

Audio Levels and Readings

by Eddy Bogh Brixen



*SSL Mixing console and no less than 3
Master Stereo Displays in the large
OB-van from TEAM JELBE, Sweden.
Photo: Team Jelbe, Sweden*



Welcome

Congratulations on your purchase of a DK-Audio Master Stereo Display

You are now the owner of one of the most sophisticated audio metering devices available today. The Master Stereo Displays from DK-Audio 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 Master Stereo Display. 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 a new book (of which this book-

let is an extract) with the same title is available from DK-Audio or one of our international distributors. To order the book just fill in the enclosed coupon and send or fax it to us.

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 Master Stereo Display 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 will have 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 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?

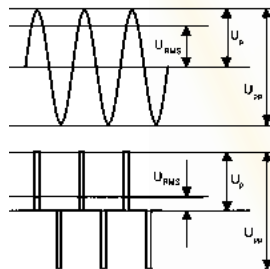
When present in the air the audio signal – or the sound – is a modulation of the static pressure. The air is vibrating. If this vibration – the sound - is above a certain threshold and within a specific frequency range, then the signal can be audible and we can therefore simply call it an audio signal. By the use of a transducer – a microphone – this acoustic information can be converted to electricity, 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 size 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 or a given time span. With reference to the amplitude as it can be seen at 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-program-meter”. We shall 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 amount of light – as long as we are talking about the rms-value of the AC.

Having a sine wave the rms-value is 0.707 times the 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. And so what? In music we use 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 cyc-



les per second 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.

It must be noted that the dB-scale is not telling the full story of how humans perceive audio

level. The duration of the signal is also of importance. For instance a tone with duration of 10 ms does not sound as loud as a tone with duration of 1 second. Concerning frequency, low frequencies are not perceived as loud as frequencies in the middle of the frequency range.

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

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.

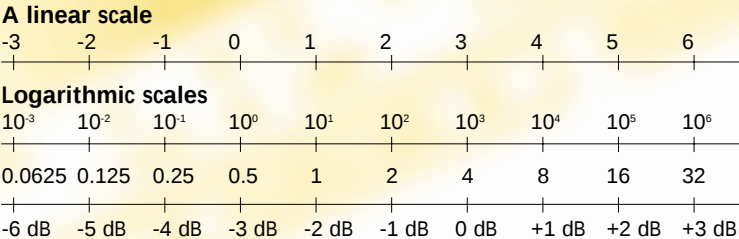
dBV: The reference is 1 volt
dBFS: The reference is full scale deflection and the signal level is defined individually.

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. Most 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. 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.

What is integration time?

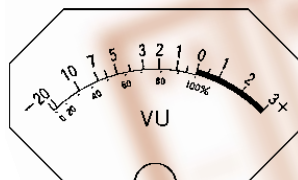
One of the most important characteristics to be defined is the integration time. To express this in a non-mathematical way, it is the time it takes the meter to reach a reference level when a constant signal is fed into it. Normally this can be checked by the use of a toneburst with a level equal to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

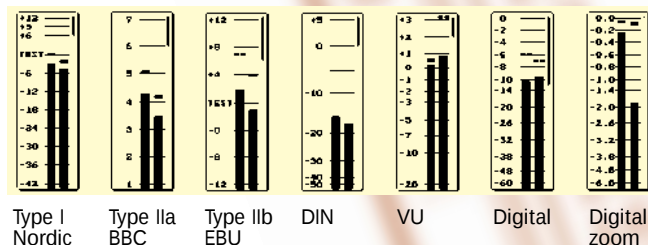
taken into account when this meter is used. Older versions of the VU-meter implemented an amplifier in order to give the meter a “lead” when program material containing transients was to be recorded. In the meantime this option has been forgotten.

In the practical use of VU-meters today, they are some times only used for level calibration. Using the “0 VU” for a test signal, and having a headroom exceeding the max 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 with 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 Germany and in the Nordic countries. The integration time is 5 ms., meaning that the meter shall reach an indication 2 dB below the reference level in 5 ms. from the time the signal was added. The fall time or return time is 20 dB/ 2 s. The reference voltage is 1.55 volt (+6 dBu).

The scale used, the so-called Nordic scale, is calibrated in dBu.

IEC 268-10 type IIa

Both type II 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.

Loudness meters

Different manufactures have introduced so-called "loudness meters". Loudness expresses how the human hearing perceives the level of sound when both duration of the sound and the frequency of the sound are taken into account. A very precise measuring method has been described and is known as the "Zwicker-method", named after a famous psycho-acoustician. Other methods are not nearly that precise and may therefore only be regarded as a kind of approach to the measurement of real loudness.

The measurement of digital signals

Measuring audio signals in the digital domain is not that easy. But one thing is sure: 0 dBFS is the max level. When all bits are activated, that is when we have reached the maximum. But still one has to define how levels are defined. For instance: When measuring a sine wave and a square wave with the same amplitude (at max level) 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 the all-bit-activated situation. The result would be random digital values followed by heavy distortion. Today most converters just stays 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, time code, consumer-/pro- mode, and emphasis.

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! The table on next page shows the relationship between these two standards.

	EBU R68			SMPTE RP155			
	dBFS	DBu	Volt	dBFS	dBu	Volt	
Maximum level	0	18	6.16	0	24	12.28	Maximum level
	-1	17	5.49	-1	23	10.95	
	-2	16	4.89	-2	22	9.76	
	-3	15	4.36	-3	21	8.70	
	-4	14	3.88	-4	20	7.75	
	-5	13	3.46	-5	19	6.91	
	-6	12	3.09	-6	18	6.16	
	-7	11	2.75	-7	17	5.49	
	-8	10	2.45	-8	16	4.89	
	-9	9	2.18	-9	15	4.36	
	-10	8	1.95	-10	14	3.88	
	-11	7	1.74	-11	13	3.46	
	-12	6	1.55	-12	12	3.09	
	-13	5	1.38	-13	11	2.75	
	-14	4	1.23	-14	10	2.45	
	-15	3	1.09	-15	9	2.18	
	-16	2	0.976	-16	8	1.95	
	-17	1	0.870	-17	7	1.74	
Test	-18	0	0.775	-18	6	1.55	0 VU
	-19	-1	0.691	-19	5	1.38	
	-20	-2	0.616	-20	4	1.23	
	-21	-3	0.549	-21	3	1.09	
	-22	-4	0.489	-22	2	0.976	
	-23	-5	0.436	-23	1	0.870	
	-24	-6	0.388	-24	0	0.775	
	-25	-7	0.346	-25	-1	0.691	
	-26	-8	0.309	-26	-2	0.616	
	-27	-9	0.275	-27	-3	0.549	
	-28	-10	0.245	-28	-4	0.489	
	-29	-11	0.218	-29	-5	0.436	
	-30	-12	0.195	-30	-6	0.388	
	-31	-13	0.174	-31	-7	0.346	
	-32	-14	0.155	-32	-8	0.309	
	-33	-15	0.138	-33	-9	0.275	
	-34	-16	0.123	-34	-10	0.245	
	-35	-17	0.109	-35	-11	0.218	
	-36	-18	0.098	-36	-12	0.195	
	-37	-19	0.087	-37	-13	0.174	
	-38	-20	0.078	-38	-14	0.155	
	-39	-21	0.069	-39	-15	0.138	
-40	-22	0.062	-40	-16	0.123		
-41	-23	0.055	-41	-17	0.109		
-42	-24	0.049	-42	-18	0.098		

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.

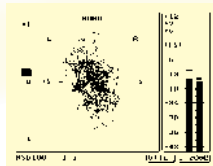
When recording the readings of the phase meter should normally be between 0 and +1.

Audio Vector Oscilloscope

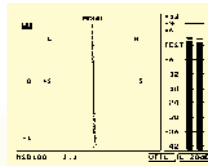
The audio vector oscilloscope is another instrument that provides more detailed information concerning the inter-channel phase relation. Basically it is an oscilloscope where the right channel deflects the beam in the X-direction and the left

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 and a horizontal line 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, meaning you can see it before you hear it: Balancing effects, limiting effects, delay effects, polarity problems, hum, and much more.

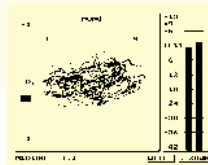
The figures explain the readings of a meter that combines the displays of levels in two channels (right scales), a phase meter (left scale) and the audio vector oscilloscope in the middle.



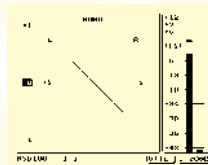
This is a perfect stereo signal. The phase meter is just above zero, so the signal is mono compatible.



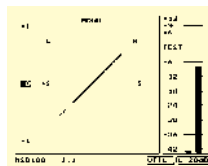
Both channels contain the same signal (same level and same phase). This reading tells us that this is pure mono.



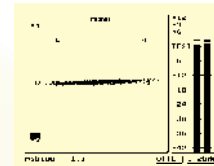
This signal is stereo, but it is no longer mono compatible.



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



This is mono in the right channel.



This is mono, but oppositely phased. (180 degrees).

“Jelly fish”

A special version of the audio vector oscilloscope is able to generate a special figure, called the “jelly fish”. It was developed to ease monitoring levels in the production of multi-channel surround sound. Besides that a special option has been developed to display (by changing colour) if neighbouring channels exceeds 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 signals are made for the line up of transmission lines. That is one reason why the test levels seldom are at the top of the scale. In the earlier days of radio transmission the purpose of the test signals was to heat the transmitter as much as done by "gewöhnlich tanzmusik" (common dance music). That is why test signals normally are 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 "re-adjust" a level meter once it has been calibrated to the standard commonly used at your place. Do notice that the level meters always operate at line level, never at microphone level or the like. However, many level meters offer an "+20 dB" option so lower level non-pro equipment can be monitored.

In the audio chain there are specific points where to connect the meter, either permanently or when needed:

- 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 is gained approximately 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.

*Audio Levels and Readings
Copyright 1999 by Eddy Bøgh
Brixen & DK-Audio*





About the Author:

Eddy Bøgh Brixen is an audio consultant with the Acoustics department of DELTA, which is an institute affiliated to the Danish Academy of Technical Sciences. 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 Chairman of the Danish Section of The Audio Engineering Society.

[*...see what
you hear!* **]**



DK-AUDIO • Marielundvej 37D • DK-2730 Herlev • Denmark

Phone: +45 44 85 02 55 • Fax: +45 44 85 02 50 • e-mail: info@dk-audio.com • www.dk-audio.com