

EBU R 128 – A new standard in audio and broadcast technology

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31.01.2013

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Declaration

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- Any direct quotation or source of ideas has been identified in the text by author, date, and page number(s) immediately after such an item, and full details are provided in a reference list at the end of the text.
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1 INTRODUCTION

Up to now level control guidelines for audio signals in public broadcasting were orientating on audio peak levels. That means a maximum level of -9 dBFS (Decibel Full Scale) was deemed to be the barrier for gaining audio signals.

In the last years the media industry has caused audio signals to increase more and more loudness without crossing the peak level trough using techniques like compression and limiting. As a consequence some kind of “war” broke loose. Producers of audio and visual media content were trying to push their productions to the limit to make them sound as loud as possible without crossing the audio peak level. This phenomenon is called the „loudness war“ among experts and consequences can be witnessed and heard in different areas.

With a new guideline set, called EBU R 128 by the European Broadcast Unit (EBU), a new measurement unit will be established.

The EBU has published a number of technical documents describing R128 and all the other recommendations and innovations that come with them, which can be downloaded from the EBU’s official website. My paper is based on these technical papers and on a lecture Florian Camerer held about R128 called “EBU R128 Introduction”.

“Have you ever wondered why commercials sound louder than your favourite TV shows? Or why you have to adjust the playback volume on your television when switching between channels?

The answer is that until recently, there was no standard way to measure the perceived loudness of sound recordings. Instead, audio productions were (and still are) normalized to peak levels, which do in no way determine how loud a signal is.”¹

¹ <https://auphonic.com/blog/2012/08/02/loudness-measurement-and-normalization-ebu-r128-calm-act/>
Accessed on 16.11.2012

„...the change of the levelling paradigm from peak normalization to loudness normalization. This change is vital because of a problem that has become a major source of irritation for television and radio audiences around the world – that of the jump in audio levels at the breaks within programs, between programs and between channels. Loudness normalization is the solution to counteract this problem.“²

2 DYNAMIC

Dynamic is very important in music and other audio content. Dynamic is the essence of every art form. Dynamic accentuates, connects and bonds single parts to a self-contained piece.

The following image shows the different stages of loudness (dynamic) in classical music. As you can see, there are quiet a lot of steps from very quiet (almost not hearable) to very loud. These are not even all of the dynamic steps in classical music, but they are the ones that are mostly used. All that dynamic, that has made music so alive and worth listening to for hundreds of years is being thrown away to accomplish a goal which means much more to a lot of people. => LOUDNESS



<i>fff</i>	as fortissimo as possible
<i>ff</i>	fortissimo (very loud)
<i>f</i>	forte (loud)
<i>mf</i>	mezzo forte (moderately loud)
<i>mp</i>	mezzo piano (moderately soft)
<i>p</i>	piano (soft)
<i>pp</i>	pianissimo (very soft)
<i>ppp</i>	as pianissimo as possible

Figure 1: Dynamic Symbols

² http://tech.ebu.ch/docs/techreview/trev_2010-Q3_loudness_Camerer.pdf Accessed on 16.11.2012

Music without dynamic is like a picture without contrast, Florian Camerer mentions in his presentation. As a picture needs contrast between light and dark, audio signals need a contrast between loud and quiet.

Dynamic in music or contrast in photography is an essential part of these art forms.

If you take away a photograph's contrast, all that is left is an empty shell.

I want to illustrate this thought by showing the following photographs.

In this image I reduced the amount of contrast to zero:

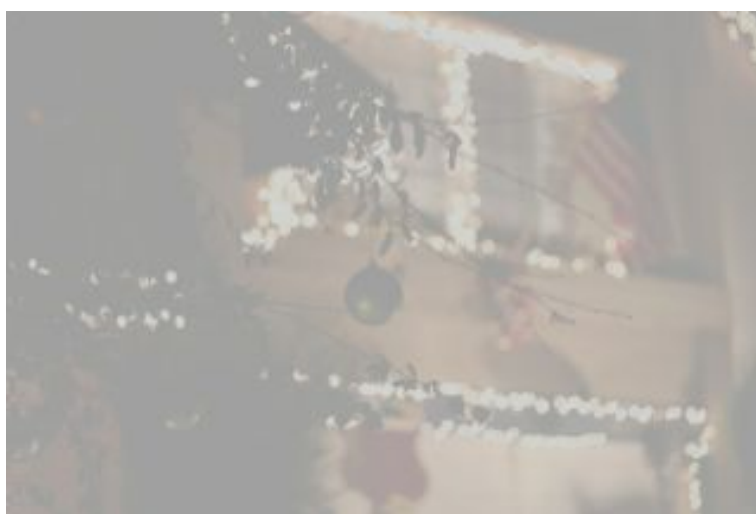


Figure 2: No Contrast

In that image the contrast is at an average value:



Figure 3: Average Contrast

In this image I reduced the amount of contrast to zero:



Figure 4: No Contrast

In that image the contrast is at an average value:



Figure 5: Average Contrast

“For short periods (about the length of a "single" played on the radio or in the disco), power and loudness can grab our initial attention. But at home, variety of dynamics maintains our interest for long periods of time. Good music written for a long-term musical experience contains a judicious mixture of variety and similarity in dynamics. A production which is relentlessly loud (or relentless in its sameness) can become boring very fast.”³

Bob Katz

3 WHAT IS THE PROBLEM?

The deficits that we are dealing with in today’s audio world can be derived from an accordance based on psychoacoustic factors. Our problems according to audio content are a result of a rather trivial thought.

3.1 Louder = Better

Even a small increase of loudness can make an audio signal sound clearer and of higher sound quality. Furthermore louder signals tend to rather attract our attention than quieter signals. This causes the average consumer to rather listen to something that is louder. Of course the industry knows about this and has adjusted the way they treat their audio signals to the way “we listen”. These attributes, the way we listen, are grounded deeply in our human nature.

For the last decade’s music, commercials, TV and so on have steadily increased in loudness. Since the 80s average CD levels went up about 20dB in loudness. This is done through using various techniques, which I will describe in the following. These techniques bring a number of problems with them. Another cause for problems is that we are metering peaks in audio technology. We are not measuring loudness, which is a subjective impression and will be discussed in the chapter “Loudness”. Metering peaks only gives us a boundary that we cannot cross. Peak metering doesn’t tell how loud a signal is.

³ <http://www.digido.com/articles-and-demos12/13-bob-katz/23-compression.html>

What's is the most commonly used button on a TV remote?

It is the volume control, used to balance between loud and highly compressed signals and more dynamic and partially quieter signals.

According to a presentation that Florian Camerer held, called "EBU R128 Introduction", these loudness differences within the broadcasting chain are causing loudness jumps between for example a loud commercial and low in level documentary. Sound engineers at TV broadcast stations have to fit highly compressed commercials between other more dynamic programs. As they are still orientating on peak levels loudness jumps occur.

The EBU R 128 recommendation focus lies on TV, but the EBU wants to bring the new standard to all areas of audio and media broadcasting and production in the future.

The presentation can be found on the Internet under following link:

<http://www.youtube.com/watch?v=iuEtQqC-Sgo>

4 LOUDNESS

4.1 What is loudness?

"Loudness is the characteristic of a sound that is primarily a psychological correlate of physical strength (amplitude). More formally, it is defined as "that attribute of auditory sensation in terms of which sounds can be ordered on a scale extending from quiet to loud.

Loudness, a subjective measure, is often confused with objective measures of sound strength such as sound pressure, sound pressure level (in decibels), sound intensity or sound power.

Loudness is also affected by parameters other than sound pressure, including frequency, bandwidth and duration."⁴

⁴ <http://en.wikipedia.org/wiki/Loudness> Accessed on 26.12.2012

Loudness is a subjective impression. That means that each individual has his/her own perception of loudness, which is based on whether he or she likes the content, age, mood, background, and so on...

The same peak level doesn't result in the same level of perception. Even with peak levels being 10 dB apart the perception could remain the same.

Perception is a subjective measure.

“As if this wasn't confusing enough, the frequency dependency of loudness perception itself varies with sound pressure level. Moreover, loudness perception is a highly subjective affair that depends on factors such as age and possible hearing damages. In short, loudness cannot be easily measured”⁵

Florian Camerer

5 THE PERCEPTION OF LOUDNESS

The way we perceive loudness depends on various factors. The following chapter will describe the perception of loudness according to Steven Errede, a Professor at the University of Illinois, USA.

5.1 The human Ear

The human ears respond to pressure variations (sound). Our ears can only detect sound within a certain frequency range, which is at the best located between 20 Hz – 20 kHz. Our ability to hear in this range of frequency is decreasing the older we get. The loss of hearing capacity is depending on external ascendancies. Furthermore our ears respond to the intensity of the sound.

Depending on where the sound, our ears receive, is originated from, we might perceive it differently. A sound coming from behind us tends to catch more atten-

⁵ <https://auphonic.com/blog/2012/08/02/loudness-measurement-and-normalization-ebu-r128-calm-act/> Accessed on 27.12.2012

tion from us than a sound coming from in front of us. This has been developed through human evolution. A sound coming from behind us could mean danger.

5.2 What is Sound?

“A type of longitudinal wave that originates as the vibration of a medium (such as a person's vocal cords or a guitar string) and travels through gases, liquids, and elastic solids as variations of pressure and density. The loudness of a sound perceived by the ear depends on the amplitude of the sound wave and is measured in decibels, while its pitch depends on its frequency, measured in hertz.”⁶

According to Prof. DI Hannes Raffaseder sound defines mechanical oscillators in an elastic medium within the frequency range of human hearing.

$$20\text{Hz} < f < 20000\text{Hz}$$

5.3 Apparent Loudness Level: Phons

The perceived response of average human hearing to constant loudness levels (sound intensity levels) is not independent of frequency. The response of the human ear for very low (< 20 Hz) and very high frequencies (> 20 KHz) is increasingly poor. Note that the open- closed $1/4\text{-}\lambda$ resonances associated with the inner ear canal affect our loudness level response.

Because human hearing is not flat with frequency, the perceived, or apparent loudness of a sound depends on frequency, and also on the actual intensity I (in Watts/m²), or equivalently, the actual loudness LI (in dB) {or sound pressure level $LP = \text{SPL}$ (in dB) of the sound.

In 1933, Fletcher and Munson obtained average values of the apparent loudness of sounds for human hearing as a function of these variables. The unit of apparent loudness L_{app} is the Phon, defined as the value of the SPL that has constant apparent loudness for (average) human hearing.⁷

⁶ <http://science.yourdictionary.com/sound> Accessed on 27.12.2012

⁷ ©Professor Steven Errede Department of Physics, University of Illinois at Urbana-Champaign, Illinois
http://courses.physics.illinois.edu/phys406/Lecture_Notes/P406POM_Lecture_Notes/P406POM_Lect5.pdf Accessed on 29.12.2012

The figure below shows the Fletcher-Munson curves – contours of constant apparent loudness $L_{app}(f)$ vs. frequency, f .

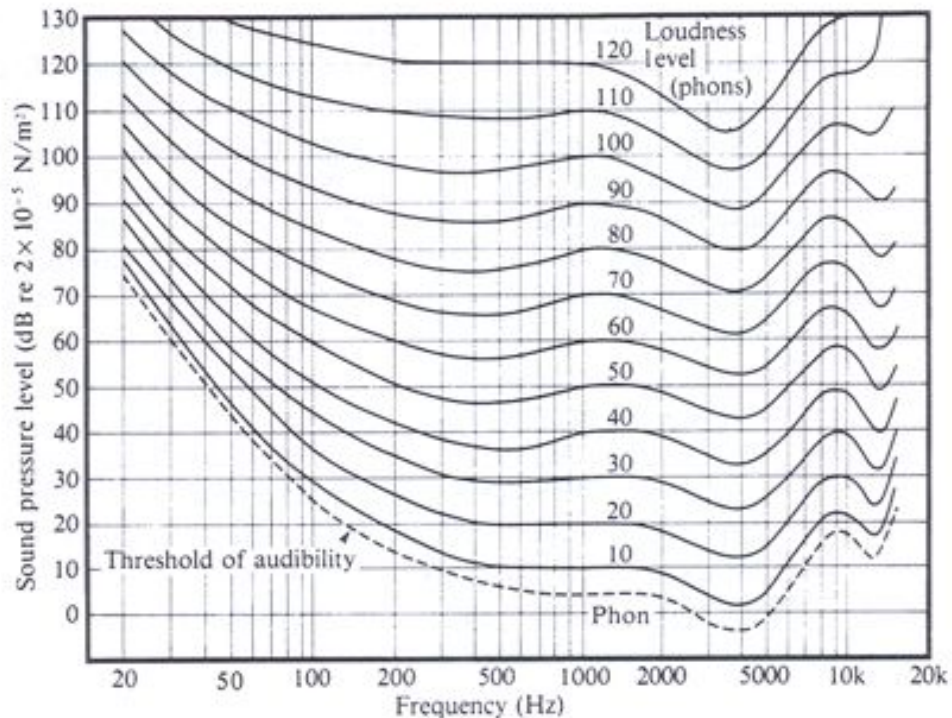


Figure 6: Fletcher-Munson Curves

As already mentioned humans hear in a certain frequency range: 20 Hz – 20Khz.

Why do we only hear in this certain spectrum of frequency?

Why don't we hear in higher or lower frequencies than 20Hz to 20 KHz?

According to Professor Steven Errede this is not all an accident. As humans being social animals he says, it is a logical consequence that we are primarily focused to hearing sounds produced by other humans.

“The frequency range of sounds produced by our own voice(s) – the totality of the physics associated with air as a medium + vibrating vocal chords in our larynx/voice box + hyoid bone + acoustic cavities of our lungs + throat + mouth + nasal passage/sinus cavity dictates what the acoustic power spectrum of the human voice can/cannot be.”⁸

Imagine this commonly occurring situation:

You are sitting in the metro surrounded by many people. Everybody is talking, laughing, and producing some kind of sound around you. The train makes its noise as it moves through the tunnel and you have your earphones plugged in. Suddenly a baby starts crying. This sound cannot be ignored. It is of such great intensity that everybody in the metro will have a hard time trying to ignore it. No matter how focused the passengers are on their newspaper, cell phone and so on...

Furthermore Professor Errede talks about the infra sound ($f < 20\text{Hz}$) region. He mentions that if we would hear well in this frequency range, we would constantly be disturbed by hearing the draft of wind and other low frequency sounds as we are walking or running.

The following image shows a brief loudness overview and compares noises and sounds that most humans hear on a regular basis with an amount in dB.



Figure 7: How Loud is too loud?

⁸ ©Professor Steven Errede, Department of Physics, University of Illinois at Urbana-Champaign, Illinois
http://courses.physics.illinois.edu/phys406/Lecture_Notes/P406POM_Lecture_Notes/P406POM_Lect5.pdf
Accessed on 29.12.2012

6 COMPRESSING AND LIMITING OF AUDIO SIGNALS

6.1 Compression

Compression is used to control energy, peak level and dynamic range of a signal. The dynamic range of an audio signal is being reduced in order to make the whole signal louder. In fact a compressor just controls the level of an audio signal automatically. That means that the compressor reduces the level of the audio signal as soon as it exceeds a certain threshold set by the user.

There is a lot more to say about compression. This is just a very brief explanation of a compressor's very basic functionality.

6.2 Limiting

A Limiter is a type of compression. It is basically a compressor that limits the level of a signal to a certain threshold to prevent a signal from clipping. The Limiter prevents signal peaks that would be too loud or cause distortion.

6.3 Tools

There are hundreds of tools on the market to compress audio signals. Compressors can be found as hard or software versions. Most DAWs (Digital Audio Workstations) come with a software plugin bundle that features compressors, limiters, EQs, reverbs and so on.

Depending on the style of music there is a compressor suitable for every signal. Some compressors are mostly used for processing vocals while others are more commonly used for editing drums etc....

A famous hardware compressor mostly used for vocals.

The UREI 1176:



Figure 8: UREI 1176

This waves software compressor Plugin is used by engineers, musicians and producers around the world. This is the so-called “Waves Renaissance”:



Figure 9: Waves Renaissance

Another Plugin made by the company called “Waves” from Israel. This is an image of the L1 Ultramaximizer. A Limiter known for adding transparency and punch to an audio signal.



Figure 10: L1 Ultramaximizer

As Mr Camerer says Compression is one solution, but it's not the best solution. This statement will be illustrated in the chapter about normalization. When an audio signal is being highly compressed its dynamic range will be reduced drastically. Highly compressed audio signals lose their sound quality and the so-called "pumping" can occur. People are recording audio on high quality equipment and then the sound quality is being sacrificed for the sake of loudness. To me that seems counterintuitive.

On the other hand when you think about broadcasting you need your audio to have certain loudness. Think of someone driving on the highway in a rather old car. It is summer and this poor guy doesn't have air condition, so he opens the window. Now the driver wants to listen to some music on the radio. Think of a classical concert with a huge dynamic range. Quiet flutes, percussions and other low level elements will not be heard, while a loud and compressed pop song will be heard in its entirety. Not in stunning quality, but loud enough. When thinking about this situation a certain loudness level makes sense of course.

7 NORMALIZATION

7.1 What is normalization?

To normalize an audio signal means applying a certain amount of gain to an audio signal to bring the average peak level to a certain target level. This can be done easily with every DAW on the market. The signal to noise ratio (SNR) will stay the same while conducting normalization. That means that the more the level of the wanted audio signal is being increased the more the noise interference will be scaled up.

7.2 Signal to Noise Ratio (SNR)

SNR is a measurement to describe the level of noise an output device generates according to the level of the desired signal. Every device generates noise. It is just about the amount of noise that is acceptable. The noise level should be as low as possible and the desired audio signal should reside as close to the target level as possible. This would mean that the signal and the noise are far apart from each other.

er and therefore the SNR would be high. If the SNR is low the audio signal will not sound as good and when gaining or normalizing the signal not only the desired signal will be made louder but also the noise level. Signal to noise ratio measurements are usually expressed in dB

(This means that gaining loudness tends to result in a loss of dynamic. This causes music to sound flat and boring. The EBU R 128 standard tries to bring back dynamic in the audio world, so that it sounds like it did in the earlier days again.)

7.3 About Peaks

In his presentation about EBU R 128 Florian Camerer explains:

He says we are not even measuring the peaks in a true sense. We are not measuring the peak but we are measuring something below the peak. That has nothing to do with loudness what so ever, because the loudness level would in this example probably be the centre of gravity of this iceberg. We don't even see it because it's hidden below the surface. So it doesn't have anything to do with the peak level. We are measuring hills not peaks => QPPM Measuring in Europe

QPPM stands for Quasi Peak Program Meter

So we are not even measuring the real peaks but we are measuring quasi peaks. Furthermore really short transients get not displayed correctly on a QPPM meter. In order for them to be still transmitted we need a concept called headroom.

See the illustrations on the next page:

This figure illustrates the way we measure peaks. What have been measuring were not true peaks.



Figure 11: Iceberg1

We did not measure what was hidden under the surface.



Figure 12: Iceberg 2

The centre of gravity represents loudness.

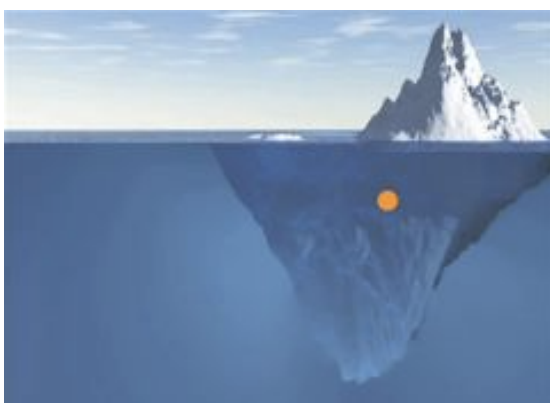


Figure 13: Iceberg 3

7.4 Peak Normalization VS Loudness Normalization

“The still widespread audio levelling concept of peak normalization with reference to a Permitted Maximum Level (PML; for example, -9 dBFS), has led to uniform peak levels of programs, but widely varying loudness levels. The actual variation is dependent on the degree of dynamic compression of the signal. In contrast, loudness normalization achieves equal average loudness of programs with the peaks varying depending on the content as well as on the artistic and technical needs. Provided the loudness range of a program is within the permitted tolerance, the listener can enjoy a uniform average loudness level over all programs, thus not having to use the remote control for frequent volume adjustments any more.”⁹

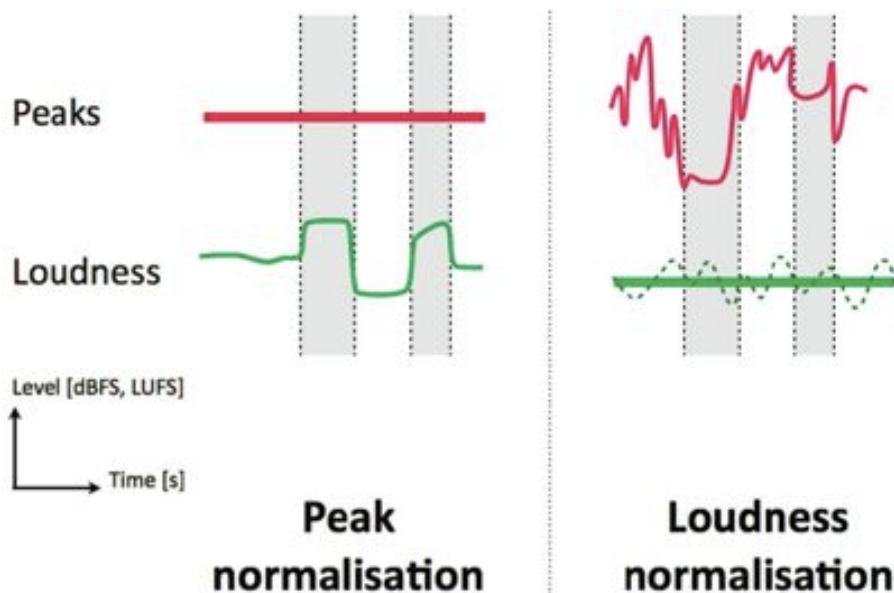


Figure 14: Peak/Loudness Normalization

In the following I will explain and illustrate the difference between peak normalization and loudness normalization according to Florian Camerer’s lecture titled “EBU R128 Introduction”.

His lecture can be found under the following link:

<http://www.youtube.com/watch?v=iuEtQqC-Sqo>

⁹ EBU – Tech 3343 „Practical guidelines for Production and Implementation in Accordance with EBU R128“ V2.0; p.15

7.4.1 Peak Normalization

When using peak normalization the gain is being adjusted in a way that the signal level equals a desired peak level (in the following illustrations the peak level is represented by a red line). Peak normalization does not indicate the apparent loudness of the signal since it is only working with the highest levels/amplitudes of the signal. Peak normalization can be useful to prevent clipping of an audio signal. Generally the concept of peak normalization has been useful in areas such as mastering or broadcasting audio signals.

All the tools in the following illustration should represent individual programs. A program is a self-contained item that is being measured from start to stop. The size of the tools represents the dynamic range of the particular program. They are all aligned according to their peaks, not their loudness.

The sword (1) in the illustration (Fig.15) is a large symbol that means that it represents a program with a large dynamic range. That might be a highly dynamic action film in which low-level audio signals such as whispering and high-level audio signals such as a gun battle can occur.

The small tool (2) next to the sword represents a highly compressed commercial. This program has a very small dynamic range.

Both signals have the same peak level, but their dynamic range and loudness is different.



Figure 15: Peak Normalisation 1

The green circles in the picture below (Fig.16) represent the loudness level, so to say the centre of gravity of these programs. You can see huge jumps between dynamic material and highly compressed material.

In the following illustration the problem is clearly recognizable.

If you draw a line from circle to circle you can see all these loudness jumps between the individual programs. This happens because peak normalization does not consider loudness levels. Loudness jumps are very unwanted effects of peak normalization and a reason for problems within broadcast technology.

As a consequence of these loudness jumps TV consumers for instance have to deal with the different loudness of audio signals that means that they have to constantly adjust the volume of their devices.

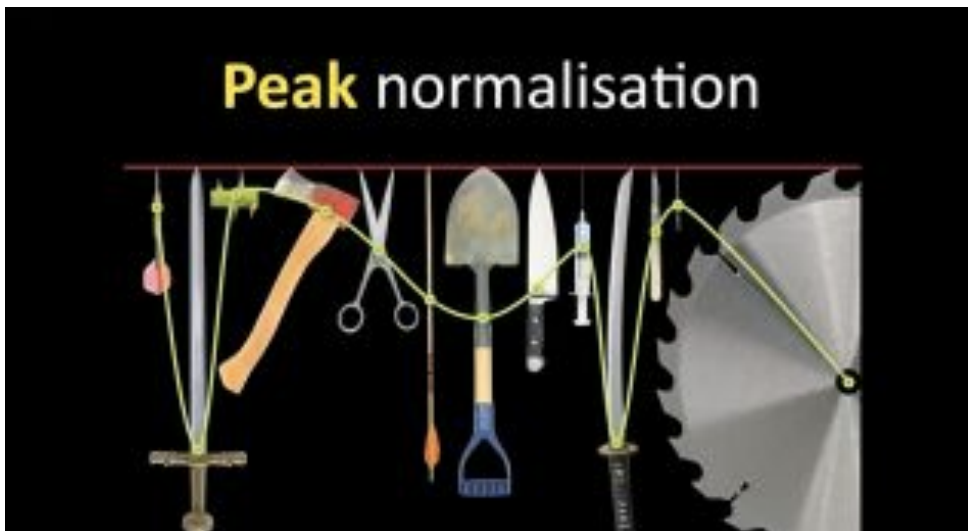


Figure 16: Peak Normalisation 2

What you see in the next picture (Fig.17) below is what happens in today's broadcast technology. All the programs are being highly compressed to reduce their dynamic range.

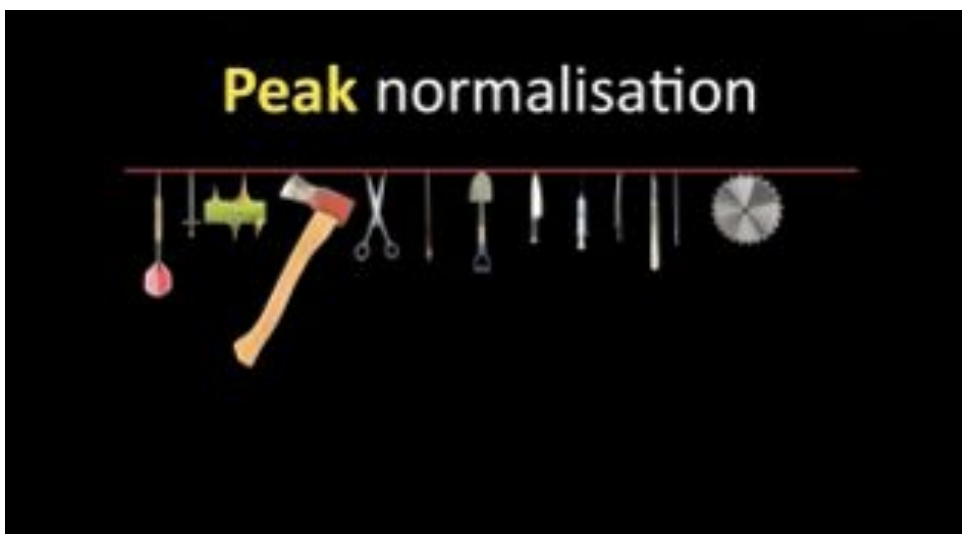


Figure 17: Peak Normalisation 3

After being compressed all the loudness levels are aligned to the same level. Theoretically the technical problem is solved but we sacrifice a lot!

We sacrifice all the program's dynamic range.

If we would look at the example of the action movie (e.g.: the sword symbol above) the dialog would now be as loud as a gun battle, which may work for an action movie but may not work for other programs. (Fig.18)

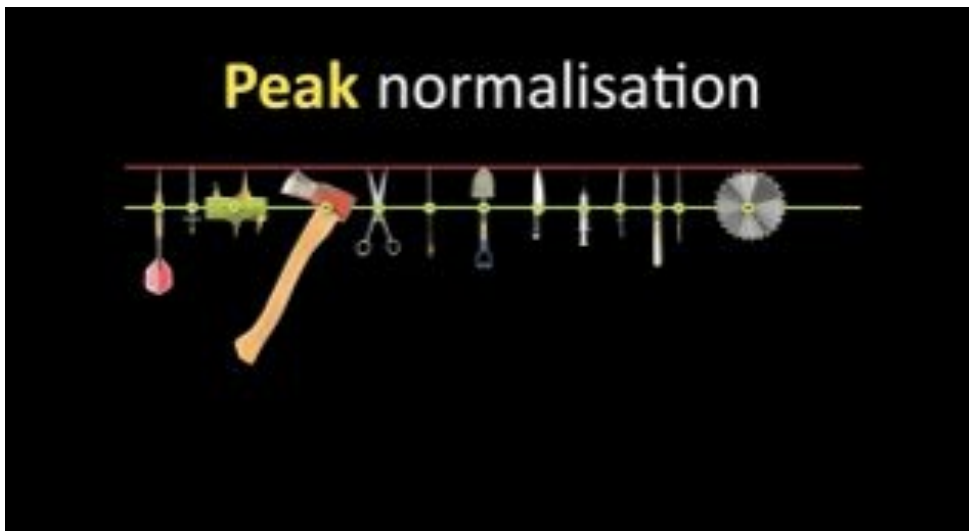


Figure 18: Peak Normalisation 4

7.4.2 Loudness Normalization

Another type of normalization is the so-called “Loudness Normalization”. Loudness normalization does not depend on a peak levels it is based on the measure of average loudness (measurement of loudness is described in the chapter loudness).

According to Mr Camerer loudness normalization is a solution to the problem. Loudness normalization gets rid of the peak reference on the top (the red line in the picture above). Instead of aligning the programs to a peak level they are aligned to a loudness reference level according to their average loudness levels so to say their centres of gravity. (Fig.19)



Figure 19: Loudness Normalisation

The EBU is considering according to Technical Documents:

EBU Tech Doc 3205-E; ITU-R BS.645; ITU-R BS.1770; EBU Tech Doc 3341;

EBU Tech Doc 3342; EBU Tech Doc 3343; EBU Tech Doc 3344

- that peak normalization of audio signals has led to considerable loudness differences between programs and between broadcast channels;
- that the resulting loudness inconsistencies between programs and between channels are the cause of the most viewer/listener complaints;
- that, when used to read peaks in the usual way, the QPPM (Quasi-Peak Program Meter) specified in EBU Tech Doc 3205-E [1] does not reflect the loudness of an audio signal, and that the QPPM is not designed to indicate a long-term average;
- that with the proliferation of digital production, distribution and transmission systems, the permitted maximum level of an audio signal specified in ITU-R BS.645 [2] is no longer appropriate;
- that an international standard for measuring audio program loudness has been defined in ITU-R BS.1770 [3], introducing the measures LU (Loudness Unit) and LUFS (Loudness Unit, referenced to Full Scale)1;

- that a gated measurement of Program Loudness (hence measuring 'Foreground Loudness') is advantageous to improve the loudness matching of program with a wide loudness range;
- and that the measure 'Loudness Range' can be used to assess the need for loudness-range reduction to fit program to the tolerance window of the target audience;

Audio loudness normalization & permitted maximum level EBU R 128-2011

The EBU further recommends:

- that the measures Program Loudness, Loudness Range and Maximum True Peak Level shall be used to characterize an audio signal;
- that the Program Loudness Level shall be normalized to a Target Level of -23.0 LUFS. The permitted deviation from the Target Level shall generally not exceed ± 1.0 LU for program where an exact normalization to Target Level is not achievable practically (for example, live program);
- that the audio signal shall generally be measured in its entirety, without emphasis on specific elements such as voice, music or sound effects;
- that the measurement shall be made with a loudness meter compliant with both ITU-R BS.1770 and EBU Tech Doc 3341 [4];
- that this measurement shall include a gating method as specified in ITU-R BS.1770 (summarized in EBU Technical Document 3341);
- that Loudness Range shall be measured with a meter compliant with EBU Tech Doc 3342 [5];
- that the Maximum Permitted True Peak Level of a program during production shall be -1 dBTP (dB True Peak), measured with a meter compliant with both ITU-R BS.1770 and EBU Tech Doc 3341.

- that loudness metadata shall be set to indicate -23 LUFS for each program that has been loudness normalized to the Target Level of -23 LUFS;
- that audio processes, systems and operations concerning production and implementation should be made in compliance with EBU Tech Doc 3343 [6];
- that audio processes, systems and operations concerning distribution should be made in compliance with EBU Tech Doc 3344 [7].

8 THE LOUDNESS WAR

What is the difference between an album released in the 80s and music that is being released today?

If you listen to records from the 70s and 80s and compare them to today's records you will realize that music has become louder. In the last decades the music and broadcasting industry has constantly increased the loudness of audio. This phenomenon is called "Loudness War". The loudness war started in the early 80s. The real loudness race started with the invention of the CD and with its clearly defined maximum peak level. Audio and mastering engineers have since that found ways to make their content louder. This is mostly being done by compressing the material drastically so that the overall level of the audio signal can be increased.

"The louder the better" has become some kind of mantra to many people in the field of media production. The louder the audio the more people pay attention.

This thought started some kind of vicious circle. Everybody had to be as loud the others to be heard. Producers of audio content were and still are afraid that if they are not releasing their audio as loud as possible it will get lost beside all the other loud programs. So the only thing left to do was and still is to be as loud or even louder as everybody else.

In the following figures you will see how drastically the loudness has increased in the last decades. You can also see how the increase of loudness brings a loss of dynamic range.

This is what a song produced in the early 80s would look like on a meter. The average dynamic range this song is about 14dB.

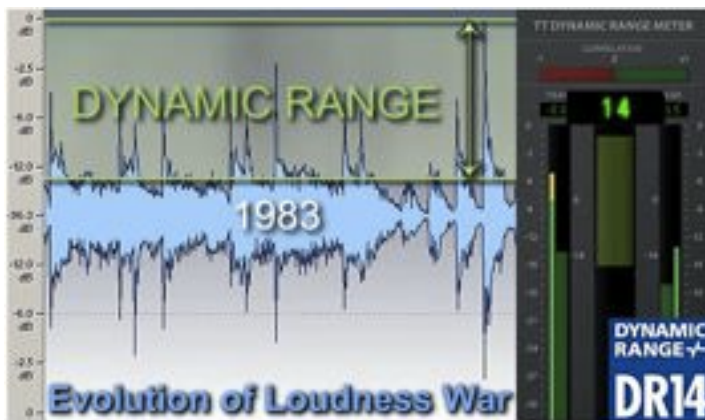


Figure 20: Dynamic Range 1983

This is what a song produced in the early 90s would look like on a meter. The average dynamic range this song is about 12dB.



Figure 21: Dynamic Range 1991

This is what a song produced in the late 90s would look like on a meter.
The average dynamic range this song is about 9dB.

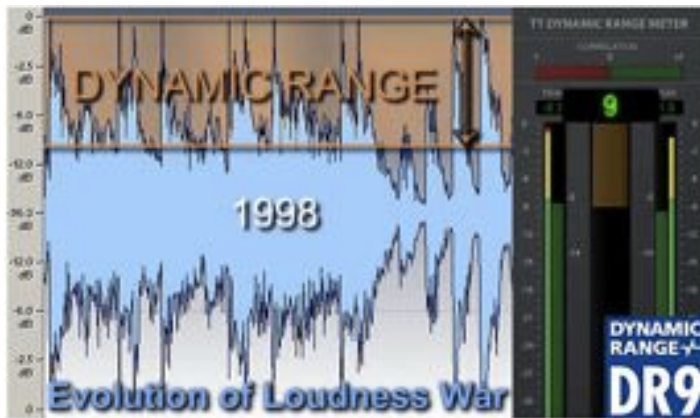


Figure 22: Dynamic Range 1998

This is what a song produced in the year 2003 would look like on a meter.
The average dynamic range this song is about 6dB.



Figure 23: Dynamic Range 2003

This is what a song produced in the year 2003 would look like on a meter. The average dynamic range this song is about 4dB.



Figure 24: Dynamic Range 2008

These images of the same song, compared from the 1980s up to the year 2008, make dimension of the problem visible. Some records, commercials, TV Shows and so on are made so loud that they just don't sound good anymore. With the new EBU standard media content producers wouldn't have to be trying so hard to be as loud as everybody else anymore. This problem just wouldn't exist anymore and highly compressed programs would even be punished because they would play very low.

The following figures and ideas are taken from a lecture Bob Katz did called "Loudness War and Peace". This lecture can be found on the Internet under following link:

<http://www.youtube.com/watch?v=u9Fb3rWNWDA>

Bob Katz is a famous audio and sound engineer who has mastered Grammy award winning albums and published books about mastering. In his lecture he talks about the "Weapons of War" and how engineers were able to constantly increase the loudness of their audio over the years.

In the figure below Bob Katz illustrates how technical inventions made the increase of loudness possible. In the 80s high compression was not possible because the necessary tools had not been invented. Heavy compression started in the 90s when DSP slowly became affordable. Another big step was the invention of the MP3 and portable MP3 players, such as the “iPod”.

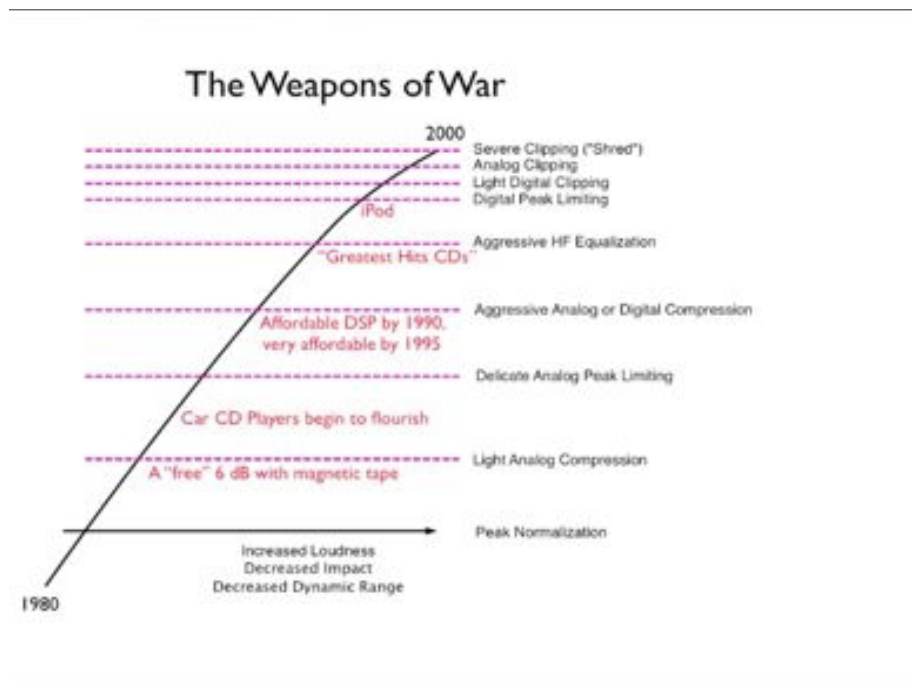


Figure 25: Weapons of War

In the following picture (Fig.26) Bob Katz shows an image of an audio waveform. This is a song he mastered for a at that time client. In the first master Bob Katz tried to get the track to an appropriate loudness level, which could compete to the loudness levels of other mastered tracks, while keeping some dynamic range so that the song could have it's louder and softer parts. Mr Katz wanted to have the bridge build up to the chorus and also have some room for louder peaks such as the drums. When presenting the well-balanced master to his client his mastered track was rejected for being too low. He went back to the studio to re-master the track according to following steps:

- He started inside of a DAW and he clipped the signal going out of the DAW into some very strong aggressive parallel compression in the digital domain.

- Out of the digital domain he clipped once again and he sent a signal into the ADC converter that was hard enough to put it 5dB into overload.
- Out of the ADC he went into the digital domain and met up with some peak limiting.

After executing these steps the song was loud enough to fulfil the client's demands. But these steps caused the audio material digital distortion, which is just being masked by the tracks loudness. Increasing the loudness over a certain level causes audible digital distortion. In the end this was what the client preferred. In the image below (Fig.26) you can see the first and second master's waveform that Mr Katz handed to his client.

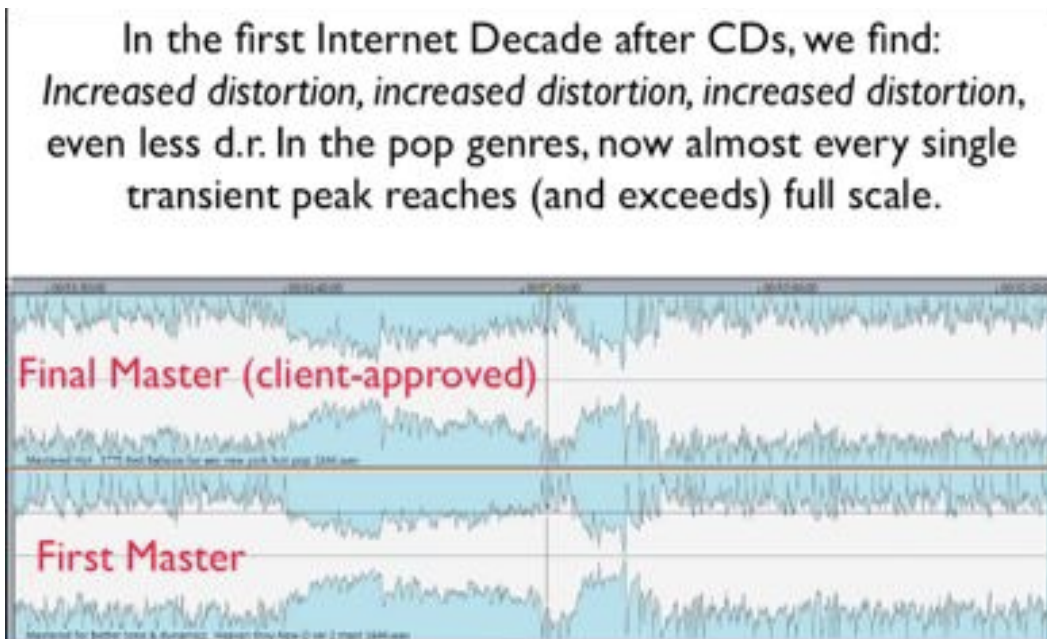


Figure 26: Master Waveform

Here is what Mr Bob Katz says about loudness normalization:

- Loudness normalized media will take away the competitive advantage of compression.
- Loudness normalized media reveals that over compressed masters sound wimpy, small and distorted.

- We hope that music mixing and mastering will return to a more dynamic style, or at least the media will permit dynamic recordings to coexist with compressed recordings.
- Artist will again have unencumbered choice on how to produce their recordings.

Loudness war and peace by Bob Katz, Accessed on 24.01.2013

<http://www.youtube.com/watch?v=u9Fb3rWNWDA>

9 WHAT IS EBU R 128?

R 128 is a new standard to measure the loudness level in audio broadcasting. So far audio measurement was bound to peak levels, which only orientated on the loudest amplitude of the audio material.

EBU R128 manages to come closer to the perceived loudness, which of course is subjective. So there is no perfect measurement for perceived loudness. "There is no perfect box for metering audio signals, but with the EBU R128 we finally come as close as we have never been before."¹⁰

This new standard can be used in areas such as television, news, sport, advertising, radio and so on.

9.1 Units

With implementation of R128 new units will be established to characterise an audio signal:

LUFS => Loudness Units relative to Full Scale

LU => Loudness Units

LRA => Loudness Range

dBTP => dB relative to True Peak

¹⁰ Florina Camerer <http://www.youtube.com/watch?v=iuEtQqC-Sgo> Accessed on 08.01.2013

EBU R 128 will bring back dynamic in audio signals. That means that there will be a balance between loud and quiet signals. For years people had been trying to make each signal sound as loud as possible. This happens by the use of compression. The signals dynamic is being limited so that the general loudness can be increased.

9.2 Audio Parameters

3 Audio Parameters are defined (that shall characterize an audio signal with all it's characteristic information):

Program Loudness => the most important one, indicates how loud your program (program is one self contained item, 45min TV show, 2 min commercial, or 2h feature film) is on average. This is 1 number, it is used to normalize, to balance to the next program.

Maximum True Peak Level => the true peak level indicates the maximum (positive or negative) value of the signal waveform in the continuous time domain

Loudness Range => indicates the difference between the average soft and the average loud parts of a program¹¹

With these 3 parameters we know:

How loud the audio signal is

We know what the true peak level is

We know how dynamic it is

With QPPN we didn't know any of these

EBU defines a new reference => one common loudness level => -23 LUFS

-23 LUFS is the reference for the future

“EBU R 128 establishes a predictable and well-defined method to measure the loudness level for news, sports, advertisements, drama, music, promotions, film etc. throughout the broadcast chain and thereby helps professionals to create robust specifications for ingest, production, play-out and distribution to a multitude of platforms. R 128 is based entirely on open standards and aims to harmonize the way we produce and measure audio internationally.

¹¹ EBU Technical Document 3343; Version 2.0; p.11 - 13

The basis of R128 is ITU-R BS.1770, the result of extensive work by the International Telecommunication Union. The purpose of that standard was to establish an agreed open algorithm for the measurement of loudness and the true peak levels of programs. It is a robust standard, which has the benefit of a simple implementation. In brief, it defines a “K-weighting” filter curve (a modified second-order high-pass filter, see Figure below), which forms the basis for matching an inherently subjective impression with an objective measurement.

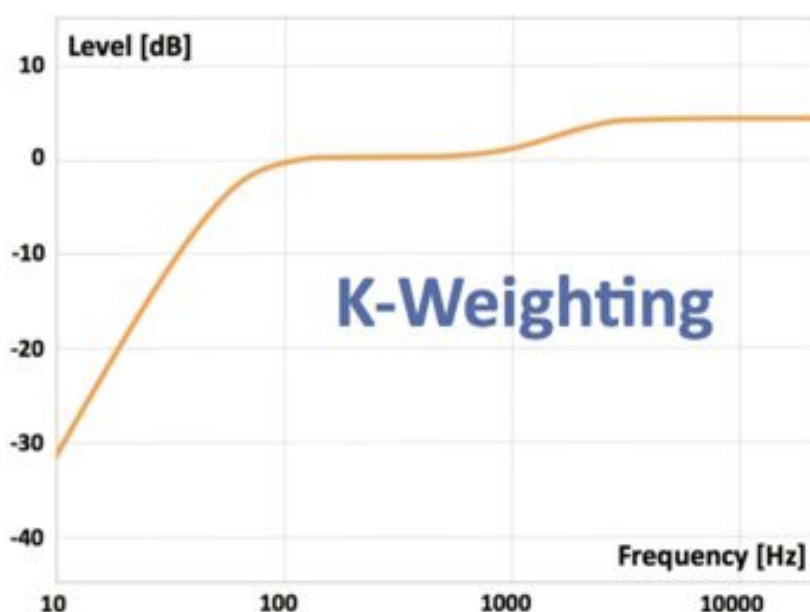


Figure 27: K-Weighting

This weighting curve is applied to all channels (except the Low-Frequency Effects Channel (LFE) which is currently discarded from the measurement; see below), the total mean square level is calculated (with different gain factors for the front and surround channels; see Figure 2) and the result is displayed as “LKFS” (Loudness, K-Weighting, referenced to digital Full Scale), or “LUFS” (Loudness Units, referenced to digital Full Scale). For relative measurements, Loudness Units (LU) are used, where 1 LU is equivalent to 1 dB

Practical guidelines for Production & Implementation of R 128 Tech 3343-2011v2 Low Frequency Effects (LFE) channel

The Low Frequency Effects channel (the “.1”-channel in “5.1”) of a multichannel audio signal is currently not taken into account for the loudness measurement ac-

According to ITU-R BS.1770. This may lead to abuse of the LFE with unnecessary high signal levels. On going investigations try to evaluate the subjective effect the LFE has on the perception of loudness as well as the appropriate way to include it in the objective loudness measurement.

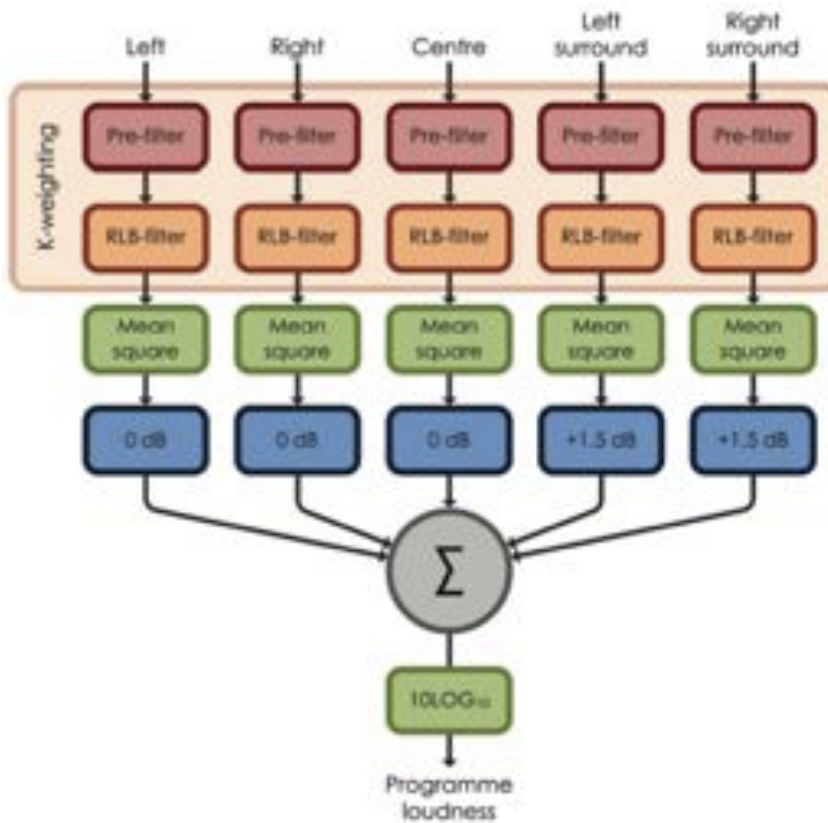


Figure 28: Channel Processing and Summation

Whereas BS.1770 defines the measurement method, R 128 extended it by actually defining a specific 'Target Level' for loudness normalization as well as a gating method to improve the loudness matching of programs, which contain longer periods of silence or isolated utterances. The EBU's development was needed to accommodate the needs of program makers, with particular regard to having a means to measure complete mixes (rather than just one component, such as speech or music) and the loudness range of the program.

To do this, the EBU has specified three new measures:

Program Loudness

Loudness Range

True Peak Level”¹²

9.3 Program Loudness

Program loudness is the integrated loudness measured over the full duration of a program. The program loudness parameter consists of one number. The first one shows the LUFS and the number after the decimal point shows how loud the program is on average. The program loudness measurement includes a gating function, which pauses the loudness measurement as the signal drops below a certain threshold. The gate is necessary to prevent the measurement from being too low in value in case of signals with long pauses. When normalizing these signals they would become too loud.

The EBU’s target level to which every audio signal should be normalized to is:

-23.0 LUFS (-10rel gate)

Absolute measure => LUFS is an absolute measure for a certain reference.

An example: “Your program has a loudness level of minus 20 LUFS”

(In digital systems we have the reference of 0 dBFS.)

9.4 Loudness Range

Loudness Range (LRA) provides signals not to exceed a certain range measured in LU. This measurement is based on the statistical distribution within a program. It excludes extremes in within the audio signal so that for example a single gunshot or explosion does not affect the whole program.

“Loudness range is a generic measure that helps to decide if dynamic compression is necessary.”¹³

¹² EBU – Tech 3343; Version 2.0; p.9-11

¹³ EBU – Tech 3343; Version 2.0; p.12

9.5 True Peak Level

True Peak Level defines the highest value of a program's constant audio signal waveform.¹⁴

10 EBU RECOMMENDATIONS AND BROADCAST STANDARDS

EBU (European Broadcasting Union) is an alliance of public service media entities comprising 74 active members in 56 countries and 37 associate members from a further 22 countries.

EBU R 128 is based on ITU BS.1770. So some of these recommendations (standards) go hand in hand.

The following chapters will give an overview about the different standards:

10.1 ITU 1770

“In many ways, The International Telecommunication Union's BS.1770 recommendation is global and one of the most important broadcast standards as many other standards are based on it. The ITU standard concerns Broadcast Loudness and True-peak Level measurement, and the loudness part is based on an Leq measurement employing K-weighting, which is a specific frequency weighting developed by the Communications Research Centre (a federal research institute in Ottawa, Canada). This baseline method is relatively simple, but it is based on excessive listening tests and has been verified independently. The True-peak part of the standard was specified by AES SC-02-01. As mentioned, many other broadcast standards are based on ITU BS.1770, including ATSC A/85 (the US), EBU R128 (Europe), OP-59 (Australia) and TR-B32 (Japan). In 2011, the recommendation was released in a revised version: ITU BS.1770-2 that employs a relative gate with regard to Program Loudness measurement, which was adopted from the R128 standard defined by The European Broadcast Union (EBU). In 2012, the recommendation was updated once again to become version BS.1770-3. ITU

¹⁴ EBU – Tech 3343; Version 2.0; p.13

BS.1770 published 2006 ITU BS.1770-2 published March 2011 ITU BS.1770-3 published August 2012

10.2 R 128

The P/LOUD group, which is part of The European Broadcasting Union (EBU), has defined the R128 standard based on ITU BS.1770. However, the group also added new tools such as a relative gate that ensures even more consistent loudness across genres and types of program material. Some of these tools have been implemented in the updated version of ITU's recommendation: ITU BS.1770-3.

Basically, the R128 standard builds on 4 tech documents: EBU Tech 3341, EBU Tech 3342, EBU Tech 3343 and EBU Tech 3344.

10.3 EBU Tech 3341

This document specifies loudness metering within the R128 domain. Quite simply, a loudness meter is allowed to use the 'EBU Mode' term if certain criteria are met. First of all, three time scales must be available: Momentary (M), Short-term (S) and Integrated (I) - also referred to as Program Loudness. Any real-time 'live meter' with EBU Mode must be able to display the three time scales, though not necessarily at the same time, and it must also be able to display the maximum value of the Momentary Loudness (reset when Program Loudness is being reset). Momentary Loudness must be measured using a sliding time window of 0.4 seconds, while Short-term Loudness must be measured using a sliding time window of 3 seconds. Program Loudness must be measured using a specific gating method that excludes measurements of parts dropping below a threshold of -10 LU relative to an ungated measurement of the same program material. Further, a meter featuring EBU Mode must also be able to display LRA (Loudness Range), which is a measurement of the variation of loudness on a macroscopic scale. This parameter is a supplement to the overall loudness measurement (Program Loudness). The terms used for expressing the measurements are LU (Loudness Units) and LUFS, which is the same as LKFS (Loudness K-weighted Full Scale) used by other broadcast standards. The target loudness of EBU's R128 standard is -23 LUFS.

For in-depth specifications, please consult the original [EBU Tech 3341](#) document.

10.4 EBU Tech 3342

This document specifies audio normalization based on loudness measurements as described in EBU Tech 3342. The average loudness level, or Program Loudness, should be used in combination with Maximum True-peak Level and Loudness Range (LRA) to correct program material in accordance to the R128 specifications. In essence, LRA is determined by analysing the loudest and the softest parts of the program material. However, the lower percentile of 10% is being ignored as is the upper percentile of 95% to avoid extreme events such as a single gunshot or long passages of silence to manipulate the overall result in an undesirably way.

For in-depth specifications, please consult the original [EBU Tech 3342](#) document.

10.5 EBU Tech 3343

This document describes guidelines for production and implementation in accordance with EBU R128. Both Program Loudness, LRA and True-peak metering is explained and strategies for implementing a loudness strategy at various stages of production are offered.

For in-depth specifications, please consult the original [EBU Tech 3343](#) document.

10.6 EBU Tech 3344

This document describes how to loudness normalize when distributing program material to various end user platforms, including radio, television and portable devices in various formats such as stereo and 5.1 surround. For in-depth specifications, please consult the original [EBU Tech 3344](#) document.

10.7 ATSC A/85

ATSC A/85 is specified by The Advanced Television Systems Committee and applies to US broadcasters. A/85 is rooted in BS.1770 Loudness and True-peak level. It specifies an anchor based or a universal approach to audio normalization, but without a clear distinction of when to use what. Often, this anchor is defined by the 'speech' of a (television) program, which is effective for streamlining the loudness of that particular program, but not very efficient in terms of matching loudness between several different types of program material. A/85 includes extensive information about calibrated monitoring environments and may function somewhat like a Dolby manual. Unlike EBU R128, A/85 is only focused on the digital television platform and on the AC3 codec. Published November 2009.

10.8 TR-B32

TR-B32 is a Japanese broadcast standard that builds on ITU BS.1770-2, which means that a relative gate is employed. However, the target level is -24 LUFS/LKFS as opposed to the -23 LUFS target level of the EBU R128 standard which also employs the gate. As a rule of thumb, a gated measurement of -23 LUFS/LKFS equals an un-gated measurement of -24 LUFS/LKFS.

10.9 OP-59

Operational practice by Free TV, Australia. OP-59 is rooted in BS.1770 Loudness and True-peak level and recommends a speech based as well as a universal approach to audio normalization. All short form programs should be measured using the universal (full mix) method.”¹⁵

¹⁵ <http://www.tcelectronic.com/loudness/broadcast-standards/> Accessed on 28.12.2012

11 HARDWARE AND METERING

In media and broadcast technology gear to objectively measure loudness is needed.

Florian Camerer quotes that it is basically impossible to construct gear respectively a box that gives a perfect representation of your subjective impression. But the EBU managed to get as close to subjective impression as nothing ever before.

According to the EBU's opinion an audio levelling paradigm based on loudness measurement is needed.

“In addition to the average loudness of a program ('Program Loudness') the EBU recommends that the measures 'Loudness Range' and 'Maximum True Peak Level' should be used for the normalization of audio signals and to comply with the technical limits of the complete signal chain as well as the aesthetic needs of each program/station depending on the genre(s) and the target audience.”¹⁶

11.1 EBU Mode

Since the release of the new EBU R128 standard many manufactures have been producing new tools to display loudness on a meter according to the EBU standards. The EBU Mode was designed in cooperation with the manufactures. The following explanations are based on the EBU – Technical Document 3341.

11.1.1 EBU Mode Compliant Device

When a device implements that it is able to work in EBU mode has to work with following parameters. These parameters define the EBU mode:

¹⁶ EBU – Tech 3341; 2011; p.5

11.1.2 The heart of EBU mode:

These 3 different integration times form the EBU Mode's core:

Momentary => 400ms integration time without gate (to know: what is my levels now, immediate levelling)

Short Term => 3s integration time without gate, that means it's moving slower

Integrated => from start to stop works with a gate

11.1.3 The Gate

The integrated loudness measurement includes a gate. This gate excludes audio under a certain threshold from being part of measurement.

For more detailed information about the gating function se EBU Tech 3341.

11.1.4 Loudness Range Measure

“The measure ‘Loudness Range’ quantifies the variation in a time-varying loudness measurement; it measures the variation of loudness on a macroscopic time-scale. Loudness Range is supplementary to the measure of overall loudness, that is, ‘integrated loudness’. The computation of Loudness Range is based on a measurement of loudness level, as specified in ITU-R BS.1770.”¹⁷

11.1.5 Units

The EBU recommends using the following units:

A relative measurement => LU

An absolute measurement => LUFS

Loudness (L) and Frequency weighting (K) => LK

¹⁷ EBU 3342; 2011; p.5

11.1.6 Display

An EBU Mode compliant meter shall be able to measure and display the three main measures ‘Program Loudness’, ‘Loudness Range’ and ‘Maximum True Peak Level’.

The ‘EBU Mode’ loudness meter shall at least provide functionality that enables the user to:

“Start/pause/continue the measurement of integrated loudness and Loudness Range simultaneously, that is, switch the meter between ‘running’ and ‘stand-by’ states.

Reset the measurement of integrated loudness and Loudness Range simultaneously, regardless of whether the meter is in the ‘running’ and ‘stand-by’ state.

Every EBU meter has to be able to display all three of these integration times.”¹⁸

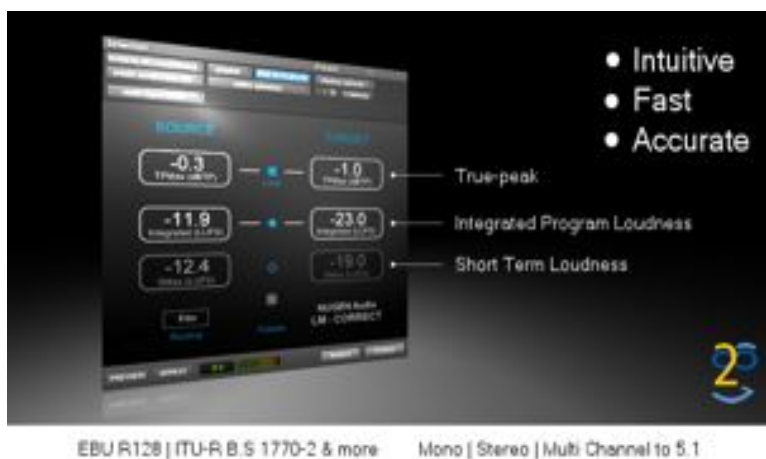


Figure 29: Nugen Audio LM Correct

¹⁸ EBU – Tech 3341; 2011; p.6



Figure 30: Nugen Audio, VisLm

WLM is an affordable, all-in-one cross-platform, multi-format loudness metering software solution.



Figure 31: Waves Loudness Meter

12 INTERNATIONAL

Will the EBU R 128 only affect Europe? R 128 has been developed in Europe, but it is not meant to be a European standard. Florian Camerer mentions in his lecture that we are now facing an audio revolution. It is his and many others intention to make a worldwide change.

“We had nothing else but the very humble goal to change the world and I think it's a very good motivation for any work you do.

Let's change the world. Even if it's only the small world around you.”¹⁹

Florian Camerer

EBU R 128 is based entirely on open standards and aims to harmonize the way we produce and measure audio internationally. That way they are useable and available to everyone around the world. Hopefully this new standard of audio measurement and levelling will be used internationally and will help to make the future of media production and broadcasting a better one.

In my opinion R 128 will become a standard of international importance in future media technology.

13 BROADCASTING

The EBU's goal is to achieve a distribution and broadcast network with equal loudness levels. Every program in TV or radio should be equally loud. This would benefit the customers so they wouldn't have to constantly change the volume settings between different programs.

13.1 Objectives and basic principles

The goal according to the EBU is the achievement of “loudness normalization between services in the distribution stage of the radio and television broadcast chain and to achieve loudness level equalization between systems and interfaces of consumer equipment for radio and television reproduction.” The aim is to provide a consistent, higher quality of sound and thus a more pleasant listening experience for the audience.

This will only work when the whole production chain is being involved. Therefore the document “tech 3344” had been released. In this recommendation all the im-

¹⁹ Florian Camerer => <http://www.youtube.com/watch?v=iuEtQqC-Sqo> Accessed on 08.01.2013

portant technical guidelines are described. For further information about the technical guidelines see:

<https://tech.ebu.ch/docs/tech/tech3344.pdf>

14 CONCLUSION

The EBU R 128 standard will be important in the future audio technology and will change the way of work in many ways, which is why I wanted to learn more about with the topic.

In my field of studies I have discovered a whole new world that has been uncovered to me. My awareness has changed and I do now know more about how perception works. Like many other times the deeper you let yourself float in an area of knowledge the more you realize there is to know. There is a lot more to say about this topic, and there will be a lot more to learn about it in the future. That is why very important to learn about R 128. EBU R 128 will be a huge change in audio and broadcast technology. This was my motivation to write this paper. Furthermore I am sure that these changes will affect the way we produce media in the future as it also changes the way we consume it.

“A loud voice cannot compete with a clear voice, even if it’s a whisper”

Barry Neil Kaufman