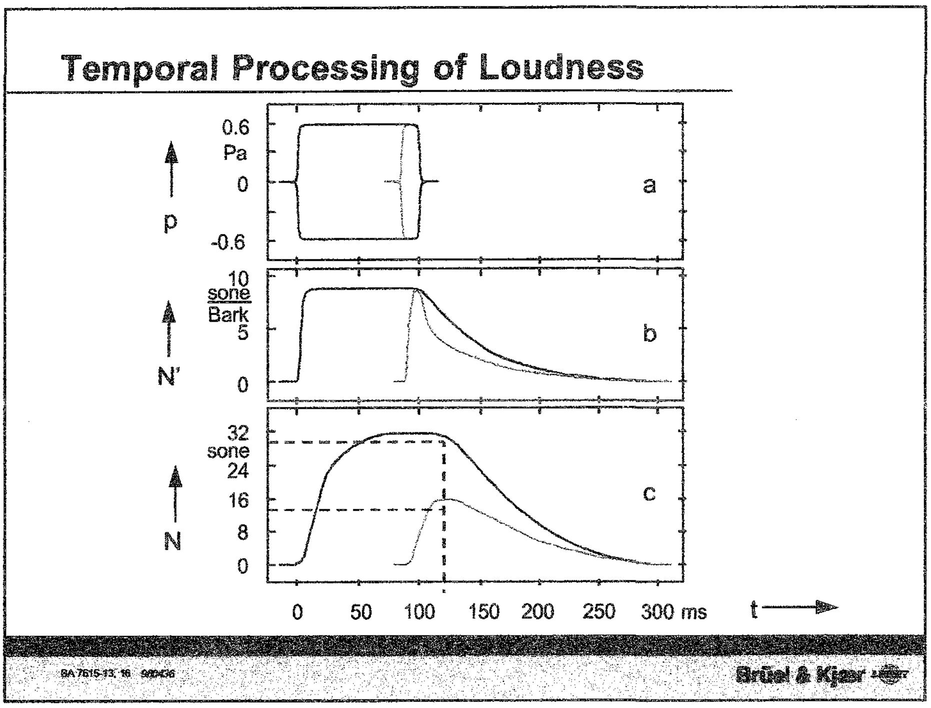
Brüel & Kjær

Another non-linear property of human hearing is that short pulses are not perceived as loud as longer ones.

In the graph the solid line indicates the perceived loudness versus the duration of a 1 kHz impulse.

First when the impulse is longer than 100 ms is its perception at maximum - around 60 phon in this example. This in contrast to a pulse of 10 ms length which only has a loudness of 50 phon.

As a comparison, the time constants found in A-weighted sound level meters are shown. It is clear that none of the standard time constants - "impulse", "fast" or "slow" - are in complete agreement with the features of the human system.



This is another illustration of the temporal processing of the human hearing system.

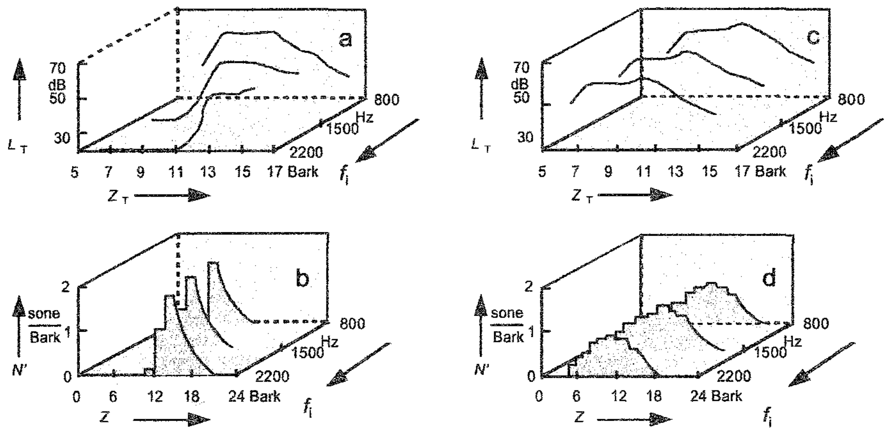
The upper panel shows the physical envelope of two tone impulses with a length of 100 ms (dark/red curve) and 10 ms (light/green curve).

The middle panel shows the specific loudness versus time of the two signals. It is clear that the decay is much steeper for the 10 ms pulse than for the 100 ms pulse.

The lower panel shows the total loudness versus time. It is seen that the 10 ms pulse only reaches about half of the loudness of the 100 ms pulse. This corresponds to a reduction in loudness of 10 phon.

Masking and Loudness Patterns for a FM-tone

Masking and loudness patterns for a FM-tone
centred at 1500 Hz with ± 700 Hz frequency deviation.
Modulation frequency 0.5 Hz (a and b) or 128 Hz (c and d)



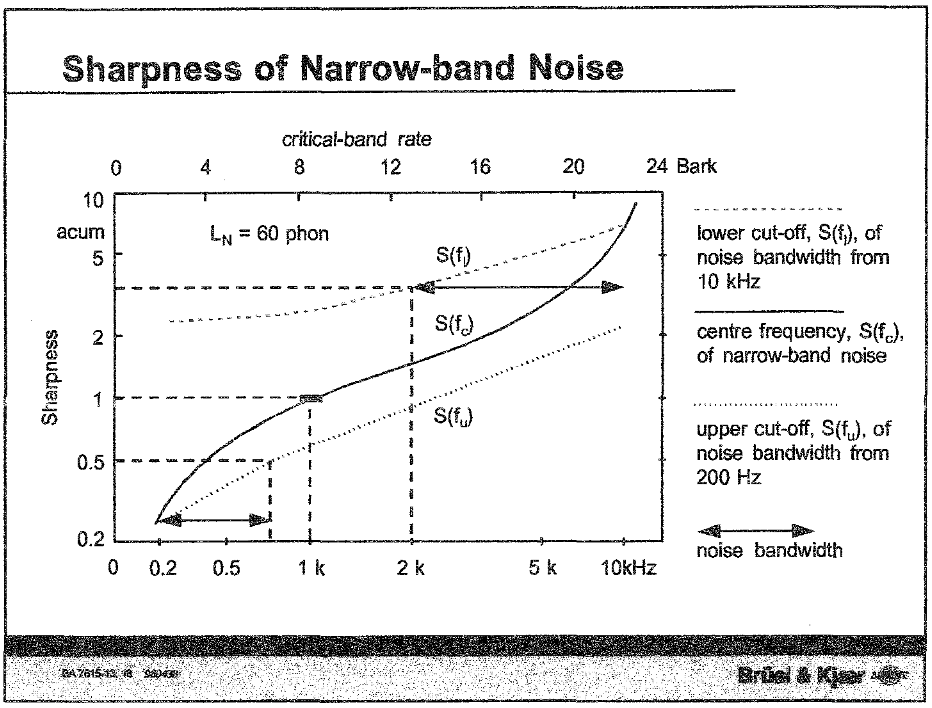
BA 7815-13, 12 980437

FM-tones represent an excellent tool for studying both spectral and temporal effects of masking and loudness.

The test tone is centred at 1500 Hz and has a frequency deviation of ±700 Hz. Snapshots are taken at the instantaneous frequency f_i of 1500 Hz as well as at the frequencies where the deviation is at its largest 800 Hz and 2200 Hz.

The two panels to the left shows the masking pattern (a) and the loudness pattern (b) when the modulation frequency is low, 0.5 Hz. It is clear that at such low frequencies the hearing system can follow the pitch excursions and, accordingly, shifts in both the masking and loudness patterns are visible. This signal has a large amount of Fluctuation Strength.

The two panels to the right, (c) and (d), show the same situation except that the modulation frequency has been raised to 128 Hz. In this case, the hearing system can no longer follow the pitch excursions and unpleasant sounds with broad spectral distributions show up. This signal has a large amount of Roughness.



The hearing sensation "sharpness" represents an attribute for the evaluation of tone color.

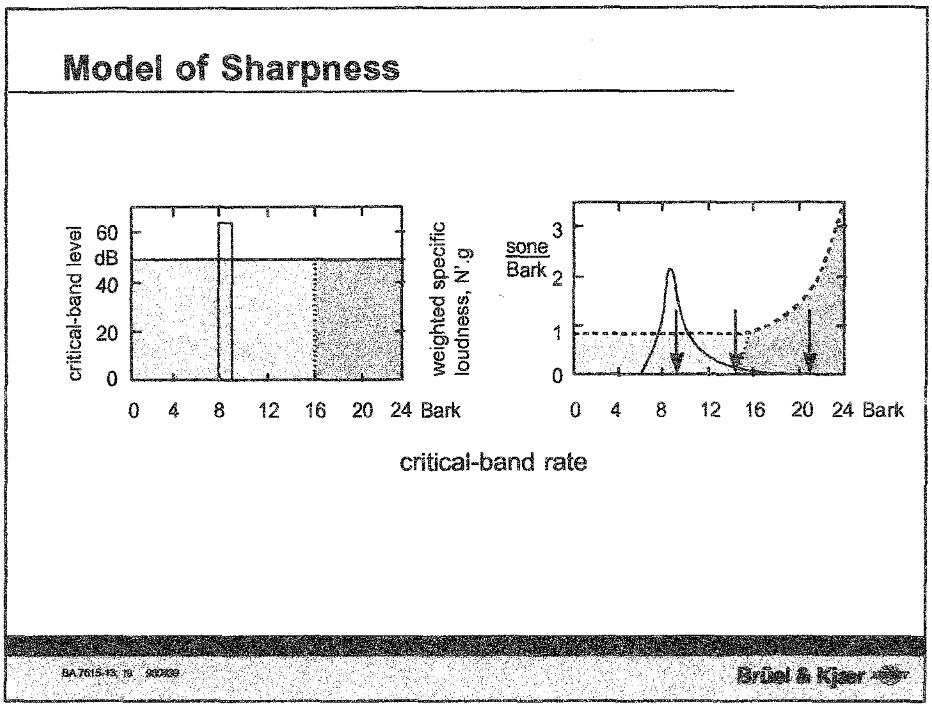
The solid line in the graph shows the sharpness of a narrow-band noise 1 Bark wide as function of its center frequency. At a center frequency of 1 kHz, the sharpness is around 1 acum. When the frequency is raised to 10 kHz, the sharpness increases to around 8 acum.

The dashed curve shows sharpness of high-pass noise with the lower cut-off frequency as parameter. For example, if the lower cut-off frequency is 2 kHz, the noise bandwidth is from 2 - 10 kHz and has a sharpness of around 3.5 acum.

The dotted curve shows the situation for low-pass noise. If the cut-off frequency is, for example, 700 Hz, the noise bandwidth is from 200 Hz to 700 Hz and has a sharpness of around 0.5 acum.

From this it is obvious that significant levels of spectral components at high frequencies gives high values of sharpness.

It is also clear that the addition of more low frequencies - for example to a high-pass noise signal - can reduce the sharpness value.

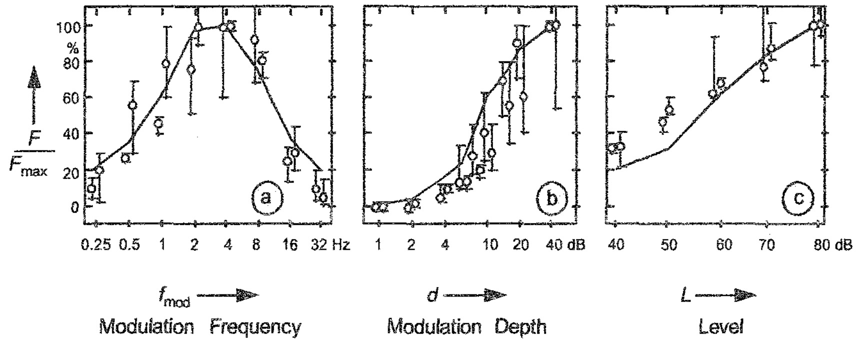


This is a model of sharpness.

The left panel shows three signals: a narrow-band noise at 8.5 Bark (1 kHz), a broad-band noise 0 - 24 Bark wide and a high-pass noise 16 - 24 Bark wide.

The right panel shows the specific loudness for the three signals, including the high frequency boost that is a part of the sharpness calculation. The arrows in the graph represents the centre of gravity for the three weighted signals which indicates the level of sharpness. The higher on the Bark scale the arrow is, the higher the sharpness value.

Dependence of Fluctuation Strength



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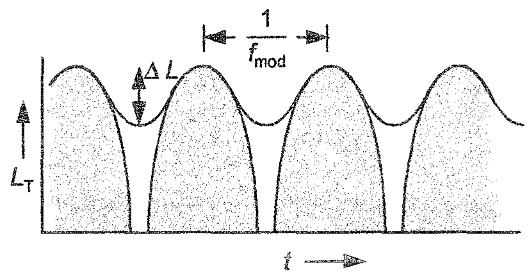
This figure illustrates some of the important factors that influence the level of Fluctuation Strength.

In the left panel it is seen that the greatest value of Fluctuation Strength is reached when the the amplitude and/or frequency modulation occurs at around 4 Hz. At 1 Hz or 16 Hz its influence is halved.

The middle panel shows the influence of the modulation depth.

The right panel shows that the signal level also affects the perceived level of Fluctuation strength.

Model of Fluctuation Strength

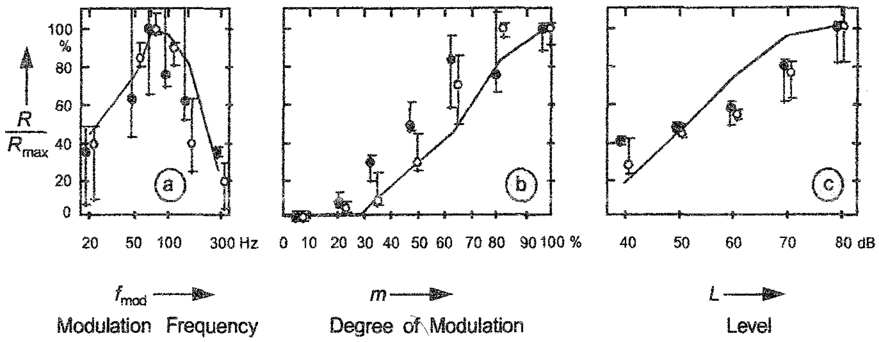


$$F \sim \frac{\Delta L}{\frac{f_{mod}}{4\ \text{Hz}} + \frac{4\ \text{Hz}}{f_{mod}}}$$

Brüel & Kjær

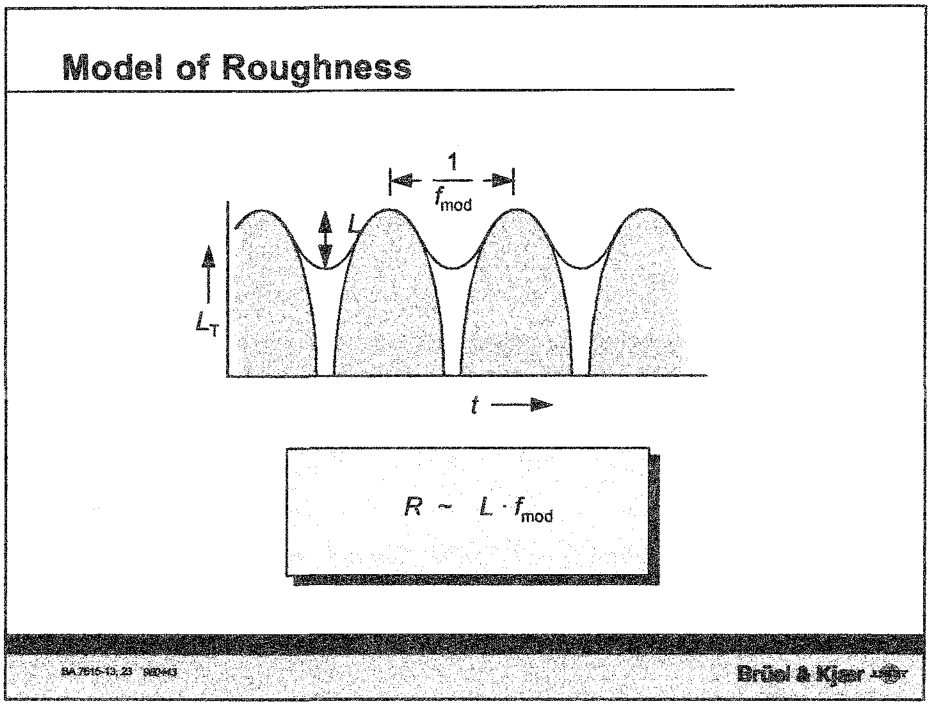
The basic terms and formulas involved in a calculation of Fluctuation strength. The formula used in an actual calculation is a lot more complicated.

Dependence of Roughness



Like *Fluctuation Strength*, Roughness is also strongly affected by modulation frequency, degree of modulation and level in a non-linear manner which is difficult to implement in a calculation formula.

Note that the human hearing is most sensitive to Roughness when the modulation frequencies are around 70 Hz.



The basic terms and formula in a calculation of Roughness.
Again a complete calculation formula is a lot more complex.

Conclusion

- The human perception of sound is a very complex process and highly non-linear. That calls for other acoustical descriptors than those used in normal "linear" sound work
- Optimising the Sound Quality of a product requires a thorough understanding of psychoacoustic properties
- For instance, a disturbing frequency need not be completely removed, but just moved closer to other frequencies or reduced in level so that it is masked
- Adding new sounds or noises to a signal can improve the Sound Quality of a signal that has too much Sharpness

BA7815-13, 24

Brüel & Kjær

This lecture only highlights the most fundamental principles behind human sound perception.

Psychoacoustics is an important and complex discipline that is still developing. It is characterised by its very limited amount of international standardisation.

This makes it a little difficult for for engineers to work with, but a good knowledge of psychoacoustics is essential for success in Sound Quality design.

Sound Quality Metrics

BA 7643-11, 1

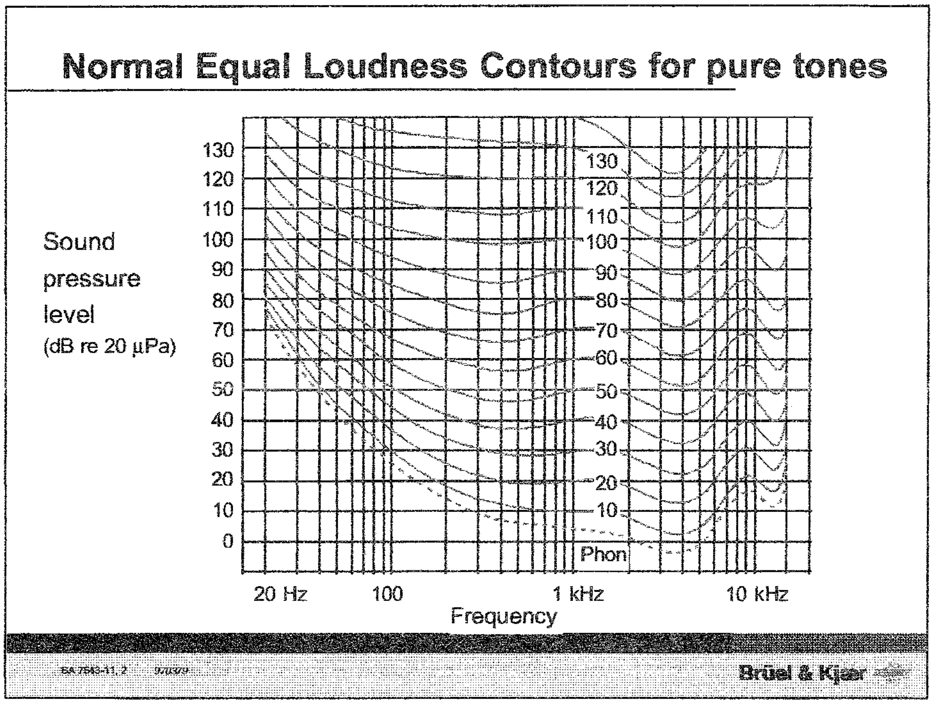
Brüel & Kjær

Abstract:

This lecture note adresses most of the metrics and their definitions as they are used in Sound Quality Software Type 7698.

Also useful description of funtion names, units and display formats are given.

LECTURE NOTE



Here is shown the normal equal loudness contours for pure tones. The dashed curve indicates the normal binaural minimum audible field.

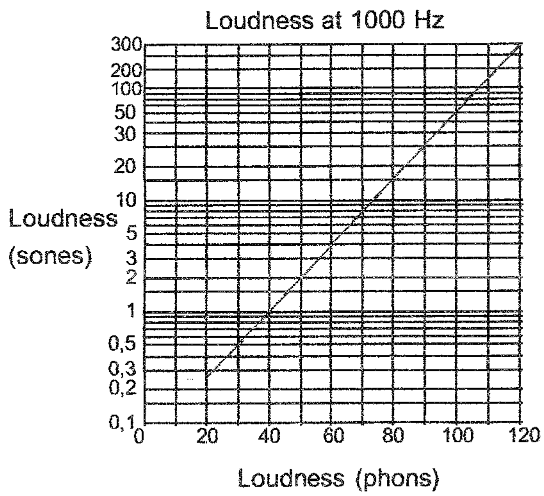
Note the very non-linear characteristics of the human perception. Almost 80 dB more SPL is needed at 20 Hz to give the same perceived loudness as at 3-4 kHz.

This observation together with frequency masking - limitations in the ears capability to discriminate closely spaced frequencies and low sound levels in the presence of higher sounds - is the foundation for the calculation of the loudness of stationary signals.

Loudness of non-stationary signals also needs to take temporal masking of the human perception into account.

A correct calculation of these loudness values are crucial for all the following metrics calculations.

Loudness P in phon and N in sone



Relation between loudness in sone and phon

$N = 2^{(P-40)/10}$

$P = 40 + 10 \log_2 N$

At 1000 Hz:
1 sone = 40 phon
2 sone = 50 phon
4 sone = 60 phon
etc.

A doubling in a sone value corresponds to a doubling in perceived loudness and a 10 dB increase in phon.

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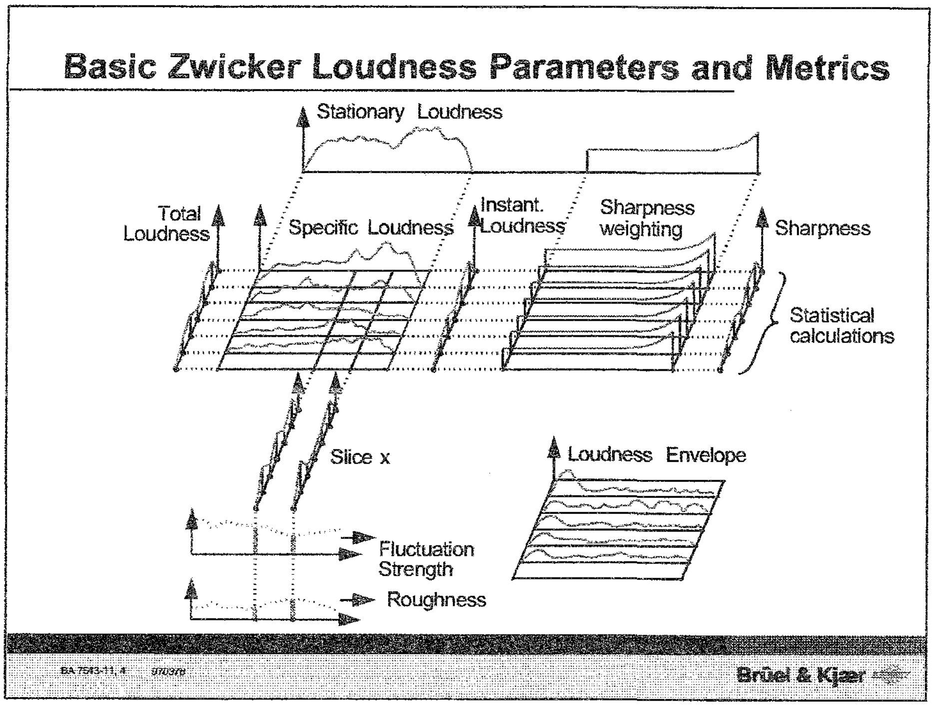
Loudness is often measured in phons. It has the advantage of giving values at 1 kHz similar to the well known SPL values.

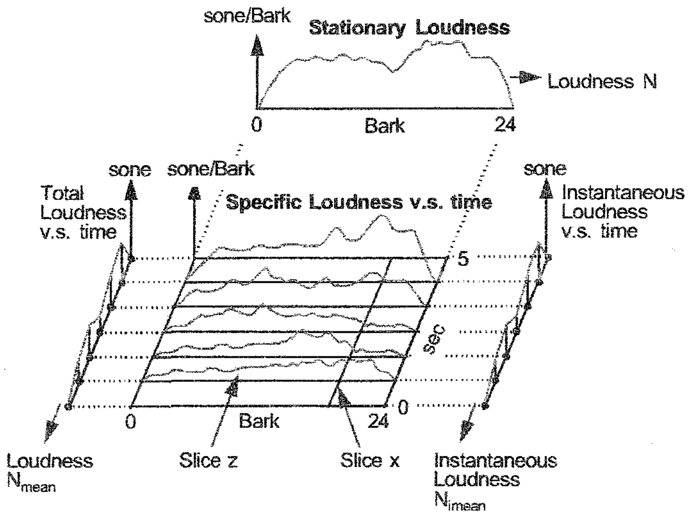
However, in discussion of perceived sound levels it is more convenient to have a unit that matches perceived sound levels better. Loudness in sones has the advantage of matching precisely when a sound is perceived twice as loud as another - then its value in sones is the double of the sone value of the lower sound.

Loudness in sones is almost exclusively used as the level unit in Brüel & Kjær's metrics calculations.



This overview shows the loudness based metrics available in the Sound Quality software type 7698.

Basic Zwicker Loudness Parameters and Metrics



Brüel & Kjær

This is the basic loudness calculations as is performed by the Sound Quality software Type 7698 and shown on the screen.

The background for the calculation is a 1/3 octave analysis of the time sample as a multispectrum with up to 2 mS resolution. Then frequency masking corrections and conversion to a Bark frequency scale.

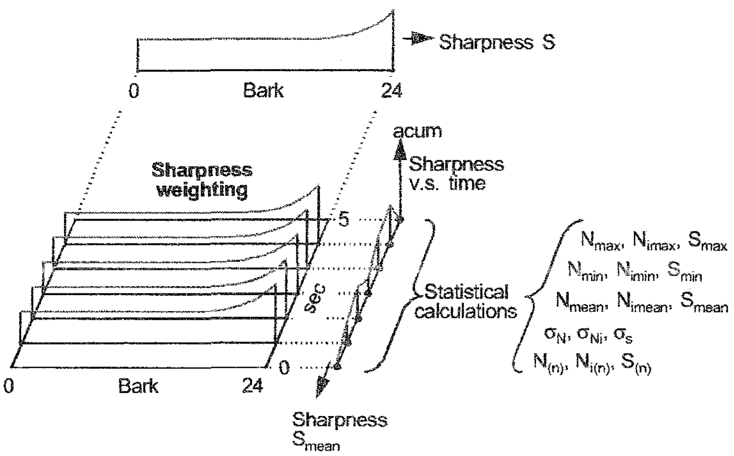
For stationary signals all the spectra are linearly averaged over the entire length of the time sample. This gives a loudness value for each bark band - sone/Bark. An integration over all Bark bands gives Loudness N.

For nonstationary signals each loudness spectra is corrected for temporal masking along the Slice x line. When that is done the result can be seen in a waterfall plot as Specific Loudness vs.. time.

Integrating all the spectra along the Bark scale -Slice z- Instantaneous Loudness vs. time can be displayed and a mean value calculated.

Adding further 3x9 mS time constants gives the Total Loudness vs. time which has a smoother shape than Instantaneous Loudness vs. time.

Basic Zwicker Loudness Parameters and Metrics



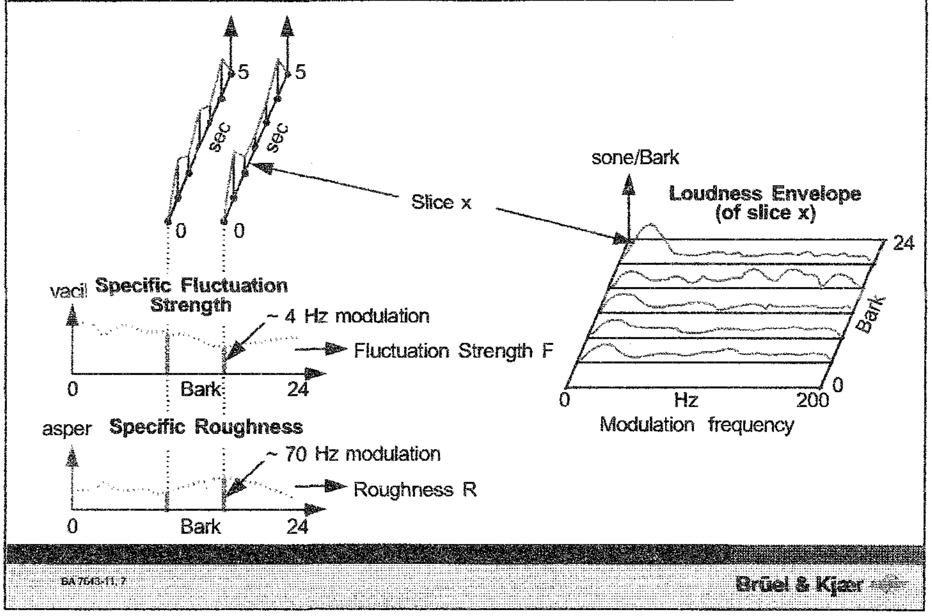
Brüel & Kjær

By adding a spectral weighting function to Stationary Loudness and Specific Loudness vs. time Sharpness can be calculated.

For Sharpness vs. time as well as Instantaneous Loudness vs. time and Total Loudness vs. time a number of statistical calculations can be performed.

This includes Maximum, Mean, Minimum, Standard deviation and Percentile values.

Basic Zwicker Loudness Parameters and Metrics



Both Fluctuation Strength and Roughness only applies to nonstationary signals. They are measurements describing the amount of amplitude and frequency modulation for each Specific Loudness vs. time - Slice x.

For Fluctuation Strength modulation around 4 Hz has the highest weighting.

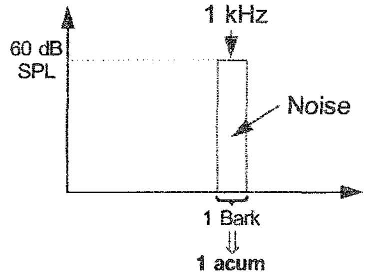
For Roughness modulation around 70 Hz has the highest weighting. To get reliable results for Roughness the resolution of the Specific Loudness vs. time must be as high as one spectra every 2 mS not to loose high frequency modulation.

A waterfall spectrum can be calculated to study the modulation frequencies in more detail.

Please note that frequency modulation - if the frequency deviation is large enough - leaves the frequency component in the neighboring Bark bands some of the time and then it will look like amplitude modulation in the analyzed bark band.

Metrics Definition

Sharpness S



Sharpness as defined by Zwicker:

$$S = 0.11 \cdot \frac{\int_1^{24} N'(z) \cdot g(z) \cdot z \cdot dz}{\int_1^{24} N'(z) \cdot dz} \; acum \quad \textbf{where} \quad g(z) = \begin{cases} 1 & \text{for } z \le 16 \\ 0.066 \cdot e^{0.171 z} & \text{for } z > 16 \end{cases}$$

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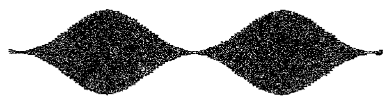
Shown here is the definition of Sharpness as defined by Zwicker. As Sharpness is not yet standardized alternatives exists. One defined by Aures is sometimes used and is also included in the Sound Quality software type 7698.

Sharpness is measured in the unit "acum" which in Latin means "sharp".

Metrics Definition

Fluctuation Strength F



Modulated freq.: 1000Hz at 60dB SPL

Modulation: 100% AM at f_{mod} 4Hz

$\Downarrow$

1 vacil

The formula for Specific Fluctuation Strength is:

$$F'(z) = \frac{0.032 \cdot \Delta L(z) \cdot \Delta z}{f_{mod}(z)/4 + 4/f_{mod}(z)} \text{ vacil} \quad \text{where} \quad \Delta L = 20\log(N'(1)/N'(99))$$

The Fluctuation Strength is a sum of all Specific Fluctuation Strengths:

$$F = \sum^{24} F'(z) \text{ vacil}$$

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The definition of Fluctuation Strength shown here is taken from "E.Zwicker and H.Fastel: Psychoacoustics, Facts and Models", but not yet standardized.

Fluctuation Strength is measured in the unit "vacil" which comes from the Latin word "vacilare".

Metrics Definition

Roughness R

Modulated freq.: 1000Hz at 60dB SPL

Modulation: 100% AM at f_{mod} 70Hz

$\Downarrow$

1 asper

The formula for Specific Roughness is:

$$R'(z) = 0.0003 \cdot f_{mod}(z) \cdot \Delta L_E(z) \cdot \Delta z \text{ asper where } \Delta L_E = 20\log(N'(1) / N'(99))$$

The Roughness is a sum of all Specific Roughnesses:

$$F = \sum^{24} R'(z) \text{ asper}$$

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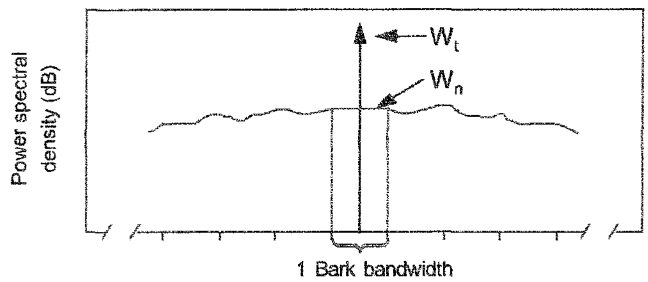
Brüel & Kjær

Also Roughness is not yet standardized.

It is measured in the unit "asper" which comes from Latin and characterizes what we call "rough".

Metrics Definition

Tone-to-noise ratio



Tone-to-noise ratio is the difference between the tone itself W_t and the SPL of the noise in a critical band centered around the tone W_n - but in absence of the tone.

If the tone to noise ratio exceeds 6 dB the tone is called prominent.

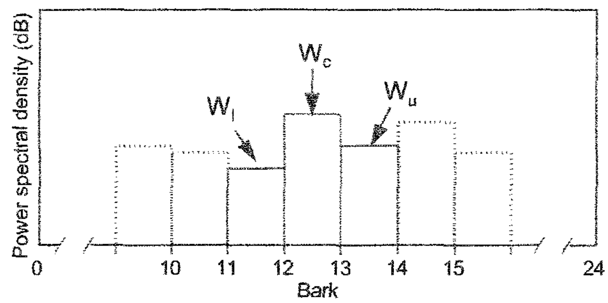
BA 7543-11, 11 Brüel & Kjær

The instantaneous Tone -to-Noise Ratio is measured by examining each spectrum in an FFT multispectrum and finding the maximum value and the frequency at which it occurs. Then a critical bandwidth centered on each of the frequencies is calculated.

The Tone-to-Noise Ratio is then determined as shown in the graph. The results are then displayed as a graph showing Tone-to-Noise Ratio versus time.

Metrics Definition

Prominence ratio



Prominence ratio compares the total power in a critical band W_c to the average power in the adjacent critical bands W_l and W_u.

If the difference is greater than 7 dB it is called prominent.

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This metric is also based on a FFT multispectrum analysis. From the lines in the FFT spectrum a Bark frequency scale is synthesized. That means that we here do not use a loudness analysis.

The Bark band with the largest power is then determined and compared to the power in the two adjacent Bark bands as shown in the graph.

The results are shown as a graph with Prominence ratio as function of time

Sound Quality Equalisation and Calibration

BA 7616-13, 1

Brüel & Kjær

Abstract:

This lecture will highlight some of the most important parameters to consider when working with Sound Quality.

The human hearing system is the final judge of the success of a Sound Quality optimisation and at the same time a non-linear device which is very sensitive to level and frequency response changes.

Accuracy in equalisation and calibration is therefore of the highest importance.

LECTURE NOTE

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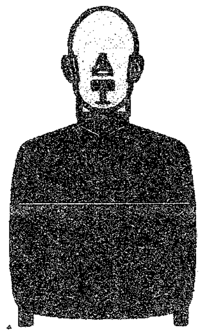


- Sound Quality Recording
- Head and Torso
- Calibration
- Sound Quality Equalisation

BA 7616-13, 2

Brüel & Kjær

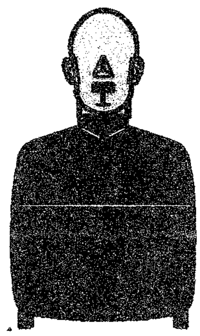
This lecture starts by discussing the different methods for recording Sound Quality signals. The basic choice is between the use of a single microphone or a binaural Head and Torso. The pros and cons for each method are treated in some detail as well as the requirements for an optimum result.

The design of a Head and Torso to act as a "stand in" for the human being is a very demanding task. All the results of Sound Quality evaluations depend on precise signal recordings. We will discus how the Brüel & Kjær Head and Torso is optimised and calibrated.

Human sound perception is very dependent on on the correct level and frequency response. Therefore, careful calibration of both the Head and Torso in the recording process and the headphones in the play back situation are crucial for a reliable result.

Working with Sound Quality, you have to make assumptions about the sound field where the recordings are made. If it is close to a Diffuse Field or a Free Field, the corresponding equalisations must be made in all relevant sections of the Sound Quality system from recording to play back.

Binaural Recording using Head and Torso



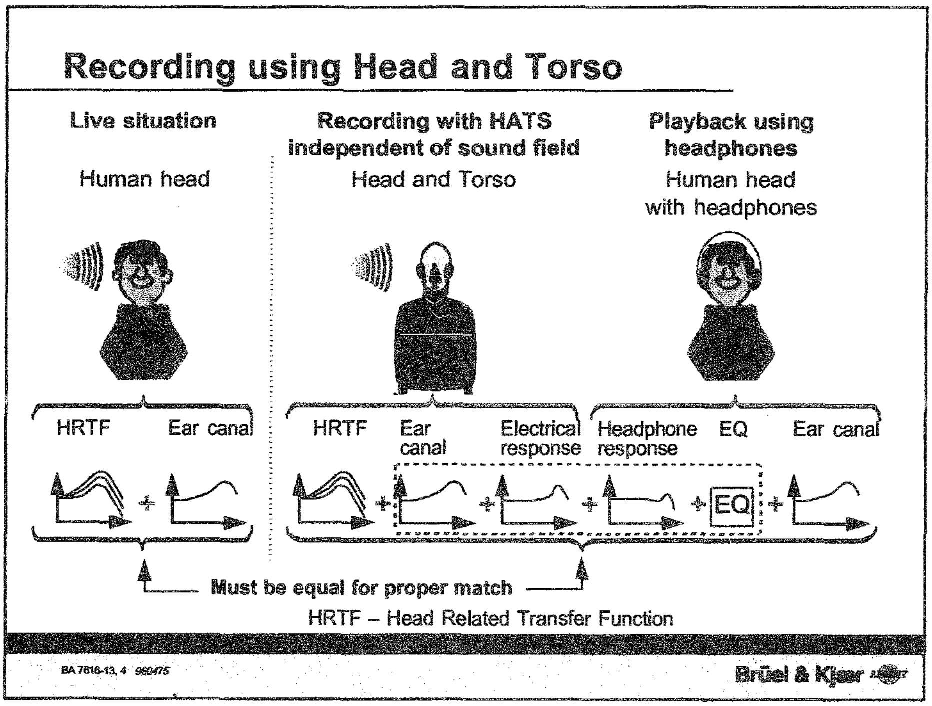
- Designed to substitute a human being
- Binaural recordings:
  - Facilitate playback to a large number of listeners
  - Facilitate Sound Engineering and Analysis
- No listener needs to be in the live sound situation

BA 7616-13, 3 Brüel & Kjær

Only a Head and Torso, when correctly designed, has a chance to be a good substitute for the human being in the sound recording situation.

Only binaural recordings played back through headphones and properly equalised can give the listener a faithful substitution for being in the live situation.

Only play back using headphones can give a large number of listeners the same listening conditions simultaneously and then save time.



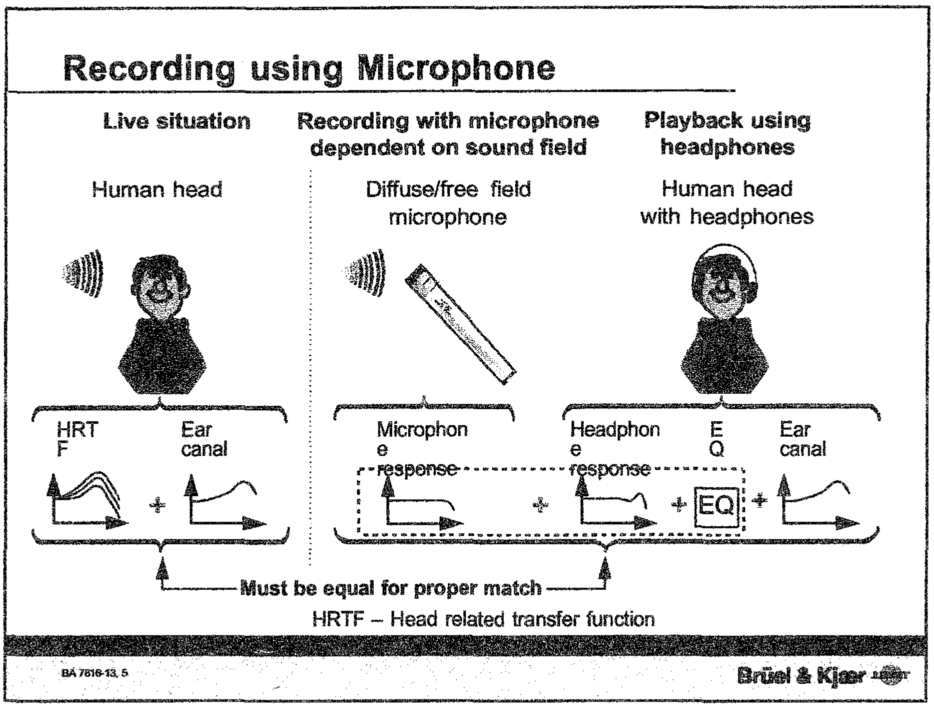
Ultimately, evaluating the Sound Quality of a product is a personal experience of the individual user of the product. However, it is of course impractical for the designer of the product to use individual evaluation. He will need a stand-in for the human being in the design and evaluation process - a Head and Torso Simulator (HATS) representing the average acoustic performance of the human population.

To design a good HATS we must realise how the human being perceives sound in the live situation. To characterise the sounds and the directions from which they come, the brain uses the two signals arriving at the ears. They carry all the information of the sound field in terms of HRTFs - Head Related Transfer Functions. These represent the frequency responses for sounds from all directions around the head, as well as the differences in time arrival for the two ears. The human ear canal only adds a change in the frequency response received by the ear drum due to resonances in the ear canal and has no directional clues.

Using a Head and Torso as a substitute for the human being, the only requirement is that it has HRTFs and interaural time differences equal to the average human head.

Using headphones for play back, they must be equalised to present a flat frequency response at the entrance to the ear canal of the listener. This means that the equalise function must take care of the response in the ear canal of the Head and Torso (if there is an ear canal), the microphone, recording electronics and the individual headphones.

All this is independent on the nature of the sound field.



Using a single microphone as a substitute for the human being when recording sounds is problematic. A microphone is unable to simulate the HRTFs of the human being and misses all information about differences in signal arrival times at the two ears. Using two or more microphones you can get a good stereophonic recording for playback via loudspeakers. But it can never replace a binaural recording with play back through headphones.

In order to improve the situation, you have to make assumptions about the nature of the sound field. It can be characterised as being either diffuse field or free field and you should choose the recording microphones accordingly.

In the playback the signal should be equalised to take care of the nature of the sound field to get an approximation to the missing HRTFs, as well as any non-flat frequency response in the electronics and the headphones.

Choice of Recording Method

Head and Torso

+ Optimal binaural sound field recording, including directional clues
+ Correct play back, using headphones and EQ

− Signals must be equalised before SQ analysis

Diffuse/free field microphone

+ Simple to use
+ Low cost
+ Signals ready for SQ analysis

− Choice of microphone according to sound field
− No directional clues
− Monaural

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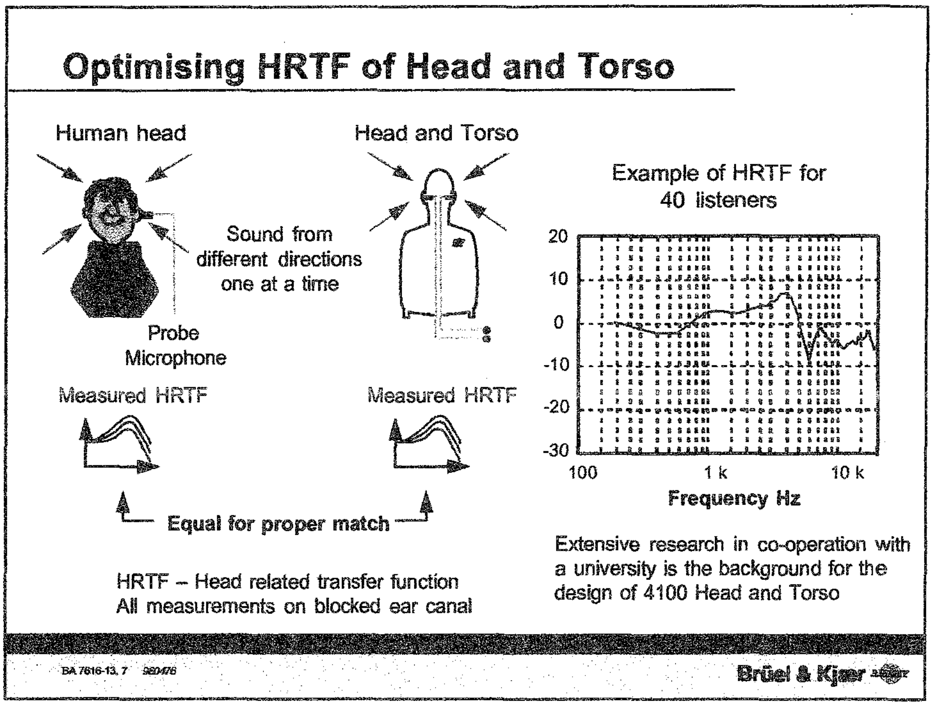
Brüel & Kjær

Single microphone recordings of sounds for Sound Quality evaluations will always be inferior to those recorded using of a good Head and Torso. It can only be recommended in situations where the directional properties of the sound field are unimportant or if instrumentation cost and complexity is a consideration. Or in situations where only objective measurements are required. They work on mono signals and require only knowledge of the sound field being diffuse or free field.



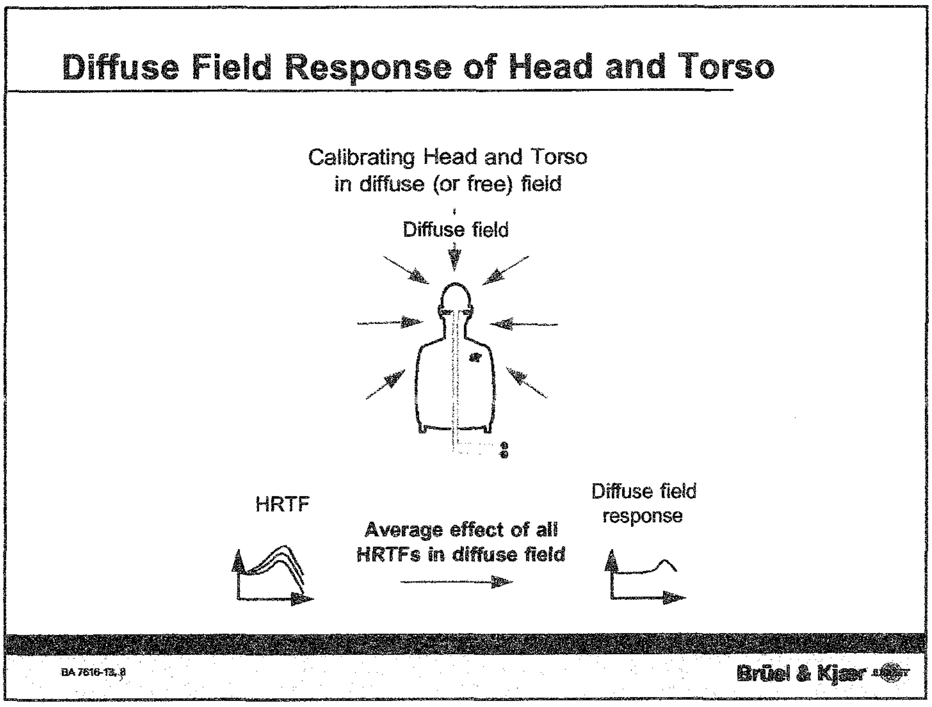
Extensive research conducted by Brüel & Kjær in co-operation with a Danish university has given a set of HRTFs of a human being as an average of measurements on 40 subjects.

A probe microphone mounted at the entrance to the subject's ear canal was used to sample to the sound pressure level. The test was conducted in an anechoic room using a loudspeaker equalised to flat response as sound source. Data was taken from 97 directions around the human head to give a complete set of HRTFs.

Similar measurements were taken using prototypes of Head and Torsos and when compared to the human HTRFs, clues to the final design were achieved.

The Brüel & Kjær Type 4100 represents a Head and Torso optimised in this way to give as reliable and accurate Sound Quality recordings as possible.

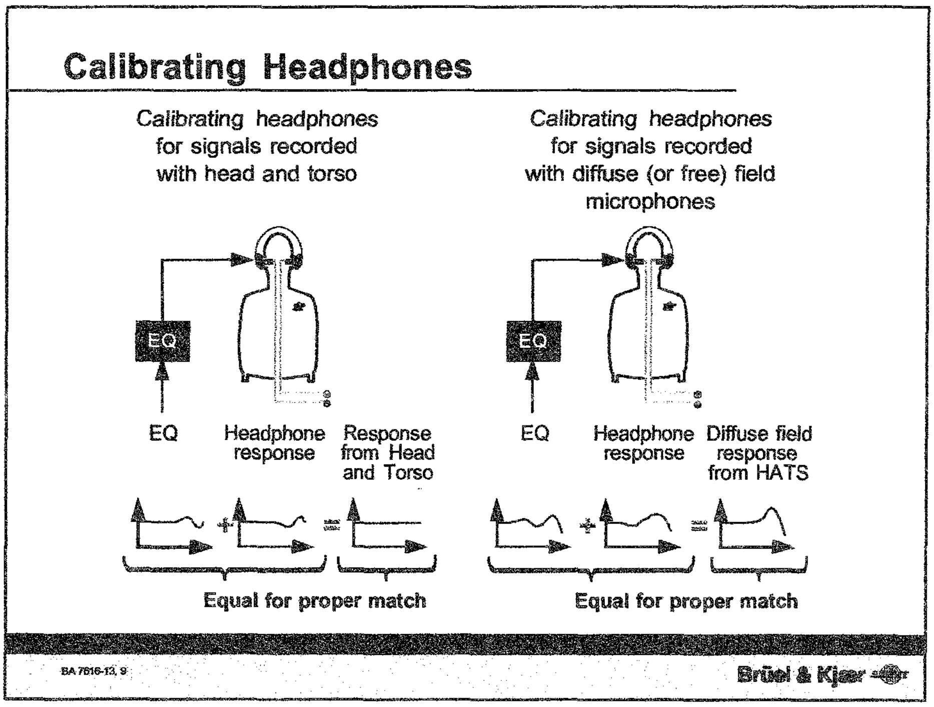
As can be seen in the graph to the left, there is a large spread in response from different subjects which indicates that a perfect mach is difficult, especially for listeners who deviate from the average.



If the Head and Torso is positioned in a diffuse sound field, its response is then an average based on all his HRTFs and called "Diffuse Field Response".

If you are working in a free sound field where the sound only comes from in front of the Head and Torso, the HRTFs representing 0 degree incidence can be used and are called "Free Field Response".

None of these responses are flat and are far from what can be measured using a good diffuse or free field microphone. Therefore they need to be corrected to a flat response before the signals from the Head and Torso can be used for Zwicker Loudness calculations. It also gives a convenient background for editing the sound signals in the Sound Quality software program.



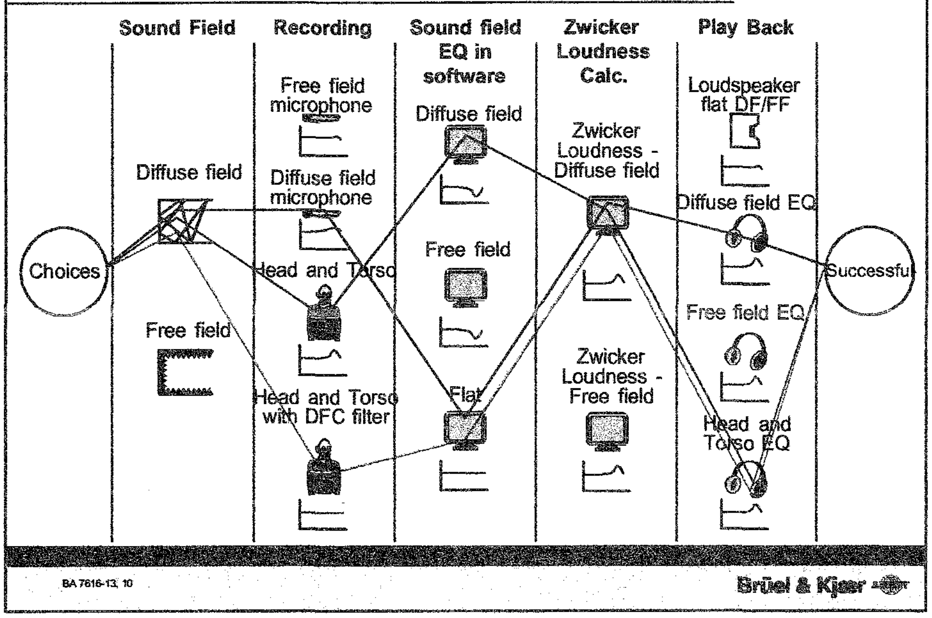
The calibration of headphones for play back of sound signals depends on how the signals were recorded.

If the signals were recorded with a Sound Quality optimised Head and Torso, they already have the correct HRTFs. The only thing needed is to equalise the headphones that were used when the sound was recorded to a flat response when mounted on the Head and Torso. This will compensate for response errors from the headphones and for small deviations in the Head and Torso microphone and preamplifier response.

Note that if calibration is made using a 4100 Head and Torso, the headphones are calibrated against a closed ear canal. When the headphone is used on the human head, it works against an open ear canal. Therefore discrepancies could be expected. However, research has shown that if the headphones used are of the "open" type, these errors can be neglected.

For calibrating headphones for sounds recorded with a single free or diffuse field microphone, a similar procedure as the first mentioned can be used. However, the equalisation must be adjusted to give a response including the effects of the HRTFs which are not in the signals recorded with the microphone. This can be done using the Head and Torso's Free or Diffuse Field Responses, respectively.

Sound Quality Equalisation – User Choices



A complete Sound Quality analysis system includes a lot of choices for the user to make - and to make correctly in order to get reliable and accurate results. The picture shows the five segments where the user has to make choices. Only the possible correct choices are shown for the situation where the sound field is diffuse.

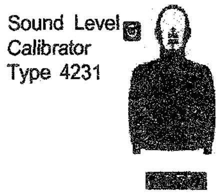
First he has to determine the nature of the sound field he is operating in. Is it with few reflections and close to a free field or is it closer to a diffuse field with reflections from all directions?

Next there is the choice of recording method and instrumentation. If the sound field is diffuse you should use either a diffuse field microphone or a Head and Torso with a diffuse field correction added either via a hardware filter in the preamplifier or as a software compensation in the computer. Done correctly, the signals are now ready for editing in the computer or calculating Zwicker loudness.

However, in order to do a correct loudness calculation you have to enter the assumption of the nature of the sound field - free or diffuse field in the calculation process. This adds a small frequency weighting to the signals so that the loudness calculation better matches a listener's impression of loudness in the actual sound field. This free or diffuse field correction should not be confused with the free or diffuse field correction of the Head and Torso.

Finally you have to choose the equalisation of your playback equipment according to how the recording was made.

Acoustic Level Calibration



Acoustic calibration is essential to sound quality.
All subjective and objective assessments are dependent on the correct level of the recording

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Perception of sound by the human ear and brain is very dependent on a lot of psychoacoustic parameters which all are dependent of level. Therefore, it is crucial to keep track of the acoustic level during recording and play back.

Before recording the sound signals it is standard procedure to record a test tone from an acoustic calibrator, for example Brüel & Kjær Type 4231.

Normally it is not convenient to do a full acoustic calibration before each sound sample and often not necessary.

Instead you can use the CIC - Charge Injection Calibration technique to check the validity of the acoustic calibration.

System Check using CIC Technique

FEATURES
- Single button operation
- Easily repeated during recordings
- No physical handling or change of setup

BENEFITS
- System check
- Tape recordings:
  - Keep track of system performance
- Direct to computer:
  - On-line system verification
- Reduced need for repeated acoustic calibration

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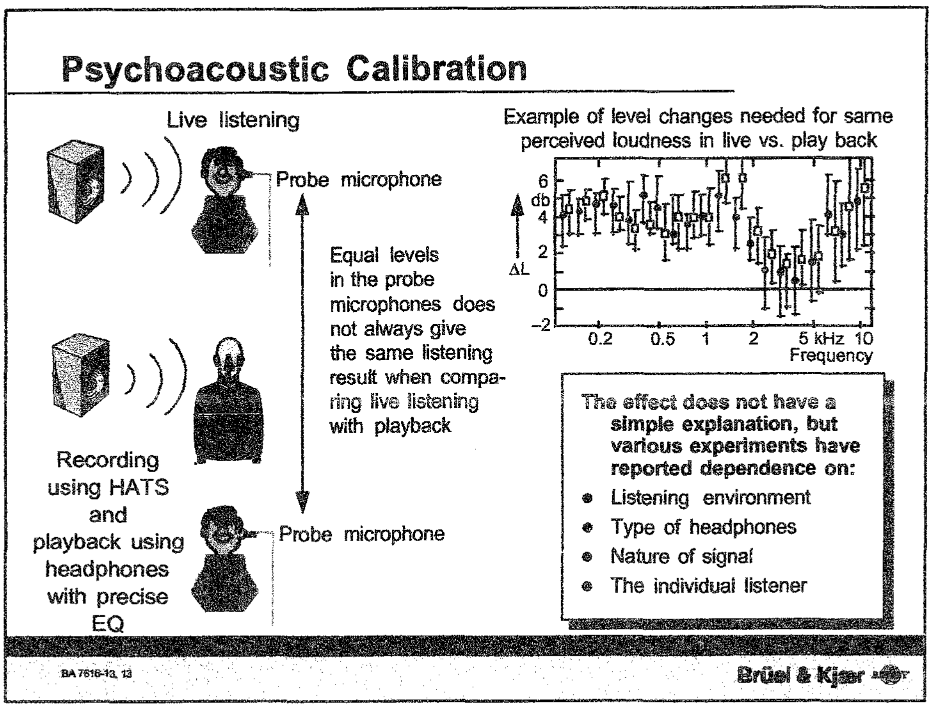
CIC is performed by injecting two frequencies, 40 Hz and 1000 Hz - one at a time - backwards into the microphone preamplifier system and then recording the responses. Later, after a sound recording session, you repeat the signal injection and compare the new levels with the first recordings.

If they are equal, there is a strong guarantee that the acoustic calibration is intact. If there is a difference, you can use the two CIC frequencies as a small diagnostic tool.

If the level change in the 40 Hz frequency is smaller than the change in the 1000 Hz tone, there is a change in the loading capacitors in the system. This is the case if the microphone capsule or the input stage in the preamplifier is defect.

If the changes in the two signals are equal, there is a change in the resistance in the system. This could be caused by a broken cable or a bad connector.

CIC can also be used to check if you have changed a 10 dB stepped gain settings in the preamplifier.



Comparing the perceived loudness from a live sound source - for example a speaker - with a carefully calibrated recording and playback through headphones of the same sound, you can experience a paradox. You will have to increase the headphone level by up to around 5 dB and maybe add a frequency equalisation to get the same perceived loudness in the two situations.

Although this is a well-known experience for both psychoacoustic experts and headphone manufacturers, nobody has been able to give a simple and universal explanation.

Various investigations have illustrated a dependence on different parameters - such as:

- Listening environment
- Type of headphone
- Nature of the signal
- The individual listener

In cases where a listening comparison of a reproduced sound against the live sound is critical for correct level match, you have to do a correction.

The best way to correct for this phenomenon is to perform a subjective comparison between the live sound and the headphone reproduction with added frequency and level equalisation.

This extra psychoacoustic equalisation could then be added to the normal headphone calibration.

Conclusion

- Only binaural recording and play back using headphones gives the correct subjective sound evaluation

- Correct Calibration and Equalisation are important for
  - Reliable subjective evaluation
  - Correct objective analysis (Zwicker analysis ...)

- Brüel & Kjær Head and Torso is based on extensive scientific research

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Brüel & Kjær

In this lecture we have tried to emphasise that binaural recording is a necessity when the best sound quality sound evaluations are required.

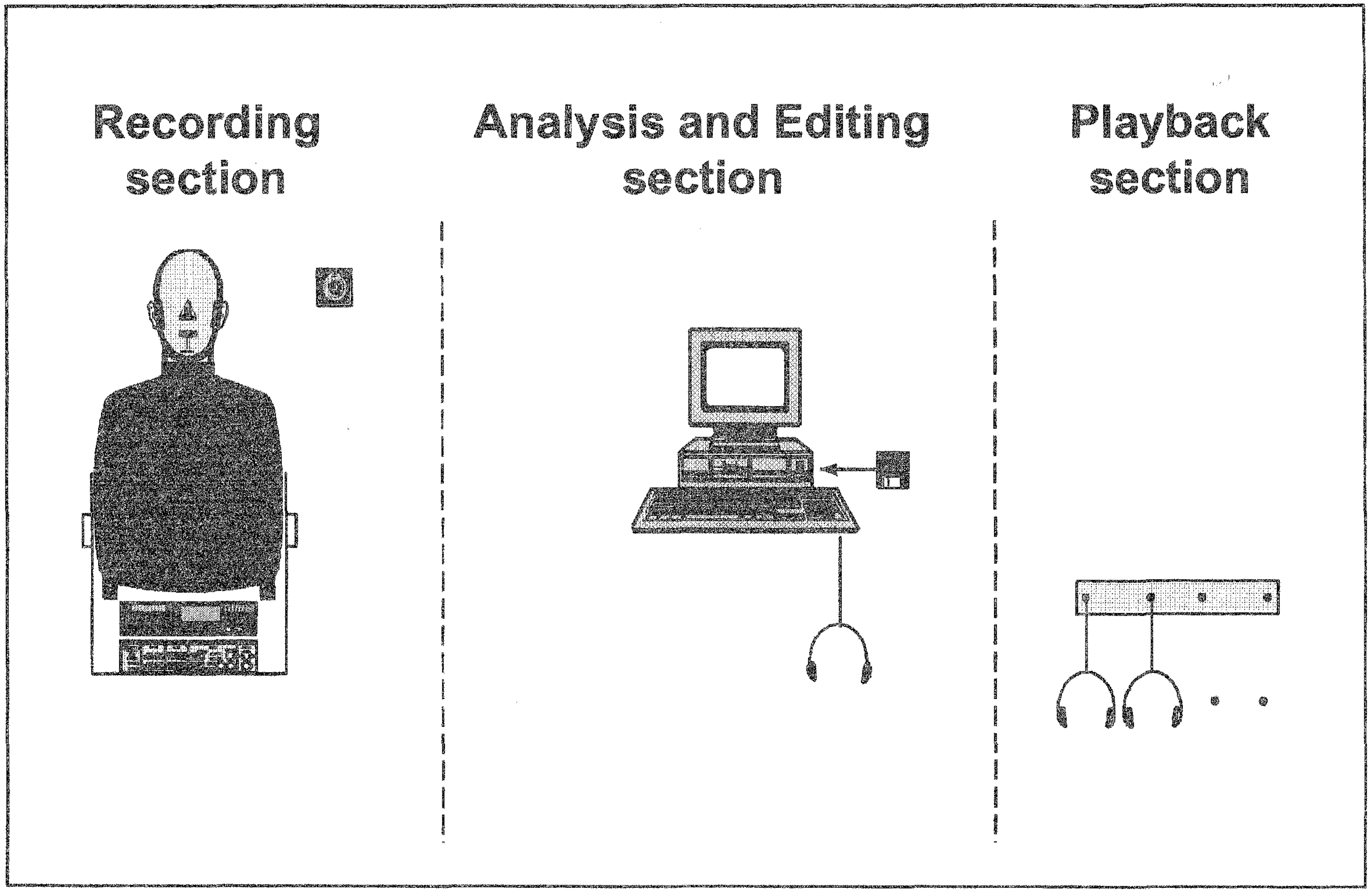
Only in a few specific situations can sound recording using a single microphone be recommended. This is especially when subjective listening tests have lower priority.

Accurate calibration and equalisation are directly proportional to the reliability and quality of the sound evaluations.

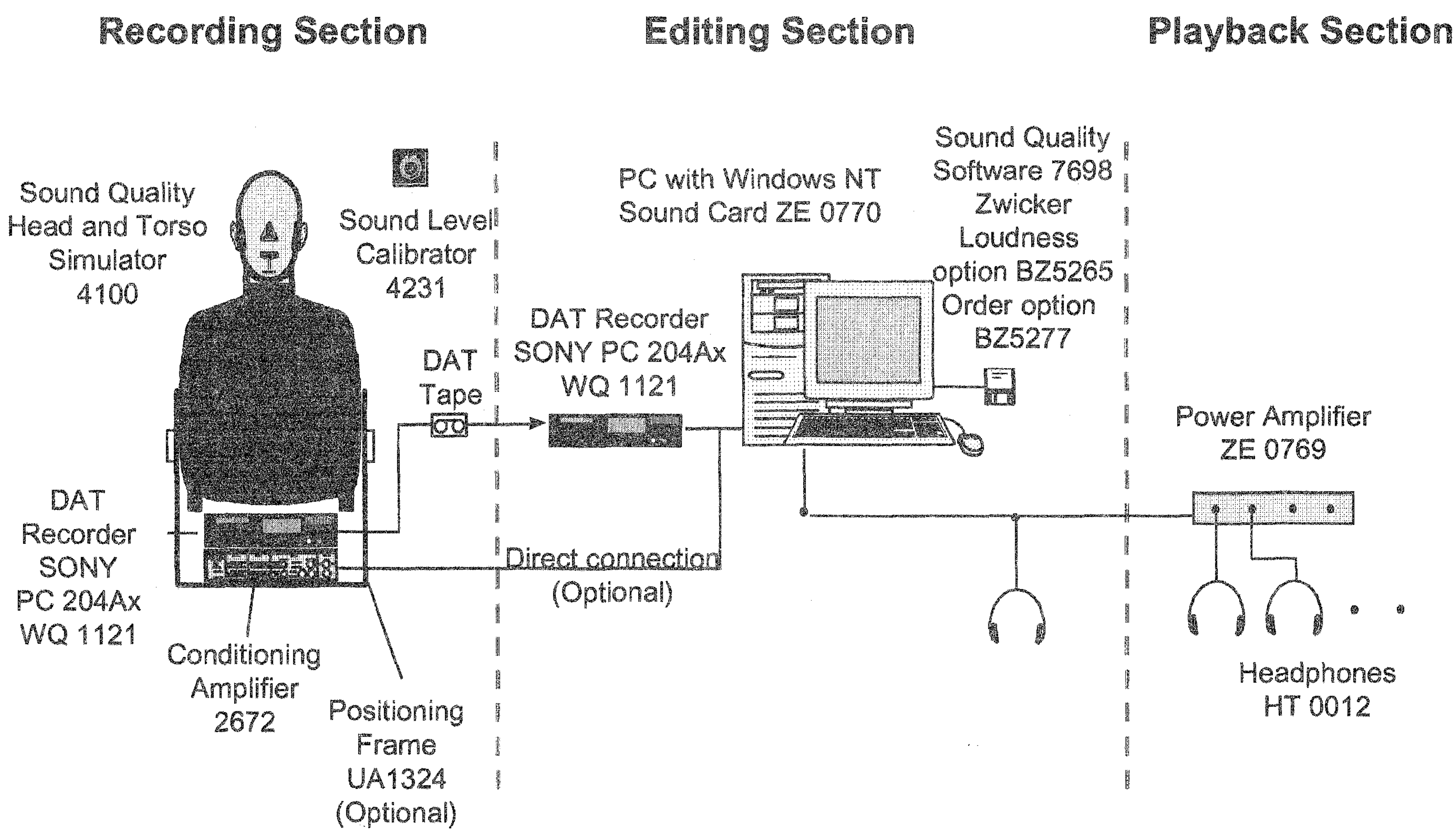
Since there is only one "stand in" - the Head and Torso - for all human beings, therefore HATS can only be an average. Deviations between a specific human being and the Head and Torso represent a potential source of errors.

Sound Quality Instrumentation

Sound Quality Instrumentation

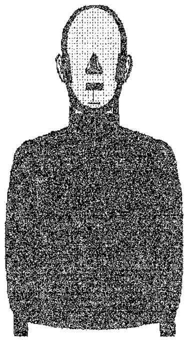


Sound Quality System Type 3801



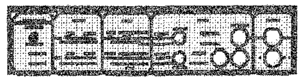
Recording Section

4100



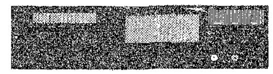
- Modeling of the average adult
- Binaural recording
- Falcon microphones.
- Easy microphone calibration
- Light and robust

2672



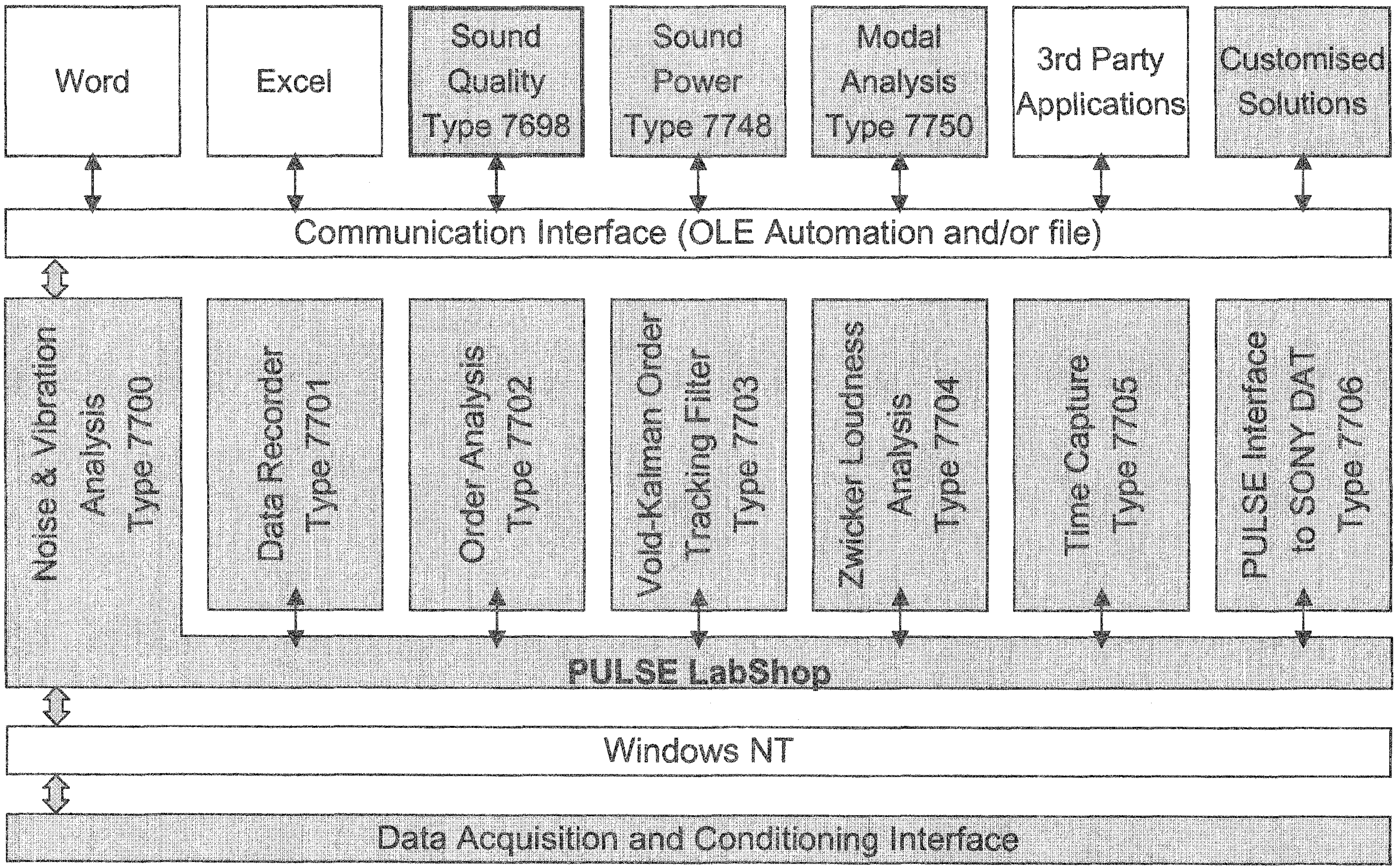
- Microphone power supply
- Highpass filtering
- -20 to + 50 dB gain
- Diffuse field filtering
- Charge Injection Calibration
- Multi-power supply

SONY DAT



- 4 channels x 20 kHz
- More than 80 dB Dynamic range
- Compact and lightweight
- Multi-power supply

Sound Quality Type 7698 - a PULSE Application Package

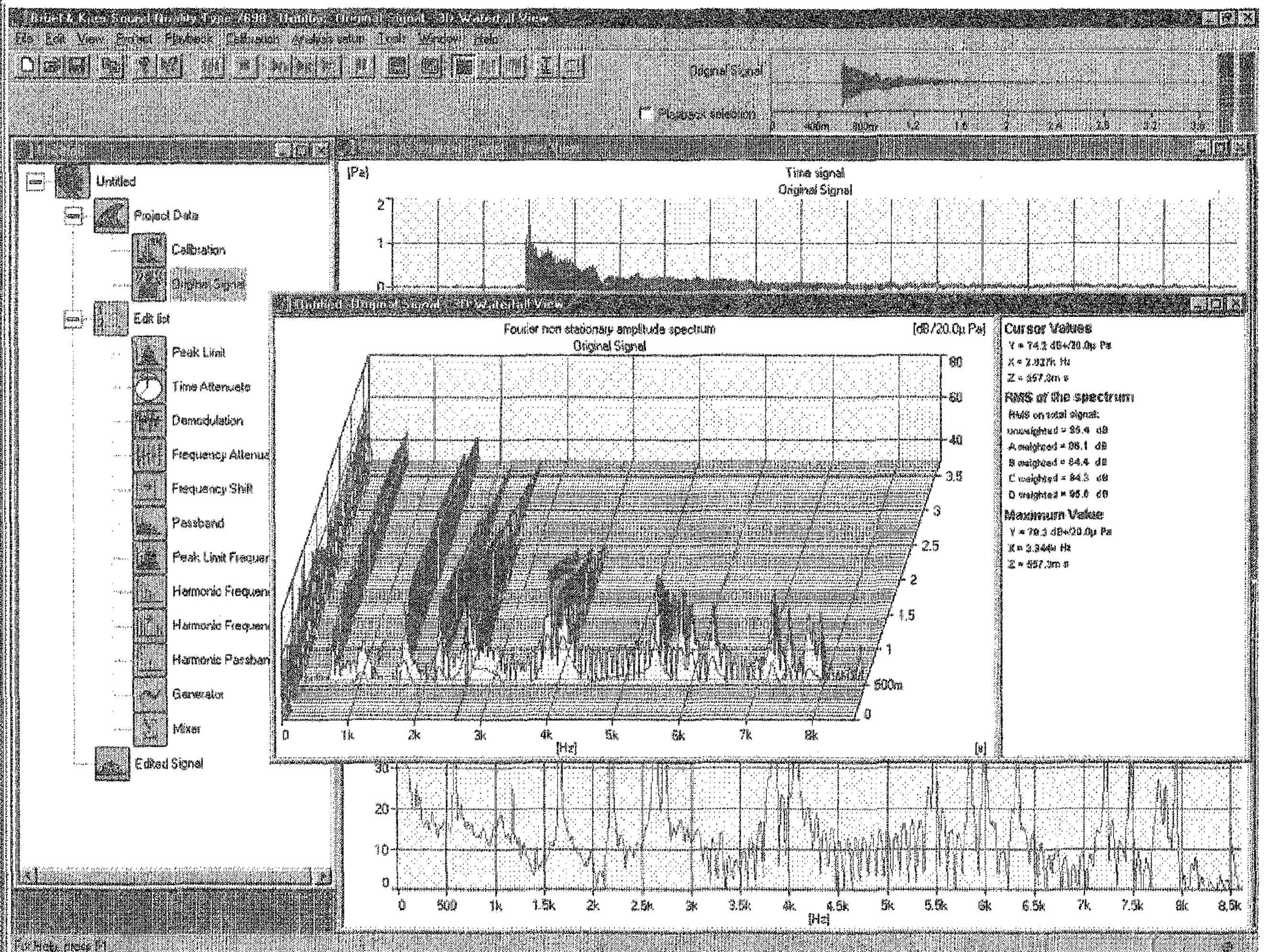


Features in 7698 version 2.0

- Native PC Data Processing
- FFT, CPB, Zwicker Loudness and Order Analysis
- Time, frequency and order domain edits
- OLE 2 programmable
- PULSE Support
- Imports .WAV and .UFF Files
- Digital I/O

Sound Quality software Type 7698

- Analysis of product sound
- Editing recorded sounds to simulate product improvement
- Determination of sound quality parameters:
  - loudness
  - sharpness
  - fluctuation strength
  - roughness
  - related parameters
- Preparing listening tests for product evaluation



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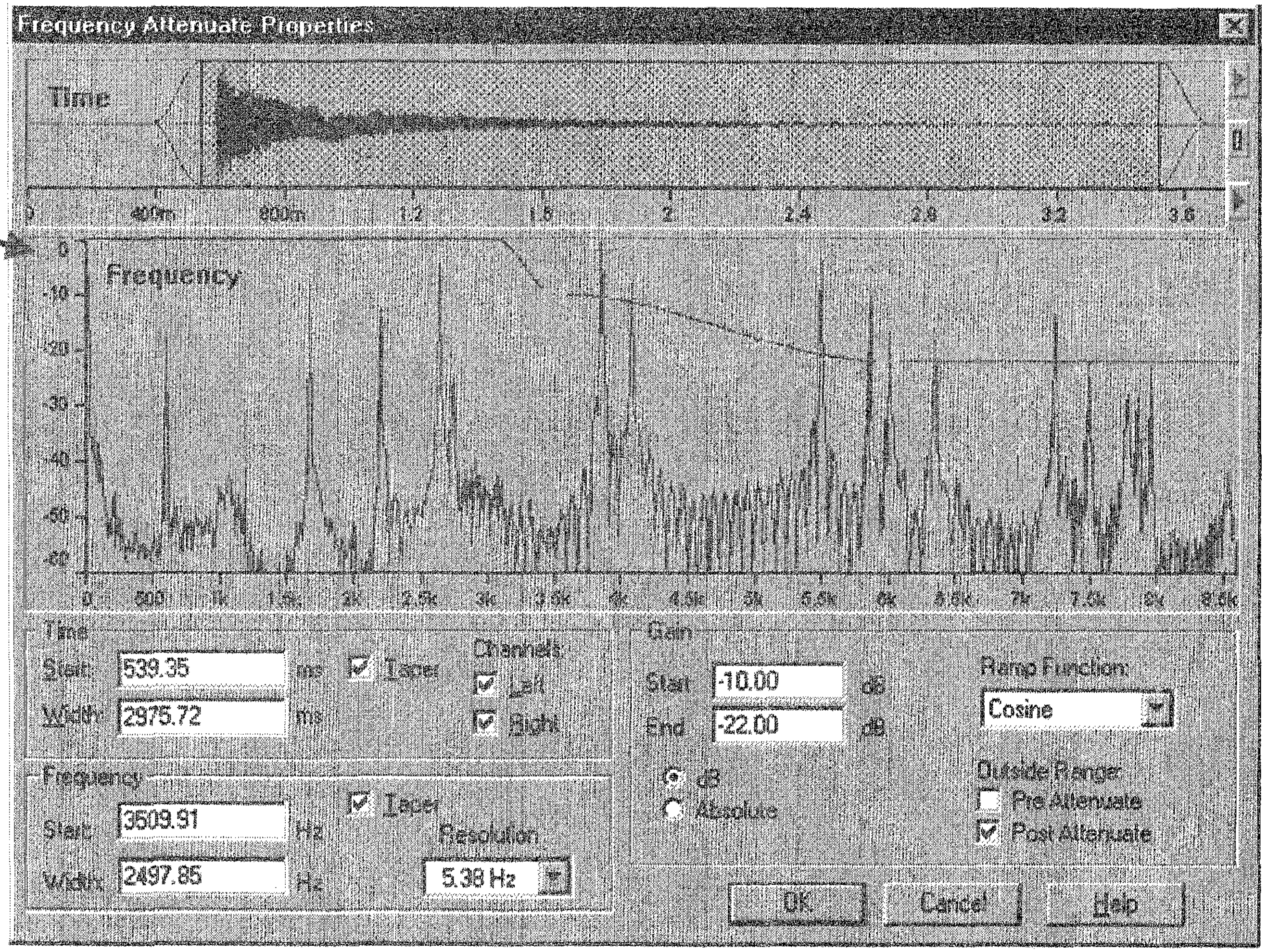
Editing tools

- Peak Limit (time)
- Time Attenuate
- Demodulation
- **Frequency attenuate**
- Frequency Shift
- Passband
- Peak Limit Frequency
- Harmonic Frequency Attenuate
- Harmonic Frequency Shift
- Harmonic Passband
- Generator
- Mixer
- Order Passband
- Order Attenuate



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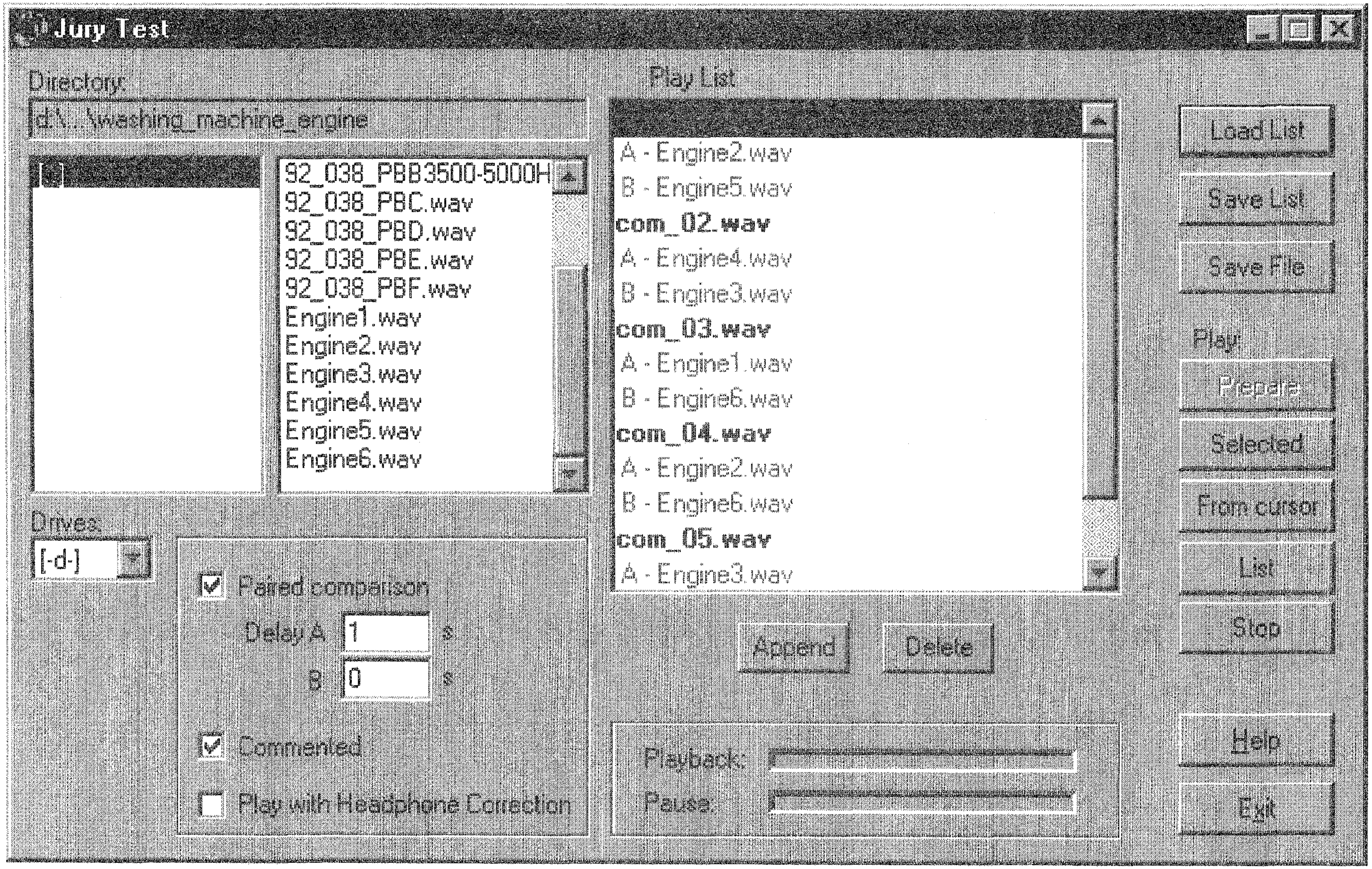
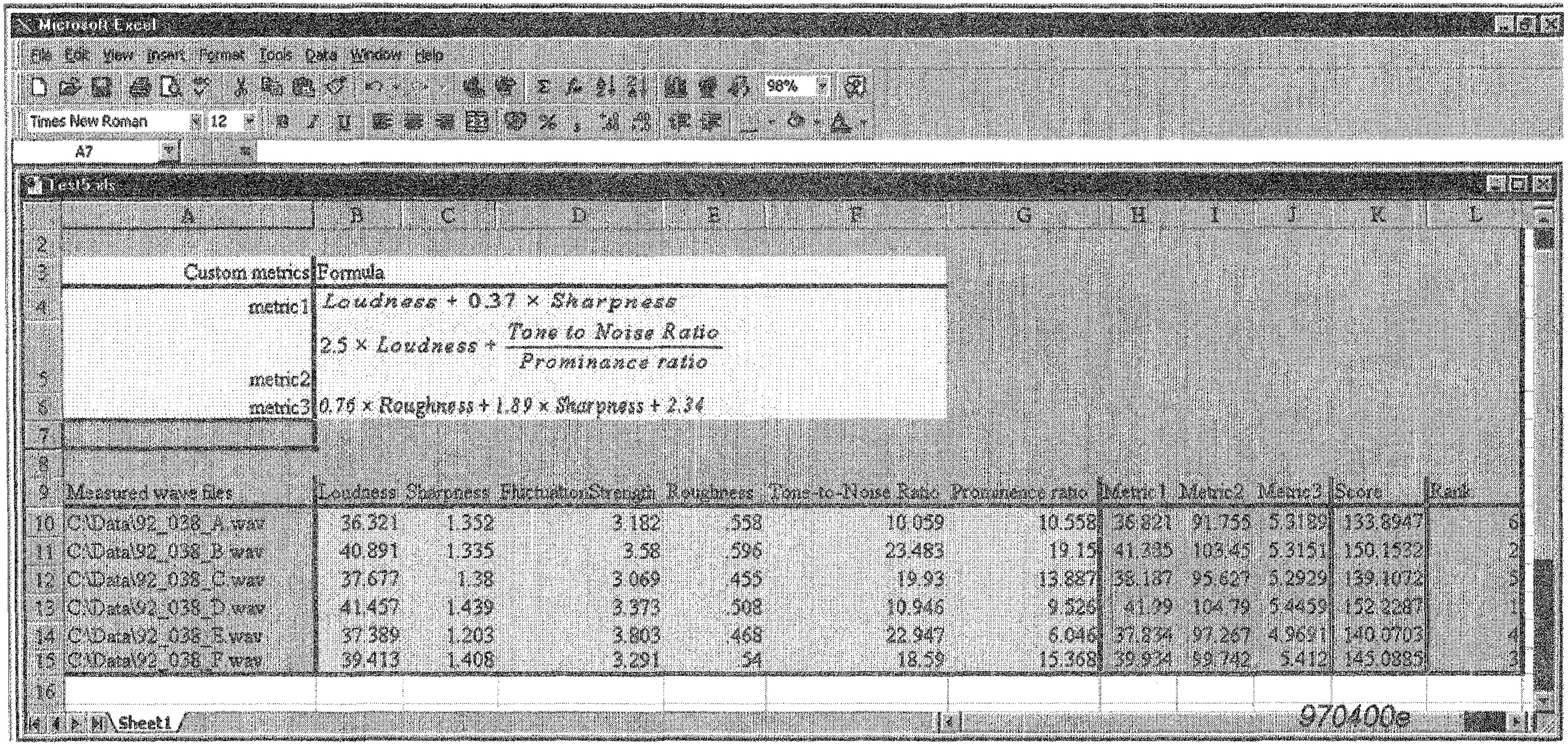
Sound Quality parameters

- Loudness
- Sharpness
- Statistical Loudness and Sharpness
  - Max.
  - Min.
  - Standard deviation
- Instantaneous Loudness
- Total and Specific Fluctuation strength
- Total and Specific Roughness
- Tone-to-noise ratio
- Prominence ratio
- RMS of time and frequency data

Subjective listening tests



OLE 2 Automation reporting

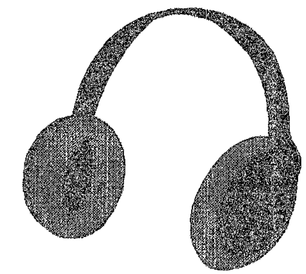


Custom metrics	Formula
metric1	$Loudness + 0.37 \times Sharpness$
metric2	$2.5 \times Loudness + \dfrac{Tone\ to\ Noise\ Ratio}{Prominance\ ratio}$
metric3	$0.76 \times Roughness + 1.89 \times Sharpness + 2.34$

Measured wave files	Loudness	Sharpness	FluctuationStrength	Roughness	Tone-to-Noise Ratio	Prominence ratio	Metric1	Metric2	Metric3	Score	Rank
C:\Data\92_038_A.wav	36.321	1.352	3.182	.558	10.059	10.558	36.821	91.755	5.3189	133.8947	6
C:\Data\92_038_B.wav	40.891	1.335	3.58	.596	23.483	19.15	41.385	103.45	5.3151	150.1532	2
C:\Data\92_038_C.wav	37.677	1.38	3.069	.455	19.93	13.887	38.187	95.627	5.2929	139.1072	5
C:\Data\92_038_D.wav	41.457	1.439	3.373	.508	10.946	9.526	41.99	104.79	5.4459	152.2287	1
C:\Data\92_038_E.wav	37.389	1.203	3.803	.468	22.947	6.046	37.834	97.267	4.9691	140.0703	4
C:\Data\92_038_F.wav	39.413	1.408	3.291	.54	18.59	15.368	39.934	99.742	5.412	145.0885	3

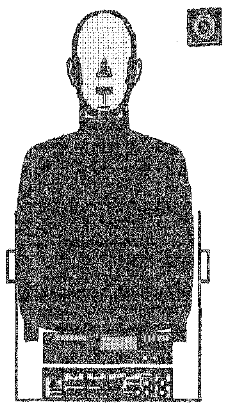
Headphones

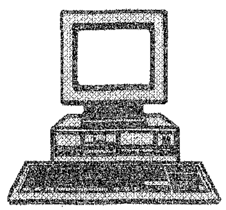
Selected Sennheiser HD 580

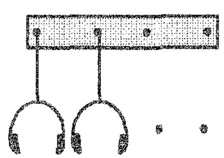


- Open, Dynamic type
  - Not sensitive to acoustic loading
  - High SPL with No distortion
- Selected for Brüel & Kjær Sound Quality
  - No individual calibration needed
  - Headphones can be interchanged without recalibration
- Headphone correction controlled by software
  - No individually dedicated hardware
- Subjectively evaluated to
  - Optimal quality of sound
  - Good wearing comfort
- Optimal choice on performance and price

Conclusion







- Full Sound Quality instrumentation
- Highest acoustic standard
- Extensive and accurate calibration
- Modular system
- Industrial standard interfaces
- PULSE integration
- Full Brüel & Kjær System support
- Maintenance contract