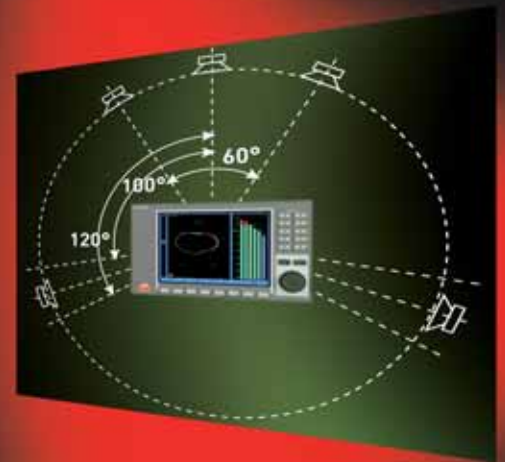
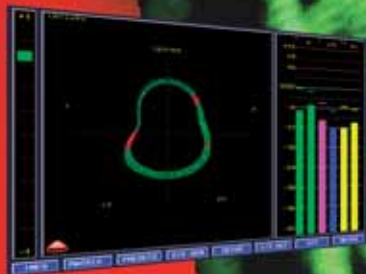
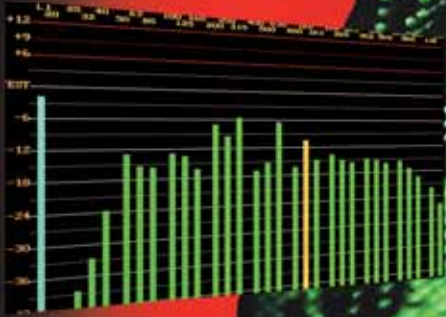


# Audio Levels and Readings



DK - Technologies

"solutions in Audio and Video"



# Welcome



**Congratulations on  
your purchase of a  
DK-Technologies' Audio Meter**

You are now the owner of one of the most sophisticated audio metering devices available today. The Audio Meters from DK-Technologies offer a wealth of features which – once you have used them in your audio engineering work – you'll never want to be without again.

This booklet is an addition to the regular User's Manual supplied with your Audio Meter. It offers a basic insight into the world of audio levels and metering and is recommended as a quick reference or as a quick audio 'brush-up'. If you want to

delve further into the world of audio metering, the book Audio Metering by Eddy Bøgh Brixen (of which this booklet is an extract) is available from DK-Technologies or from one of our international distributors.

Good audio requires technological as well as musical skills. To turn these into perfect audio you will also need the best audio metering – and we trust that your new Audio Meter will prove to be just that! And, if this booklet can help you a little bit on the way towards a clearer understanding of audio levels and readings, it has served its purpose.

# Audio Levels and Readings

## Why do we measure audio levels?

How hard can you drive the audio channel – and how loud is it? These are the eternal questions to everyone dealing with the practical work of audio recording, transmission, and sound reinforcement. In this case a measuring device - the meter - can be of great help, if you know how it works, and if you know how to use it.

These few pages will provide you with some basic knowledge concerning the use of meters and how to get the right information from the readings.

## What is an audio signal?

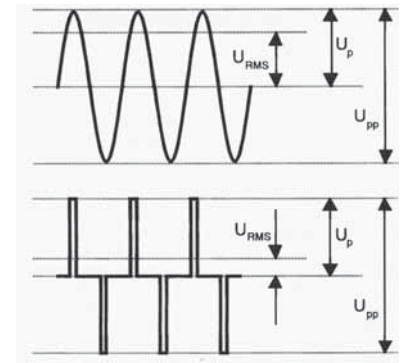
When present in the air the audio signal – or the sound – is a modulation of the static pressure. The air is vibrating. If the level of this vibration is above a certain threshold and within a specific frequency range, then the signal can be audible and we can therefore simply call it an audio signal. By the use of a transducer – a microphone

– this acoustic information can be converted to an electric signal, i.e. voltage, current, or electric power. Mostly we refer to the voltage. Another conversion of the audio signal can be carried out, if it is stored as magnetic information on a hard disk or a magnetic tape or as optical information on a CD/DVD or a film. At the end of the chain we can use another transducer, i.e. a loudspeaker or a headphone, and turn this electric/magnetic/optical information back to audible sound.

## How do we measure level?

When the volume of an audio signal has to be determined, it is very important that there is an agreement on how this is done. If not, one might end up with numbers and figures that are not comparable. When the audio signal exists in its electrical form, it can be described by the amount of voltage or current. Or the signal can be described by the energy contained within the signal, i.e. the power delivered into a given load in a given time

interval. With reference to the waveform as it can be seen on the oscilloscope, we can describe the definitions used.



The peak-to-peak value ( $U_{PP}$ ) is the absolute maximum deviation (plus and minus). The peak value is measured from the zero line to the maximum amplitude at either side. [Do remember that the peak we are talking about here is not the same peak as the one we use in the term “peak-programme-meter”. We will come back to that.]

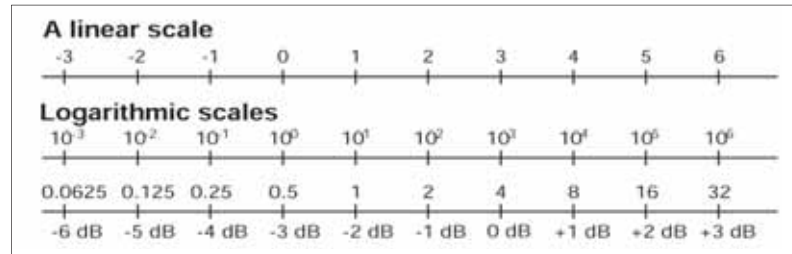
The expression we use the most is the “rms value”. The term rms is short for “root mean square”. It is a mathematical way of expressing the energy contained within the signal.

It makes DC and AC directly comparable if the AC is expressed by its rms value. To put it in another way: If we have a 12 volt electrical light bulb, then 12 volt DC and 12 volt AC will make it produce the same light intensity, as long as we are talking about the rms-value of the AC.

Having a sine wave the rms value is 0.707 times its peak value. If the waveform is a square wave, then the rms value equals the peak value. This tells us that two signals with the same peak value may have different rms values depending on the waveform – or two signals with the same rms value may have different peak values.

### Why are we using the dB scale?

The way we perceive sound is to some extent logarithmic both in pitch and loudness. In music we use frequency intervals like the octave. Starting at a specific frequency, each octave higher is the double of the frequency. By and large these intervals sound equal to the ear. The amount of



hertz is doubled in each interval, but the relation is constant. In this case the factor is two.

Talking about audio levels the same phenomenon exists. To perceive equal steps in level – up or down – the relation between each two steps has to be constant. For instance, if an audio signal of 1 volt is fed into a loudspeaker, we will hear a certain level. Supplying 2 volts will make it sound louder. To go one step further and for this step to be perceived equal to the first one we must supply not 3 volts but 4 volts. Then 8 volts, 16 volts and so on (if the speaker is not blasted by then).

The dB-scale expresses the relation between two values in a logarithmic manner. Each unit on the dB-scale is to some

extent perceived as being steps of equal size. That is why most level meters are using the dB-scale or scales strongly related to it.

To make the dB-scale absolute we must define a reference level. When the reading on a dB-scale is “0”, it does not mean that we have no signal. It means that the level of the signal we are measuring right now is exactly that of the reference signal. A positive value (+xx dB) means that the level is above the reference and a negative value (-xx dB) tells us that the actual level of the audio signal is below the reference level.

The “0 dB” reference is often indicated by the use of an additional letter:

# Audio Levels and Readings

**dBm:** The reference is 1 mW into a 600 ohm load.

**dBu:** The reference is 0.775 volt, corresponding to the voltage across a load of 600 ohm when 1 mW is delivered into it.

**dBV:** The reference is 1 volt

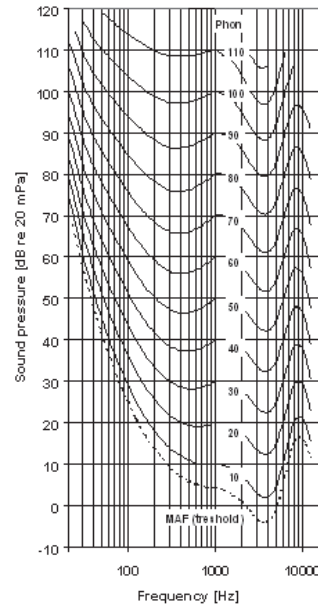
**dBFS:** The reference is full scale deflection and the signal level is defined individually. In general this is used with digital audio.

## Loudness

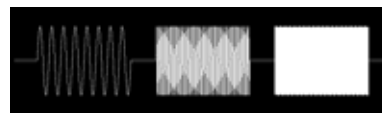
It must be noted that the dB scale is not telling the full story of how humans perceive audio levels. The duration of the signal is also of importance. For instance a tone with a duration of 10 ms does not sound as loud as a tone with duration of 1 second having the same level.

Concerning frequency, low frequencies are not perceived as loud as frequencies in the middle of the frequency range.

This phenomenon exists especially at low levels.



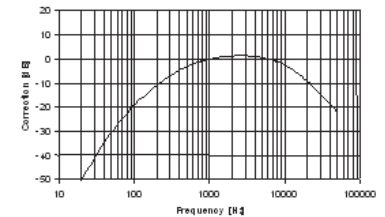
*The equal loudness curves.*



*These three sine waves (40 Hz, 400 Hz, and 4 kHz) have the same amplitude and the same duration. However, the loudness of these tones is very different.*

The nonlinear frequency response of the human ear has led to a

number of weighting filters, that are used for the measurement of acoustical noise (for instance A-weighting) and electrical noise in the equipment (CCIR/ITU weighting).



## A-weighting

*The A-weighting curve that to some degree corrects the measurement of acoustical noise as perceived by the human ear.*

Furthermore, the direction of the sound has an influence on the perceived level and spectrum. This can have an influence on the production of surround sound.

Also part of the assessment of a signal is the fact that the perceived loudness will fall during extended exposure due to the effects of fatigue. Finally the

perceived level is influenced by a “like/dislike” of the program, a totally subjective factor that is very difficult to calculate.

If we want to measure the level of loudness, many factors therefore have to be taken into consideration. The measuring device ideally would perform like the average perception of a sound signal by a group of people with normal hearing. That is not as easy as the simple determination of the rms value of a voltage.

The only way to provide information for this purpose is by carrying out a lot of experiments. From the results of these experiments measuring algorithms can be defined and implemented in the circuitry designed for the loudness metering.

The basic work on this was done by the psycho acoustician E. Zwicker. His model for calculation loudness has been a standard for years. However, his way of doing it is predomi-

nantly related to the perception of noise. International standard bodies like the ITU as well as many private companies have undertaken much work on how to measure the loudness of program material for cinema, broadcast, etc. This has led to – unfortunately – a number of slightly different methods. Users therefore have to consider the algorithms/standards they want to implement in their production. Is it loudness or rather annoyance you want to measure? Is it used for music or speech – or both? Is it for mono, stereo, or multichannel programs? Is it for broadcast or podcast?

### **What is a meter doing?**

The meter is a measuring device. It can perform relevant measurements with regard to the equipment or transmission lines, we are using for audio. Some meters available are performing the measurement of analogue signals only. Others can perform measurements of digital audio signals. Some meters are able to do both. In order

to obtain stable, reliable, and comparable readings the electric characteristics are normally specified in a standard.

Concerning the standard level meters, these characteristics comply with the rms-process, time constants or integration time, fall time, level range, reference level, scaling, and so forth. This means that we must know which standards are met by the specifications of the actual meter before we know what the reading is actually telling us. The standards may either be originated by national or international standardisation organisations. Or they may be factory or so-called de facto standards.

Years ago a meter was always built as a mechanical device with a needle attached to a moving coil placed in a magnetic field. Later came the LED-ladder type of meters. Today, all kinds of displays are used as for example LCD, electroluminescent, or video monitoring screens.

# Audio Levels and Readings

## What is integration time?

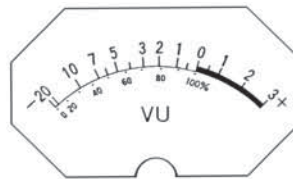
One of the most important characteristics to be defined is the integration time. To express this in a non-mathematical way, it is the time it takes the meter to reach a reference level when a constant signal is fed into it. Normally this can be checked by the use of a tone burst with a level equal to the reference of the meter. The different standards have their own definitions of how the reading is reached in the right way.

To be able to see what the level was, when it was there, most of the faster meters (those with short integration time) also define a relatively slow fall back time. But this has nothing specific to do with the integration time itself. Now let us take a closer look at the different meters in use.

## VU

VU (= Volume Unit). This instrument was originally invented by the Bell Lab and by American broadcasters in collaboration.

It became a standard in 1939. The purpose of the instrument was to provide the users with information, when using telephone lines for transmission and distribution. Also it was an attempt to give a visual information corresponding to the perceived level – the volume - of the sound. For this reason – as well as for practical mechanical and especially ballistic reasons – the instrument is acting rather slowly. It is more like a moving average. By definition the meter will reach full deflection (or 99% of full deflection to be more precise), if the signal has a duration of 300 ms or more.



The scale covers the range -20 VU to +3 VU. 0 VU is reached at the voltage 1.23 V (across a load of 600 ohm), corresponding to +4 dBm.

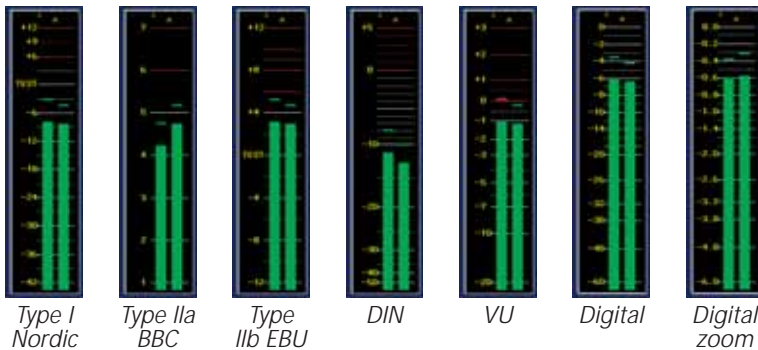
As the VU-meter moves relatively slowly, impulses of short

duration are not shown. As a matter of fact the peaks of the program material reach levels 6-12 dB above the actual deflection. That has to be taken into account when this meter is used. Older versions of the VU-meter implemented an amplifier in order to give the meter a “lead” when program material containing transients was to be recorded. In the meantime this option has been forgotten. In the practical use of VU meters today, they are sometimes only used for level calibration. The “0 VU” is used as a test signal, and the meter is calibrated to have a headroom exceeding the maximum reading of the scale by 5 to 6 dB.

## Peak Program Meter (PPM)

This instrument is telling us something about, how hard we are driving the channel. That is because the integration time is relatively short, 5 or 10 ms as a standard. Program material will normally not contain peaks that exceed the reading by more than 3 dB.





The major difference between different PPM standards is the definition of the scales. The definition of integration time and fall time only differs slightly. Let us take a look at the different versions.

### IEC 268-10 type I

The type I instrument is used in the Nordic countries. The integration time is 5 ms. This means that the meter shall reach an indication of 2 dB below the reference level in 5 ms measured from the time the signal was applied. The fall time or return time is 20 dB/ 2 s. The reference voltage is 1.55 volt (+6 dBu).

The scale used, also known as

the Nordic scale, is calibrated in dBu.

### IEC 268-10 type IIa

Type IIa and type IIb PPM instruments have an integration time of 10 ms. The type IIa is also known as the BBC or the UK PPM. The scale has the numbers from 1 to 7. Between each partition is 4 dB. The reference level is 1,94 volts, which is reached at "6".

### IEC 268-10 type IIb

This PPM is specially designed for transmission lines within the European Broadcasting Union (EBU). The scale ranges from -12 to +12. Reference level is reached at 2.18 volt (+9 dBu) which equals +9 dB on the scale.

### DIN – Pflichtenheft 3/6

The dynamic response of this PPM instrument is designed to reach -3 dB in 3 ms. And -2 dB in 5 ms. The time of return is 1.5 s. from 0 dB to -20 dB. The scale covers at least a range from -50 dB to + 5dB on the scale.

### ABC-meter

The meter and the scale are identical to the IEC type I.

### NBC-meter

This meter is working like the VU-meter, however the scale ranges from -60 to 0 VU.

### Manufacturer defined scales

Manufacturers or broadcasters may define their own standards. The scales are often related to specific products, specific set ups, etc. The scales may refer to internal setting of the product.

### The measurement of digital signals

Measuring audio signals in the digital domain is not that easy. But one thing is sure: 0 dBFS

# Audio Levels and Readings

is the maximum level. When all bits are activated, that is when we have reached the maximum. But still one has to find out, how the levels are defined. For instance: When measuring a sine wave and a square wave with the same amplitude (even at 0 dBFS) then the rms-value of the square wave is 3 dB higher than that of the sine wave!

In the early days of analogue-to-digital converting a nasty problem occurred when the input signal exceeded full scale. The result would be random digital values followed by heavy distortion. Today, all converters just stay at the highest value, resulting in clipping and thus third harmonic distortion. This is not that bad, if only the harmonic components are beyond the audible frequency range. Many (most) modern pop recordings are distorted in this way even if the dynamic range is less than 10 dB!

If using semi-professional digital audio equipment the level indicator may have a very poor design. In this case it is a good idea to use an auxiliary professional level meter. The digital meter may have the ability to display some of the additional information that is carried along with the signal in the interface connection: Sampling frequency, timecode, consumer/pro mode, emphasis, etc.

## **How are analogue levels related to digital scales?**

When using an analogue-to-digital converter it is essential to know, how the level of the analogue signal relates to the digital resolution. Basically there are two standards describing this conversion: one American (SMPTE RP155) and one European (EBU R68). Most American manufactures fulfil the SMPTE standard and most European manufacturers meet the requirements of the EBU standard.

The difference concerning the analogue level is 6 dB!

## **DMU I and II**

These scales are related to digital audio.

DMU I - 60 dBFS to 0 dBFS  
DMU II (Digital zoom): -6 dBFS to 0 dBFS.

## **Dynamic scales**

It is a fact that the dynamic range can vary in different types of programme material. Pop music can be very compressed and thus have a very limited dynamic range. Classical music and films mixed for the cinema have a large dynamic range.

## **Dialnorm**

Dialnorm is an abbreviation of dialogue normalisation. It has been introduced by Dolby to be used with programme material for television. According to this procedure it lets the programme material be recorded in an optimum manner, but the level at which the dialogue is situated is specified. All the audio can sub-

sequently be shifted in level so that the volume of the dialogue is uniform from programme to programme.

The standard states that the speech level is determined as A-weighted Leq.

### **K-system**

This has been introduced by Bob Katz, an American whose experience includes many years of recording and mastering in the American music industry. The system builds upon three scales, namely K-20 with 20 dB headroom over 0 dB, K-14 with 14 dB and K-12 with 12 dB headroom.

On the scales, the colour green is used below 0 dB, yellow between 0 and +4 dB and red for over +4 dB. Each scale can be used with three different time/frequency weightings, named RMS, Leq(a) and Zwicker.

Thus K-X\*/RMS is used with a frequency response of 20 Hz - 20 kHz  $\pm$ 0.1 dB.

K-X\*/Leq(a) uses A-weighting (IEC A) and an integration time of 3 seconds.

K-X\*/Zwicker uses, as the name suggests, Zwicker's model for loudness.

For calibrated playback, pink noise at a signal level of 0 dB on the scale will correspond to 83 dB(C) in the listening position.

### **Loudness meters**

As discussed earlier it is quite difficult to make objective measurements of subjective parameters. This is a reason for the existence of different approaches to the design of loudness reading meters.

The loudness calculated in accordance with the original Zwicker method uses the phon-scale. (At 1 kHz the sound pressure level expressed in dB re 20 mPa equals the loudness level in phon).

### **Dorroughs**

The meter produced by Dor-

roughs has a scale that show the "relative loudness to peak modulation". The scale ranges from - 36 dBm to +3 dBm. The loudness is calculated by a relative simple average function.

### **Leq(m)**

The Leq(m) is introduced by Dolby. It expresses the energy based equivalent sound level (Leq). The "m" stands for "movie". It is a method that was developed to measure the sound level of movies, particularly trailers and commercials. It was initially claimed to measure loudness. However, it is now rather recognised as a methodology for measuring of the annoyance.

The measurement of the film's sound is made electrically since it is presumed that the film will be played in a cinema with a calibrated sound system. The sound from each track is frequency-weighted and filtered with CCIR 468 (level-corrected to 0 dB at 2 kHz). All signals are detected and then summed on an energy basis to a single va-



presented by different researchers and manufacturers. So other different solutions to the same problem are found in the marketplace.

### The Phase Meter

In production, transmission and storage of stereo signals one has to be aware of the phase relation between the channels. Two instruments are very important with regard to the monitoring of phase.

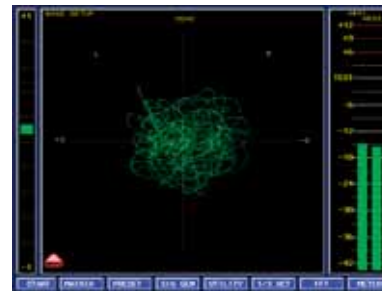
The phase meter involves a scale that ranges from +1 to -1. Actually this scale is expressing a cosine function. What is shown is the cosine to the phase angle between the two channels. If the phase is 0 degrees, the resulting value is "1". If the phase angle is 180 degrees the resulting value is "-1". 90 degrees of phase shift (or only one channel connected) gives a "0" on the scale. To stabilise the reading and to emphasise the phase readings at lower frequencies, the phase meter is quite slow. During recording, the readings of the

phase meter should normally be between 0 and +1.

### Audio Vector Oscilloscope

The audio vector oscilloscope is another instrument that provides more detailed information concerning the interchannel phase relation. Basically it is an oscilloscope, where the right channel deflects the beam in the X-direction and the left channel deflects the beam in the Y-direction. The system is arranged in a way that provides a vertical line, if the channels are in phase. A horizontal line is displayed, if the signals are oppositely phased. If the phase relation is 90 degrees and the signals have equal level, a circle is shown on the screen. When you have learnt to read this instrument, it provides you with a lot of information concerning all kinds of tricky problems you are not always aware of. This means that you can see it, even before you hear it: Balancing effects, limiting effects, delay effects, polarity problems, hum, and much more. The figures explain

the readings of a meter that combines the displays of levels in two channels (right scales), a phase meter (left scale) and the audio vector oscilloscope in the middle.

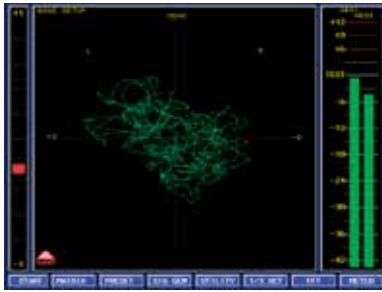


*This is a perfect stereo signal. The phase meter is just above zero, so the signal is mono compatible.*

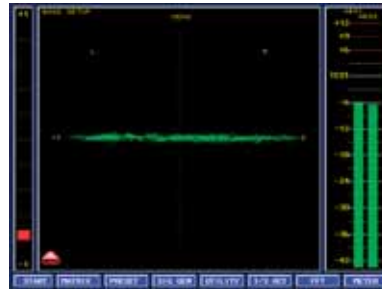


*Both channels contain the same signal (same level and same phase). This reading tells us that this is pure mono.*

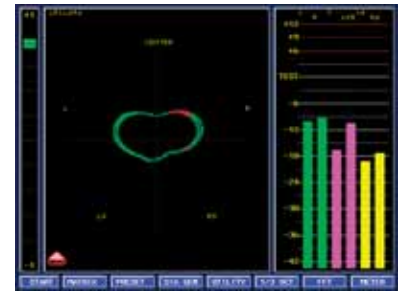
# Audio Levels and Readings



*This signal is stereo, but it is no longer mono compatible.*



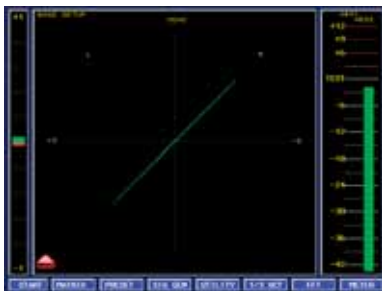
*This signal is in stereo (speak) but oppositely phased (180 degree).*



*JellyFish Display*



*This is mono in the left channel. There is no signal in the right channel.*



*This is mono in the right channel.*

## JellyFish Vector Display

A special version of the audio vector oscilloscope is able to generate a special figure, called the "JellyFish". It was developed to ease monitoring levels in the production of multi-channel surround sound. Further, a special option has been developed to display (by changing colour), if neighbouring channels exceed a phase angle of 90 degrees.

Basically the figure presented is a circle. Each part of the circumference is allocated to a specific channel. If the level in one channel is raised, the corresponding part of the circle will expand.

## What are test levels meant for?

First of all, test levels are standardized for the line up of transmission lines. This is one reason, why the test levels seldom are at the top of the scale. In the early days of radio transmission the purpose of the test signals was to heat the transmitter as done by "gewöhnlicher Tanzmusik" (usual dance music).

This is why test levels are normally found at least 9 dB below the permitted maximum. Later more test tones at different levels have been introduced for testing dynamic range, frequency response and so on.

### **Where to connect the meter?**

You must realise that a level meter is a calibrated tool for audio measurements. That is why you should never “read-just” a level meter once it has been calibrated to the standard commonly used at your place. You should also notice that the level meters always operate at line level, never at microphone level or the like. However, many level meters offer a “+20 dB” option, so that lower level non-pro equipment can be monitored. In the audio chain there are specific points where to connect the meter, either permanently or as required:

- At the outputs of the mixing desk, master left/right (and centre/surround), groups, aux. sends and busses.
- At the input of the A-to-D converter, to ensure the optimisation of the dynamic range.
- At the output of the different sound sources like the tape machines, tie lines, external production units (especially in broadcasting).
- In some installations care must be taken, if emphasis is added to the signal for instance in connection with FM or satellite transmission. It may be important to measure the levels including the emphasis as the signal has an increased gain of 5 to 10 dB at higher frequencies.
- If working in the field of PA or SR it is important to run each piece of equipment in the total chain at the best dynamic range possible. The meter can be inserted to check all points from line inputs to the power amplifiers.



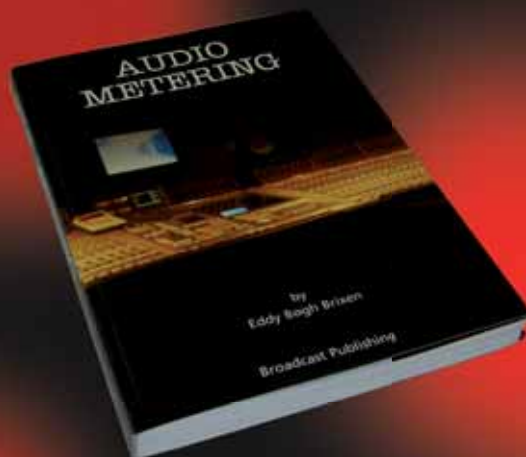
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Eddy Bøgh Brixen is an independent audio consultant with EBB-consult. He is the author of several books and articles on audio subjects, and the driving force behind much of the audio education in Denmark. He is active in the The Audio Engineering Society.

### **Audio Levels and Readings**

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More details can be found in the book "Audio Metering", which can be purchased in our webshop at [www.dk-technologies.com](http://www.dk-technologies.com)



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