Inter BEE 2010

International Broadcast Equipment Exhibition

CONTENT FORUM

Audio Symposium

"Can Audio Loudness Standard definitely Free Users from Remote Volume Control?"

Thursday, November 18th, International Conference Room, 2F,
International Conference Hall

Japan Electronics and Information Technology Industries Association

■Coordinator

The 2010 Acoustic Forum raised loudness monitoring issues.

Mr. Mick M Sawaguchi

Advisor, Pioneer Corpration,
Fellow AES / IBS

Conventional corrective tools to monitor sound levels changed from VU meters a long time ago to peak and digital meters, which monitor the size of pure electrical signals. In recent years, the establishment of international standards for “loudness” monitors which take into consideration psychoacoustic aspects has centered on the ITU-R (ITU-RBS-1770), which attempt to integrate levels suitable worldwide. This “loudness” based sound monitoring will bring about major changes, shifting from an era of volume wars to an era of appropriate sound levels.

In the broadcasting, music, and TV/radio commercial sectors, fierce volume competition, or a so-called “loudness war”, was created founded on a prevailing attitude that “loud is best”. The result was that users listened to extremely compressed and artificially produced sounds, and had to constantly use a remote control to correct volume differences that occurred at stations, programs and TV commercials etc. To solve such problems, the ITU-R is working to standardize “loudness monitoring meters” using auditory models, and based on this, loudness monitor production standards are set to be established primarily in Europe (EVBU), America (SMPTE/NAB) and Japan (Association of Radio Industries and Businesses (ARIB)).

These movements will be briefly introduced, and we will then discuss what kind of problems and possible merits these movements will bring to areas ranging from production, transmission, broadcasting and media to viewers.

After this session, loudness meter products made by various companies will be jointly displayed so please check out the workings of actual equipment yourself.

Porfile:

After graduating from the Faculty of Electronics Engineering, Chiba Institute of Technology in 1971, Mick Sawaguchi joined Japan Broadcasting Corporation (NHK) in the same year and was assigned to the Yamagata Bureau. He worked as a mixer for audio drama productions at the Program Production Engineering Department in NHK Broadcasting Center. After serving as Director of Program Production Engineering Department starting in 2003 and retiring from NHK in 2005, he joined Pioneer Corporation, working in audio promotion at the Technical Strategy Department, Research & Development Group. Currently, he conducts research activities, seminars, lectures and papers about the development of audio technologies.

Since April 2006, he has been in charge of sound design in the Musical Creativity and the Environment Course, Faculty of Music, Tokyo National University of Fine Arts & Music. He launched the jazz label UNAMAS in September 2007 as a tool for production activities.

His expertise is in sound design for drama productions. Since 1985, in particular, he has focused on designing studios and developing software for multi-channel surround-sound audio, setting a premise for the digital age. From 1987 on, he worked on an FM radio drama in Dolby Surround, and from 1992, was engaged in developing surround production software for 3-2 surround HD-TV drama series, while developing production guidelines to spread the technology and studying next-generation audio. More recently, he has overseen the planning and operation of the sound category of the Inter BEE International Symposium, and serves as a member of the judging panel for the mixing category of the JPPA-Awards. He is a board member of AES' Japan section as well as the JAS, in addition to serving as Vice Chair of the AES Technical Committee on Studio Practices & Production.

In 2002, he was awarded a fellowship from the AES for his contribution to surround-sound, and was similarly recognized by Europe’ s IBS in 2003. In 2004, he was recognized by the ABU for the Best Technical Paper of FY2004, and in 2005, to observe the 10th "Day of Sound," JAS declared Sawaguchi a "Master of Sound" for his many years of work on surround audio.

In 2009 , he was given the AES Japan Award for the activities of SURROUND TERAKOYA and in 2010 he co-published "Surround Handbook" from Tokyo University of the Arts.

Overseas

A Fellow of the Audio Engineering Society (AES), an international audio organization

Currently Vice Chair of the AES Technical Committee on Studio Practices & Production (TC SPAP)

A Fellow of the Institute of Broadcast Sound (IBS), a European organization for broadcast sound

■Coordinator

The 2010 Acoustic Forum raised loudness monitoring issues.

Mr. Toru Kamekawa
Professor,
Tokyo University of the Arts' Department
Musical Creativity and the Environment

Profile:

Mr. Kamekawa joined Japan Broadcasting Corporation (NHK) after graduating from the Kyushu Institute of Design's Department of Acoustical Design in 1983. He worked on program production (audio), which involved overseeing music program production such as NHK Symphony Orchestra concerts, as well as research into new recording production techniques including hi-vision 5.1 surround.

In October 2002, he took up the post of Assistant Professor (currently Associate Professor) at the Tokyo University of the Arts' Department of Music. He supervises research of audio and recording technology at the Department of Musical Creativity and the Environment and Graduate School Music Culture Program for Music and Audio Creativity.

He is a board member of the Audio Engineering Society (AES) Japan Section; and a member of the Acoustical Society of Japan, the Japanese Society for Music Perception and Cognition, the Japan Audio Society and the Japan Association of Professional Recording Studios.

His field of specialty is surround music recording. This includes a number of projects such as surround music mixing for television, film and games. He is also currently conducting research on space expression through surround sound recording, and ways to listen to music in public spaces.

In October 2009, he received a Board of Governor award from the AES head office for his activities on behalf of the AES Japan Section.

In April 2010, he was promoted to the professor of the university.

■Presenter

Development of the Loudness Meter and NHK Efforts

Mr. Mikihiko Okamoto

Senior Manager, Technical Operations & Engineering Center,
Technical Operations & Engineering Div.,
Broadcast Engineering Dept., Japan Broadcasting Corporation (NHK)

The word “loudness” has become a common term in recent times. “Loudness” refers to “the volume of auditory sound (sense of sound volume)”, and is the most basic characteristic of sound perception.

So, why is the word “loudness” attracting so much attention?

One reason for this is a desire to come up with a volume adjustment meter that better meets our auditory senses. There is a limit to adjusting volume just by using a VU (volume unit) meter to control volume when producing and transmitting programs, and it is not possible to resolve volume variations during a program.

Another reason is the three loudness meter-related international standards recommended by ITU-R (Radiocommunication Sector of the International Telecommunication Union (ITU)): “Levels for International Program Exchange”, “Meter Requirements”, and “Measurement Algorithms”. Due to this, a number of meter manufacturers have announced a succession of new products, and program production and exchange using loudness meters has gained momentum across the globe.

NHK developed real-time loudness level meters using auditory psychology in 1999, and has contributed to ITU-R loudness discussions since 2000. Loudness meters developed by NHK are used to control volume by people who are not sound specialists such as those working in video editing or preview rooms, but this has not led to them being used instead of VU meters by mixers on TV productions.

However, when TV broadcasts are completely digitalized in July next year, TV production sound levels will be delivered untouched straight to people’s homes, so greater controls of sound level will become crucial. Further, it is predicted that productions using loudness levels will become the standard when producing international programs such as Olympic broadcasts in the future. Therefore, it has become urgent for NHK to examine ways to manage loudness meters on TV productions.

This symposium will give an explanation of the ITU-R recommendations and the background behind the development of loudness meters, and introduce NHK efforts to utilize loudness meters for broadcasts.

Profile:

1986 Graduated from the Department of Electronic Engineering at the University of Electro-Communications. Joined NKH the same year, and took up a new post at the Yamaguchi broadcasting station.

1990 Involved in sound work focusing on TV and radio drama programs at the Technical Operations & Engineering Center of the Broadcast Engineering Department. Responsible for studio design including MA studio CD809 dealing with 5.1 sound and sound program development.

2004 Involved in high-presence acoustical research at the Science and Technology Research Laboratories. This research included 22.2ch audio, which is a super hi-vision audio system. Participating in the launch of the ARIB Studio Audio Operation team, he contributed to loudness meter deliberations at ITU-R.

2006 At the Nagoya Station Engineering Division (Operations and Engineering) worked on the expansion of regional broadcast sound production to improve regional services for terrestrial digital TV broadcasting.

2009 Responsible for period drama “Ryoma-den” in Technical Operations & Engineering Center, Technical Operations & Engineering Div., Broadcast Engineering Dept.

2010 Senior Manager of Technical Operations & Engineering Division (Sound). Committee member of the ARIB Studio Audio Workgroup committee.

■Presenter

National Association of Commercial Broadcasters in Japan, “TV Sound Level WG” Activities and Future Issues

Mr. Eiichi Matsunaga

Audio Management Division Manager,
Technical Production Center, Production Technology Department,
Fuji Television Network, Inc.

July 2011 marks the end of the terrestrial analog broadcasting service which we have come to know and love for the nearly 60 long years since its launch in 1953. At that time, all terrestrial and BS TV broadcasts will convert to digital broadcasts. Digital broadcasting has a number of advantages such as high image and sound quality which is clearer and ensures no more ghost images, and also enables the broadcasting of data. There are also disadvantages because, as a fundamental rule, sound is delivered without using volume control equipment which was used for analog broadcasts to maintain qualities such as high-quality sound. Therefore, volume variations that had caused problems up to now will be reflected unchanged in broadcasts.

The National Association of Commercial Broadcasters in Japan established the “TV Sound Level WG” in July, 2009 as a subsidiary organization of the “Engineering Committee”, and conducted various investigations over a year and a half to solve such problems. I will report the status of current activities, provisional schedule, and future issues.

Status of Current Activities

(1) Study to Formulate NAB Technical Standard T032 - “TV Broadcasting Sound Level Guidelines”.
(2) Production of “TV Sound Reference Source” to supplement technical standards.
(3) Production of “Basic Free Software to Measure Loudness”
(4) Correction of sound level gaps between broadcasting stations including NHK
(5) Grasp of national and international trends
(6) Collaboration between the Business Committee and associated organizations, and environment improvement

Future Issues

(1) Increased awareness of loudness management, and popularization of loudness meters
(2) Increased awareness of NAB Technical Standard T032, and establishment of operational procedures
(3) Solving problems after actual operation

Profile:

1979 Joined Fuji Television Network, Inc., and was assigned to the Broadcasting Division. He was in charge of master tapes and lines.

1981 He was transferred to the Production Technology Department, Audio Section and has been continuously involved in audio work since then. During that time, he was in charge of music programs such as “Music Fair” and “Our Music”, variety shows such as “Yuuyake Nyan Nyan” “Quiz Doremifa Don” and “Naruhodo! The World” and drama and a number of films.

1999 He was in charge of audio of “Shonen H” which received the excellence prize at the Arts Festival by the Agency for Cultural Affairs.

2002 He received the excellence award in the broadcasting division of the Japan Professional Music Recording Prize with the ballet program “Dragon. Quest”.

2009 2009 He became a chief examiner of "Television Audio Level WG" which is a branch of the Technical Committee of the National Assembly of Commercial Broadcasters.

He developed audio equipment in 1997 when the headquarters of Fuji Television was moved to Odaiba and in 2007 when the Wangan Studio was newly built. He has also been working as a lecturer at the audio technology training session of the national assembly of commercial broadcasters for more than 20 years.

■Presenter

How will sound production engineering be changed by "loudness" technology?

Mr. Hideo Irimajiri

Specialist Manager,TV Operation Engineering Dept.,
Mainichi Broadcasting System, Inc

At the end of September, 2010, the Commercial Advertisement Loudness Mitigation (CALM) bill relating to commercial sound volume regulations passed the Senate after passing the House of Representatives. Since detailed revisions were made, it seems that it will have to go to a revote before being passed, but it looks like being in the final stages of becoming law.

It would be no exaggeration to say that sound volume problems in Japan are at a critical level even on a global scale. Program volume variations extend to a maximum of nearly 20dB according to statistics, and when it comes to digital TV broadcasting sound, volume variations between programs and channels are expanding regardless of high performance, ironically making remote controls (volume adjustment) essential to watch TV.

Meanwhile, so-called "loudness measuring" engineering to coordinate TV program volumes has been developed and recommended. Program exchange by loudness values has dramatically improved volume disparities. In America, loudness recommendations by the ATSC have been enforced to correspond with legislation, and even in Europe, loudness recommendations were established in September.

In Japan, analog broadcasts will conclude in July, 2011, and be converted to digital broadcasts. Conversion to loudness operation may be the last chance for improvement. All organizations are conducting investigations into exchange standards for programs using loudness technology. Fast becoming a worldwide trend, we are right on the brink of managing sound levels using loudness technology.

While the operation of loudness technology is gaining momentum, what kind of work will be required from engineers working in the field? What kind of mixing methods will be needed in this coming era of loudness technology?

When thinking about loudness volume standards, regulating programs with excessive volume comes to mind, such as setting maximum permitted values for average loudness values for all TV programs. It will, therefore, be necessary to perform mixing by setting a value lower than the maximum permitted value. However, average loudness values are values fixed for when programs start and finish.

While loudness meters undoubtedly display volume levels from start to finish, they are cumulative meters so they usually stop responding when operating rapid faders. It is unlikely that mixing will rely only on loudness meters for such extremely sluggish movements.

For programs where MA work is performed, methods such as measuring loudness during the final playback and adjusting overall levels so they are kept within the permitted levels are available, but they cannot be applied to live broadcasts. In the end, it comes down to mixing with one eye on the loudness meter while relying on the familiar VU meter.

Profile:

1979 Gained degree at Kyushu Institute of Design, Department of Acoustic Design

1981 Graduated from graduate school of the same university. Involved in recording activities from student days, particularly exploring 4ch recording and spatial acoustics, becoming the origin of my own surround sound.

1981 Joined Mainichi Broadcasting System (MBS). Gaining experience via the Video Engineering Department, Sound Engineering Department, Hall Technology, Post-Production Department, and currently the Transmission Department, studied universal sound including various transmitted sound volume issues in broadcasting.

In the Sound Department, involved in projects such as the first Dolby surround high school baseball relay broadcast.

Outside committee appointments include serving on the ARIB Sound Workgroup committee, National Association of Commercial Broadcasters in Japan TV Sound Level Workgroup committee, and AES Japan Section board.

Therefore, today I will be considering whether loudness technology is actually “usable”.

① If we use the VU meter, is it possible to carry out mixing that meets standards?
② Are the short-term loudness meter and momentary loudness meters recommended by the EBU actually “usable”?

And, if we have time, I would like to discuss the following topics:

③ Who needs the loudness meter?
④ What are the common pitfalls that are easy to fall into when actually producing programs?

Lastly, the introduction of loudness technology is a good opportunity to review current production techniques.

Although we have principally worked with VU related equipment, I feel that VU meter swings have expanded on average to a 3VU level over the past 30 years or so since I entered this industry.

I would have to say that when a slight amount of level-over no longer became distorted due to advances in equipment, this ignited the volume war founded on the devilish idea that “the best noise is a loud noise”. This was a global phenomenon devised for mixing methods that cheat the VU meter. The purpose of the compressor was, in a word, to raise volume levels, and an effector that increases volumes without raising meter swings called a loudness maximizer even appeared on the scene.

However, this has resulted in a need to consider whether the world has truly been provided with good sound. Even if sound can be heard loudly and clearly, this does not necessarily mean that it is a good sound. I believe that the use of loudness meters is the first step to a universal design for sound. Universal design for broadcasting will become a crucial technology for Japan as it enters an era of having an "aged society."

■Presenter

～ CM Production and Post-Production Issues ～

Mr. Hiroyuki Murakoshi
Sound Engineer,
Gotanda Operation Group/JPPA/Surround CM Study Group/,
Post Production Dept. IMAGICA Corp.

Examination of the level of awareness, current status and problems relating to loudness issues faced by post-production sound engineers who produce sound delivered to broadcasting stations

Reporting results of questionnaires aimed at sound engineers in JPPA member companies.

Mixing techniques that correspond with new product standards

Porfile:

1986	Joined Onkio Haus. Gained sound production experience for TV and radio commercials, and many other video works.
1998	Joined Toyo Recording. Performed MA work centering on TV programming.
1999	Joined Imagica Dio. Assigned to the Tokyo Imaging Center. Involved in sound production for TV commercials, event videos, cinema ads, and various video works such as announcements for theaters. Gained wide experience from 16mm film monaural to 3D & Surround.
2007	Participated as sub-secretariat from the launch of the Commercial Sound Study Group. Awarded silver award for mixing engineering in the drama category of the JPPA Awards 2007.
2008	Wrote and published an article with Commercial Sound Study Group members entitled “TV Commercial Sound” in the November edition of the “Broadcast Engineering” journal. Awarded special award by the documentary category judges for mixing engineering at the JPPA Awards 2008.
2009	Participated as a panelist at AES Tokyo 2009 for WS2 “What Should be the Volume Controls in the Digital Broadcasting Era”, and WS3 “Exhaustive Study about 5.1 Commercial Sound! – Production Know-how and Future Issues”. Started by looking at broadcasting loudness issues and activities, followed by opinion-exchange with National Association of Commercial Broadcasters in Japan’ s TV Sound Level WG. Awarded Minister of Economy, Trade and Industry award, grand prix, and gold award for mixing engineering in the drama category of the JPPA Awards 2009. Participated in JPPA Audio workshop.

■Presenter

On the way to Loudness Nirvana
Audio Levelling with EBU R 128

Mr. Florian Camerer
Sound Engineer,
ORF – Austrian TV,
Vienna, Austria

Introduction

This article describes one of the most fundamental changes in the history of audio in broadcasting: the change of the levelling paradigm from peak normalisation to **loudness normalisation**. This change is vital because of the problem which 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 in programmes, between programmes and between channels. Loudness normalisation is the solution to counteract this problem (*see figure 1*).

Loudness normalisation is a true audio levelling revolution!

[Loudness refers to the perceived strength of a piece of audio (music, speech, sound effects...). The loudness depends on the level, frequency, content and the duration of the audio, amongst other things.]

EBU Recommendation R 128 [1] 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.

Four supporting documents have been created to aid the audio industry to work with the Recommendation.

R 128 is based entirely on open standards and aims to harmonise the way we produce and measure audio internationally. In addition to **'Programme Loudness'**, R 128 introduces two more audio descriptors, **'Loudness Range'** and **'Maximum True Peak Level'**. All three are designed to work together, forming a set of essential descriptors that characterise an audio signal.

Profile:

Florian Camerer joined the Austrian Broadcasting Corporation (ORF) in 1990. From very early on, he has been mixing small programs and working on location. In 1995 he became a staff-sound-engineer („Tonmeister") mainly in the field of production sound and post-production. High quality audio for documentaries developed into his field of special interest. In 1993 he started to get interested in Multichannel Audio. He mixed the first program of the ORF in Dolby Surround („Arctic Northeast") and is now involved with all aspects of Multichannel Audio at ORF. He is lecturing on an international basis especially in dramaturgical aspects of surround sound productions, microphones for surround sound and multichannel audio for HD.

Productions:

Several Dramas (also boom operator) („Eurocops", „Tatort")
Post-pro and location recording on numerous Wildlife-programs („Universum") and Adventure-programs („Land der Berge")
All sound for major ORF-series „Arctic Northeast – Franz-Joseph Land" (12 program-hours, partly Dolby Surround) 1993-96
Remix of several parts of Arctic Northeast in 5-channel Discrete
All sound for the program „Russia´s holy war" – a documentary about the orthodox church 1998 – Remix in 5-channel Discrete
Multichannel production for André Heller´s „Voices of God" – project in Marrakesh/Morocco
Major millenium documentary „A-Watch – Austria´s Nature-heritage"
Remix of several classical concerts in 5.0
Generic 5.1 mix of major reedit of "Arctic Northeast" ("The Ice Trap" and "Paradise Found") for TV and DVD
Successful promotion of ORF´s move to 5.1 satellite transmission with the New Year´s Concert 2003
DVD Musicedit and Audio-Design for the New Year' s Concert DVD since 2003

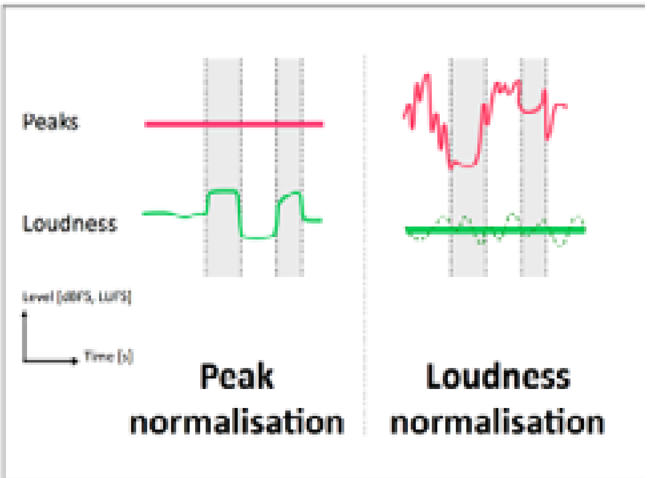


Figure 1: Peak level normalisation vs. Loudness level normalisation of a series of programmes

Loudness metering and **loudness normalisation** signify a ***true audio levelling revolution***.

Furthermore, this loudness-levelling paradigm affects ***all stages*** of an audio broadcast signal, from acquisition to distribution and transmission. Thus, the ultimate goal is to harmonise audio loudness levels within broadcast channels as well as between channels to achieve an **equal universal loudness level** for the benefit of the listeners. To be clear: the loudness level can (and should!) still vary according to artistic and technical needs within a programme. Rather, the loudness normalisation method uses the **average loudness** of a programme, to make the level between programmes and between channels consistent.

Experience of several EBU members has shown that working with the loudness paradigm is both liberating and satisfactory. The fight for "Who is the loudest" disappears, mixes can be more dynamic, there are fewer dynamic compression artefacts such as pumping and thus there is an **overall increase of audio quality!** Programme makers who favoured dynamic mixes in the past are now relieved from potential compromises because their programme no longer sounds softer than more-compressed ones. With loudness normalisation, this compromise is gone. Nirvana is closer than ever!

The origin of the problem – the "Loudness War"

Audio metering in broadcasting today is based typically on quasi-peak programme meters (QPPMs – more usually known as just PPMs). It is 'quasi' because of its finite reaction time of 10ms (although 5ms also is also found). In practice, this means that signal peaks shorter than this reaction time won't be displayed correctly, if at all (for example, transients such as those created when keys are jangled). In order to provide headroom for these transients, which one wouldn't see on the meter, but which should nevertheless be there so as to contribute to the "openness" of the audio signal, the agreed *Permitted Maximum Level* (PML) was set at **-9 dBFS**. This value was based on the familiar – and in many places, still extant – method of delivering sound to the home on an FM carrier. The carrier's maximum deviation for TV was standardised in many countries at 50 kHz and the PML at 30 kHz deviation (equating to -9 dBFS), which thus allowed 20 kHz, or 4.4 dB of **headroom**.

However, commercial pressures have taken a hold and the response to the pressure to stand out from the competition has been to be louder than it. Modern peak metering and powerful dynamic range processing allowed organisations to realign the PML to equate to the maximum deviation (50 kHz) of the FM carrier (*see figure 2*). All the transients have to be chopped at the PML to avoid distortion, but that has been thought an acceptable compromise by those who have implemented it.

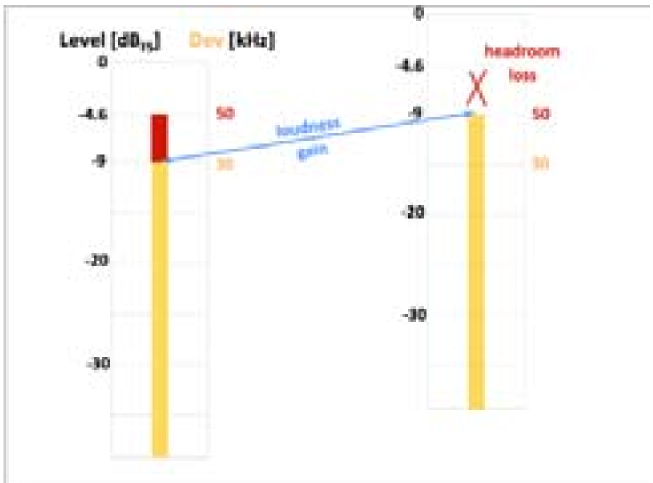


Figure 2: Relationship between Permitted Maximum Level (with QPPM) and FM Deviation and the abuse of it: gaining loudness, but losing headroom

When someone at home switches between one of these stations and a station which has not joined the "war" or is taken to a loud advertisement in a well-balanced programme with a wider dynamic range, the audio level jumps and there is a grab for the remote control to adjust the sound to a more acceptable level. In the case of the loud advertisements, the sound then has to be re-adjusted when the main programme returns. It is no wonder that so many complaints are received by the broadcasters. Other people solve the problem by muting the sound during the advertisements and so the message is at least greatly diluted.

A Standard Emerges – and the EBU develops it

The International Telecommunications Union (ITU) recognised the problem and its work gave rise to **ITU-R BS.1770** [2]. The purpose of that standard was to establish an agreed algorithm for the measurement of loudness and the true peak levels of programmes. It is a robust standard which has the benefit of a simple implementation. In brief, it defines a "**K-weighting**" curve (a modified second-order high-pass filter, *see figure 3*) which forms the basis for matching an inherently subjective impression with an objective measurement. This weighting curve is applied to all the channels (except the Low-Frequency Effects Channel (LFE) which is discarded from the measurement), the total mean square energy is calculated (with different gain factors for the front and surround channels; *see figure 4*) and the result is displayed as "LKFS" (Loudness, K-Weighting, referenced to digital Full Scale). For relative measurements, Loudness Units (LU) is used, where 1 LU is equivalent to 1 dB.

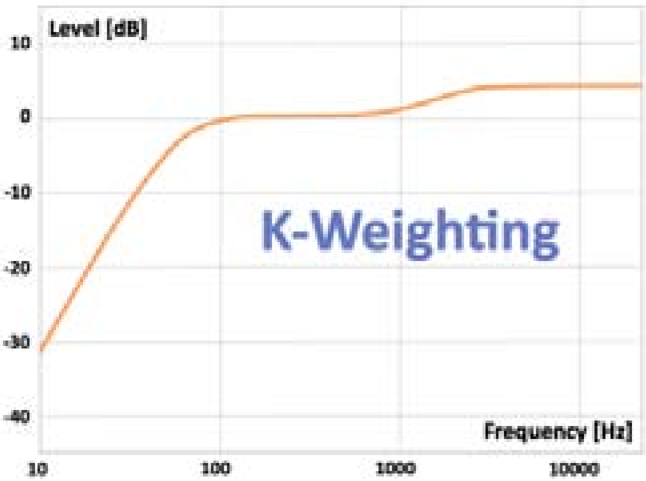
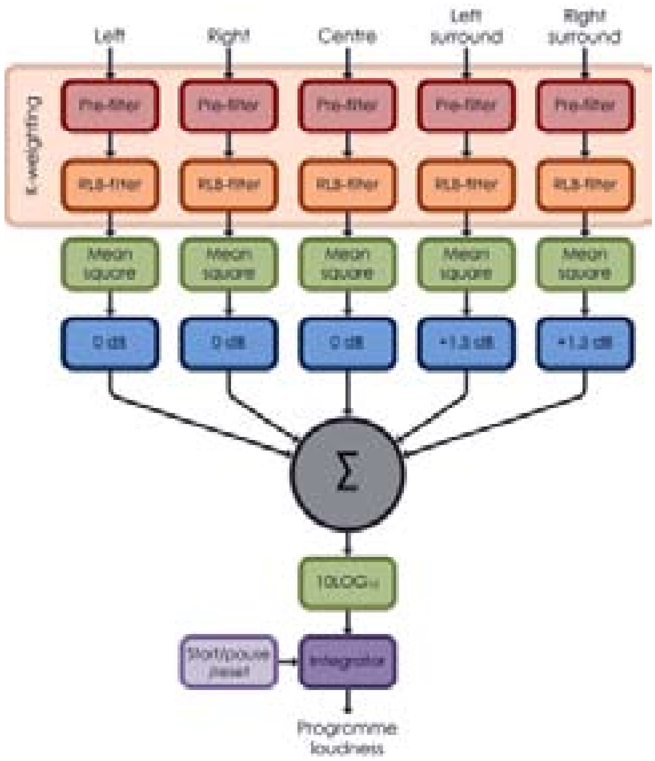


Figure 3: 'K-Weighting' as the basis of loudness measurement in ITU-R BS.1770



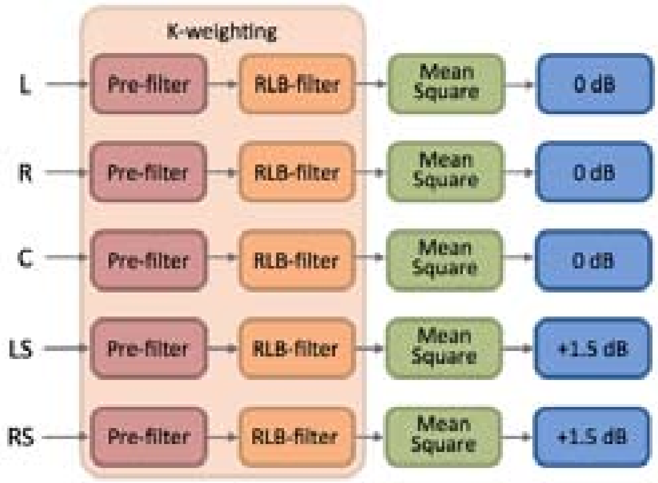


Figure 4: Channel processing and summation in ITU-R BS.1770

A more detailed study of the algorithm can be found in ITU-R BS.1770 [2] as well as EBU Tech Doc 3343 (*'Practical Guidelines'* [3]).

BS.1770 also defines and recommends the use of a true peak meter for measuring peaks Such a meter runs at a higher sampling rate than the audio signal (usually 4x oversampling) to catch inter-sample peaks which might otherwise exceed 0 dBFS and thus cause distortion later in the chain.

BS.1770 provides the basis for the EBU's Recommendation **R 128**. It extends it by actually defining a specific '**Target Level**' (*see below*) for loudness normalisation as well as a **gating** method to improve the loudness matching of programmes which contain longer periods of silence or isolated utterances. The EBU's development was needed to accommodate the needs of programme makers, with particular regard to having a means to measure complete mixes (rather than just one component, such as the dialogue) and the loudness range of the programme. To do this, the EBU has specified three new parameters:

- **Programme Loudness**
- **Loudness Range**
- **True Peak Level**

EBU R128 and ITU-R BS.1770 are the basis. Four more EBU Technical Documents provide details.

Programme Loudness

Programme Loudness describes the long-term integrated loudness over the duration of a programme. (In R 128, the definition of the word 'programme' is also used to refer to advertisements, bumpers and other interstitials) The parameter consists of one number (in LUFS[1], with one number after the decimal point) which indicates "how loud the programme is on average". This is measured with a meter compliant to ITU-R BS.1770 with the addition of a gating function. The gate serves to pause the loudness measurement when the signal drops below a certain threshold. Without this gating function, programmes with longer periods of silence or low-level background sounds or noise will get too low an integrated loudness level value. Such programmes would subsequently be too loud when broadcast.

Following a series of tests, a gate of -8 dB relative to the ungated LUFS measurement with a block length of 400 ms was agreed. The tests also confirmed, along with other findings, the choice of the **Target Level** to which every audio signal will be normalised; it is:

-23 LUFS (-8 rel gate)

-23 LUFS is the new centre of the audio levelling universe !!!

A deviation of ± **1 LU** is acceptable for programmes where an exact normalisation to the Target Level of -23 LUFS is not achievable practically (such as live programmes or ones which have an exceedingly short turn-round). In cases where the levels of a programme's individual signals are to a large extent *unpredictable*, or where a programme consists of only background elements (for example, the music bed for a weather programme) this tolerance may be too tight.

It is therefore anticipated for such cases that the integrated loudness level may lie outside the tolerance specified by R 128.

'LUFS' indicates the value of K-weighted loudness with reference to digital full scale. The EBU recommends this unit to overcome an inconsistency between ITU-R BS.1770 and ITU-R BS.1771. This unit complies with ISO 80000-8.

Loudness Range

Another major consideration was the loudness range which would be needed to accommodate *all* programmes (provided that they don't exceed the tolerable loudness range for domestic listening). The *Loudness Range* (LRA) descriptor quantifies (in LU) the variation of the loudness measurement of a programme. It is based on the statistical distribution of loudness within a programme, thereby excluding the extremes. Therefore, for example, a single gunshot is not able to bias the outcome of the LRA computation.

EBU Recommendation R 128 does not specify a maximum permitted LRA, as it is dependent on factors such as the tolerance window of the average listener to the station, the distribution of genres of the station etc. R 128 does, however, strongly **encourage the use of LRA** to determine if dynamic treatment of an audio signal is needed and to match the signal with the requirements of a particular transmission channel or platform. More details about LRA may be found in EBU Tech Doc 3342 [4].

Loudness Range is a generic descriptor that helps to decide if dynamic compression is necessary.

First experiences at broadcasting stations suggest a maximum LRA value of approximately **20 LU** for highly dynamic material, such as action movies or classical music. The majority of programming will never need to fully use such a high LRA value or, indeed, be able to reach it!

True Peak Level

The True Peak Level of an audio signal indicates the maximum (positive or negative) value of the signal waveform in the continuous time domain; this value is in most cases higher than that shown by a quasi-peak meter or even a sample-peak meter, both of which would miss the true peaks which potentially lie between samples. The use of an oversampling meter, compliant to BS.1770, allows those peaks to be detected.

Oversampling Peak Meters provide a good estimate for the true peak of an audio signal. Sample Peak Meters don't.

An oversampling meter may still slightly under-read the actual peak value (depending on the oversampling frequency) and so the *Maximum Permitted True Peak Level* for Production is thus:

-1 dBTP

Note that some parts of the chain, such as analogue re-broadcasters and users of low bit-rate coders require a lower True Peak Level. The PLOUD Distribution Guidelines (EBU Tech Doc 3344 [5]) contains comprehensive coverage of the topic.

Strategies for loudness normalisation

Figure 5 shows two approaches mainly for **production**; the first is more relevant for the early stages of the transition and it is perhaps especially useful to those who work on live programmes. The existing meters, limiters and mixing practices are retained and a level shift is done at the output of the console (after the main meters) to achieve the loudness Target Level of **-23 LUFS**. A loudness meter is placed after the level shift to enable the engineers to understand the exact amount of shift (which initially is still a bit of guesswork). Using a loudness meter on past programmes of the same genre gives good guidance as to where the levels sit. Early experience at NDR, ORF, and RTBF has shown that it is certainly possible for live mixes to fall into the ±1 LU window permitted by R 128.

Those who work with files have an easier task as the whole programme may be normalised to **-23 LUFS** by means of a level shift quickly and easily.

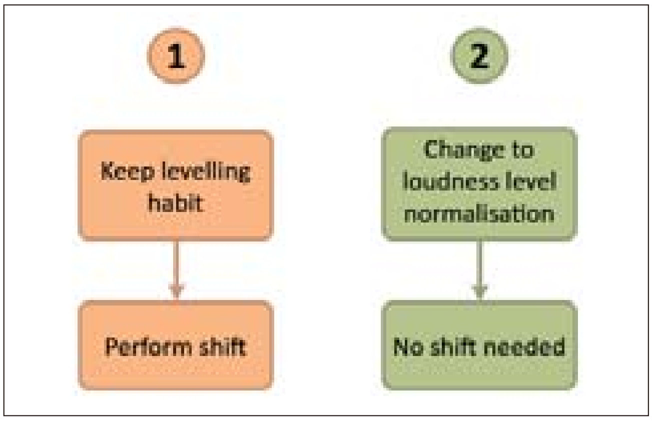


Figure 5: Two principal working methods to achieve uniform loudness in production and

Those organisations or sections which can make the move to loudness metering straight away can be expected to reap the benefit straight away. The greater dynamic range possible will be a welcome bonus for crowd noise in sports programmes, for example, enhancing the impact of a game for the viewers and listeners.

EBU Tech Doc 3343 [3] gives practical guidelines for the new way of working with audio levels.

There are basically also two ways to achieve loudness normalisation for the **consumer**: one is the actual **normalisation of the source** itself, so that the programmes are equally loud by design – the other method is with the use of **loudness metadata** that describes how loud a programme is. For the latter, the actual average programme loudness levels don't need to be changed to a normalised value and can still vary a lot. For those with up-to-date equipment, the normalisation can be performed at the consumer's end using the individual loudness metadata values to gain-range the programmes to the same replay level.

Equal loudness can be achieved by normalising the source or by using loudness metadata.

Within the EBU R 128 loudness levelling paradigm the first solution is encouraged due to the following advantages:

- simplicity
- potential quality gain at the source.

Once levels are set, the audio engineer can switch to **mixing only by ear**. Watching the Momentary or Short-term Loudness and an occasional glance at the Integrated Loudness value should give confirmation that the mix is within the tolerance allowed around the Target Level. With a numerical readout of the I-value with one decimal point precision or a graphical display of similar resolution, trends can be anticipated, and the appropriate countermeasures taken.

To summarise: improving audio metering by replacing the PPM with the loudness meter is a step closer to the best measurement tool: the human ear.

Loudness in the Distribution Chain

EBU Tech Doc 3344 [5] specifies the relevant settings and processing of the audio after it has left the broadcast centre and takes the loudness paradigm all the way to the consumers' equipment, including set-top boxes and Audio-Video Receivers. In doing so, it also encourages non-compliant broadcasts to become compatible with EBU R 128. Digital transmissions, analogue transmissions, rebroadcast transmissions, locally-inserted advertisements, newly-added services and more are covered in this comprehensive document.

Rather than measuring the loudness of individual programmes, the distribution company monitors a service over 24 hours, with special care being taken when considering switched or shared services. As well as the programmes, any loudness metadata provided with the digital services which carry it is monitored so that the actual loudness level of the service may be compared with the stated level, both of which should be -23 LUFS of course! Once a day, the data is analysed and where a service is more than 0.9 LU (the exact value and method is still under discussion) from its Target Level, a steering unit applies a correction factor so that the long-term average of the audio remains at the Target Level ±1 LU.

*'**M**' and '**S**' are commonly used in stereophony for 'Mid' and 'Side' . To distinguish the integration times 'Momentary' and 'Short-term' , the versions '$\mathbf{ML_K}$' and '$\mathbf{SL_K}$' (as well as '$\mathbf{IL_K}$') may be used. '$\mathbf{L_K}$' stands for 'Level, K-weighted' , and complies with the international naming standard ISO 80000-8.*

Business Consequences

Because the loudness-levelling paradigm affects all stages of an audio broadcast signal, from acquisition to distribution and transmission and because the ultimate goal is to harmonise audio loudness levels within broadcast channels as well as between channels to achieve an equal universal loudness level for the benefit of the listeners, ***all audio professionals and audio-metering equipment in all parts of the chain will be affected by this change.***

For many, a major question will be whether all the existing QPPM meters will have to be replaced and whether all the relevant personnel will need instruction to accommodate the new way of working. The answer is "In the long run, undoubtedly, yes", but this transition needn't happen all at once, although at least some loudness meters should be brought into service at the earliest opportunity, to sit alongside the QPPMs. The replacement of meters can happen in line with the technical refresh cycles of other equipment, refurbishing of production facilities or as separate step-by-step projects (whatever is feasible on a reasonably short time scale), and people can be trained at the appropriate time.

Those responsible for the purchase of equipment should also be aware that safety limiters to prevent over-modulation will have to be able to work in *true-peak* mode and they will need to be adjusted to the appropriate maximum permitted true peak level, in production as well as at the output of master control, at the distribution head-end and at the transmitter site.

Conclusion

EBU R 128 and the four supporting documents provide a means to end the "Loudness War" – at last! The use of audio dynamics becomes a creative art again. There are still things to learn and people will take a while to get used to the new ways of working, but it will be worth the effort. Over 240 participants have joined the EBU PLOUD group (status in August 2010), the E-Mail reflectors have shown a level of activity never seen before and meter manufacturers have prepared units for display at IBC 2010 before the specification was even published. It is now time to put R 128 into action!

References

[1] EBU Technical Recommendation R 128 *'Loudness normalisation and permitted maximum level of audio signals'*

[2] ITU-R BS.1770 *'Algorithms to measure audio programme loudness and true-peak audio level'*

[3] EBU Tech Doc 3343 *'Practical Guidelines for Production and Implementa tion in accordance with EBU Technical Recommendation R 128'*

[4] EBU Tech Doc 3342 *'Loudness Range: A descriptor to supplement loudness normalisation in accordance with EBU R 128'*

[5] EBU Tech Doc 3344 *'Practical Guidelines for Distribution of Programmes in accordance with EBU R 128'*

[6] EBU Tech Doc 3341 *'Loudness Metering: 'EBU Mode' metering to supplement loudness normalisation in accordance with EBU R 128'*

The second solution is not forbidden (*see also the Distribution Guidelines document – EBU Tech Doc* 3344 [5]), but having one single number (**-23 LUFS**) has great strength in spreading the loudness-levelling concept as it is easy to understand and act upon. And the active normalisation of the source in a way 'punishes' overcompressed signals and thus automatically encourages production people to think about other, more dynamic and creative ways to make in impact with their programme. In other words, the actual technical change of the source level through active normalisation to -23 LUFS has direct consequences on the artistic process – and in a positive way!

Nevertheless, it must be stated that both methods can complement each other, they are not to be seen as opponents or a black-and-white view of the same issue. Both approaches are a part of R 128 – but because of the advantages listed above, the **normalisation of the source** is recommended.

Loudness normalisation of the source is recommended in production because of simplicity and potential quality gain.

Working with loudness meters

Until now, mixing with QPPMs and normalising the peaks has been done with reference to a permitted maximum level (typically, -9 dBFS) and a peak limiter at that level provided a "safety ceiling", which could be hit as hard as desired – at the expense of less engaging sound, naturally.

In contrast, the loudness-levelling paradigm more resembles "floating in space" as the schematic representation of a bar-graph meter in *figure 6* shows. *Figure 7* shows how a software meter based on a "needle" could look. The EBU has deliberately not specified any graphical or user-interface details of a loudness meter, but it has specified enhancements of the algorithm described in ITU-R BS.1770 and two scales: "**EBU +9 Scale**" which ought to be suitable for most programmes and "**EBU +18 Scale**" which may be used for programmes with a wide LRA. Both scales can either display the relative Loudness Level in **LU**, or the absolute one in **LUFS**. The meter manufacturers in the PLOUD Group have agreed to implement the "EBU mode" set of parameters, to make sure their meters' readings will be aligned. Many more manufacturers have adopted "EBU mode" too, or are in the process of doing so.

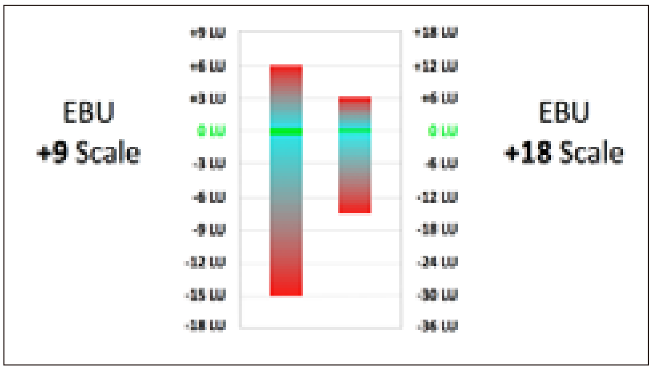


Figure 6: A schematic representation of the two loudness scales (here in LU) as described in by EBU Tech 3341

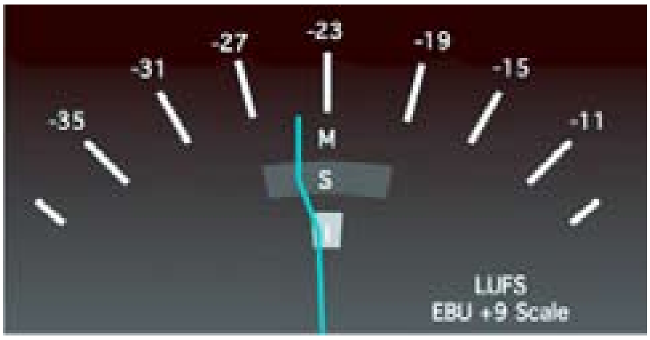


Figure 7: A schematic representation of an emulated loudness meter with a "bendy needle"

An 'EBU Mode' loudness meter as defined in EBU Tech Doc 3341 [6] offers 3 distinct time scales:

- **Momentary** Loudness (abbreviated "**M**") – time window: **400 ms**
- **Short-term** Loudness (abbreviated "**S**") – time window: **3 s**
- **Integrated** Loudness (abbreviated "**I**") – from '**start' to 'stop**'

The **M** and **S** time windows[2] are to be used for the immediate levelling and mixing of audio signals. Initial level setting may be performed best with the Momentary Loudness Meter, adjusting the level of key or anchor elements (such as voice, music or sound effects) to be around the Target Level of **-23 LUFS**.

It is advisable to set levels with a bit of caution initially, as it is easier to gradually increase the integrated loudness level during a mix than to decrease it. Usually, a slight increase in the course of a programme is also more natural – and an initially "defensive" strategy leaves the engineer room to manoeuvre in case of unexpected or unpredictable signals and events.