EBU

OPERATING EUROVISION AND EURORADIO

Tech 3401

GUIDELINES FOR RADIO PRODUCTION AND DISTRIBUTION IN ACCORDANCE WITH EBU R 128

SUPPLEMENTARY INFORMATION FOR R 128

Geneva
June 2021

This page and others in the document are intentionally left blank to maintain pagination for two-sided printing.

Contents

Guidelines for Radio production and distribution in accordance with EBU R 128

EBU Committee	*First Issued*	*Revised*	*Re-issued*
TC	2021		

Keywords: Loudness, normalisation, Radio production & distribution, metering, FM Radio, Internet streaming, Digital Audio Broadcasting (DAB), Media Player.

1. Introduction, key concept

EBU R 128 [1] recommends the shift from peak normalisation to **loudness normalisation**, with the average loudness level of a programme being **-23 LUFS** (this is called *Programme Loudness*). EBU R 128 also encourages the use of programme dynamics creatively, eliminating any advantage of compression as a weapon to be louder than the competition. This game-changing paradigm was first adopted by the TV community. In the Radio realm, the uptake has been slower due to reasons like a more heterogenous distribution systems landscape, technicians and journalists alike producing and mixing programmes, the "loudness war" in popular music and the lack of loudness functionality in Radio Content Management Systems.

The goal of loudness-based levelling is to provide the **best possible listening experience**. Easy-to-use loudness meters that "show what you hear" are the basis for production. Consequently, workflows can be level-optimised providing a new quality experience for all involved – the broadcasters and the audience.

In addition to R 128, the EBU PLOUD group has published seven other documents:

- **EBU R 128 s1** *'Loudness parameters for short-form content (adverts, promos etc.)', Supplement 1 to EBU R 128* [2];
- **EBU R 128 s2** *'Loudness in Streaming', Supplement 2 to EBU R 128* [3];
- **EBU R 128 s3** *'Loudness in Radio', Supplement 3 to EBU R 128* [4];
- **EBU Tech Doc 3341** *'Loudness Metering: 'EBU Mode' metering to supplement loudness normalisation in accordance with EBU R 128'* [5];
- **EBU Tech Doc 3342** *'Loudness Range: A descriptor to supplement loudness normalisation in accordance with EBU R 128'* [6];
- **EBU Tech Doc 3343** *'Guidelines for Production of Programmes in accordance with EBU R 128'* [7] and,
- **EBU Tech Doc 3344** *'Guidelines for Distribution and Reproduction in accordance with EBU R 128'* [8]

The present document is linked to **R 128 Supplement 3** (*'Loudness in Radio'*) [4] and joins the above suite of loudness publications. Based on EBU Tech 3343 and Tech 3344 and drawing on the practical experience of EBU Members, it introduces guidelines for adopting loudness normalisation and loudness-based level control in Radio production and Radio distribution workflows.

These guidelines will help Radio broadcasters to have more control over the loudness and dynamics of a programme to achieve the desired sound and increase signal quality, speech intelligibility and audience satisfaction. Loudness normalisation is also a key aspect for automated workflows in production and distribution for on-demand content to the Internet as well as on linear platforms.

The **key concept** is the clear distinction between the Radio *production* side and the *distribution* side.

Loudness in Radio is facilitated by a clear distinction between production and distribution.

Despite different workflow details and programme genres, there is no general difference between loudness normalisation in Radio production and TV production. Thus, the same principles apply. Loudness measurements are performed according to **ITU-R BS.1770** [9], on the complete signal.

For Loudness normalisation in *production*, the same principles apply for Radio and TV.

A common Loudness Level of −23 LUFS enables straight forward exchange of content between Radio, TV and Online in the production stage (see Figure 1).

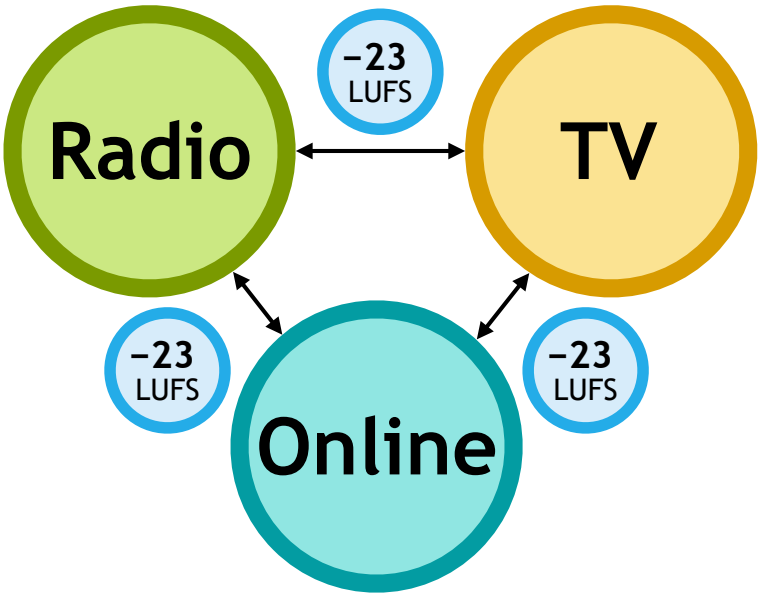

Figure 1: Programme exchange between different platforms with content at −23 LUFS

1.1 Document structure

§ 2 describes the Radio **infrastructure** and different workflows. Typical characteristics of **Production** and **Distribution** are listed;

§ 3 offers **advice** on what to consider before moving to a loudness workflow;

§ 4 gives **directives** how to adopt loudness in Radio **Production** workflows and encourages an "EBU R 128 technical culture". § 4.2 outlines the Graphical User Interface (GUI) of a simple **loudness meter** with the goal of easy visual feedback and universal appeal;

§ 5 gives **directives** for loudness in **Distribution**;

§ 6 looks at potential **future developments** in loudness for Radio broadcasting;

§ 7 presents **Use cases** from EBU members having applied EBU R 128 in Radio. These outline the successes of the transition to the loudness paradigm as well as the challenges and the compromises currently required in the daily operational work.

1.2 Definition of "programme"

For clarification purposes, the hierarchical structure of a "programme" is given based on the definition in EBU R 128. Therein, a **programme** is *"an individual, self-contained audiovisual or audio-only item to be presented in Radio, Television or other electronic media. An advertisement (commercial), trailer, promotional item ('promo'), interstitial or similar item ("Short-form Content") shall be considered to be a programme in this context"*. **Figure 2** shows the structure and the terms used with the example of a Radio morning show. A *Programme* consists of at least one *Segment* which in turn has at least one *Component* or *Element* (such as voice, music, background and sound effects). If a *Segment* is self-contained, it can be a programme too. The top-hierarchy *Programme* can thus contain other programmes.

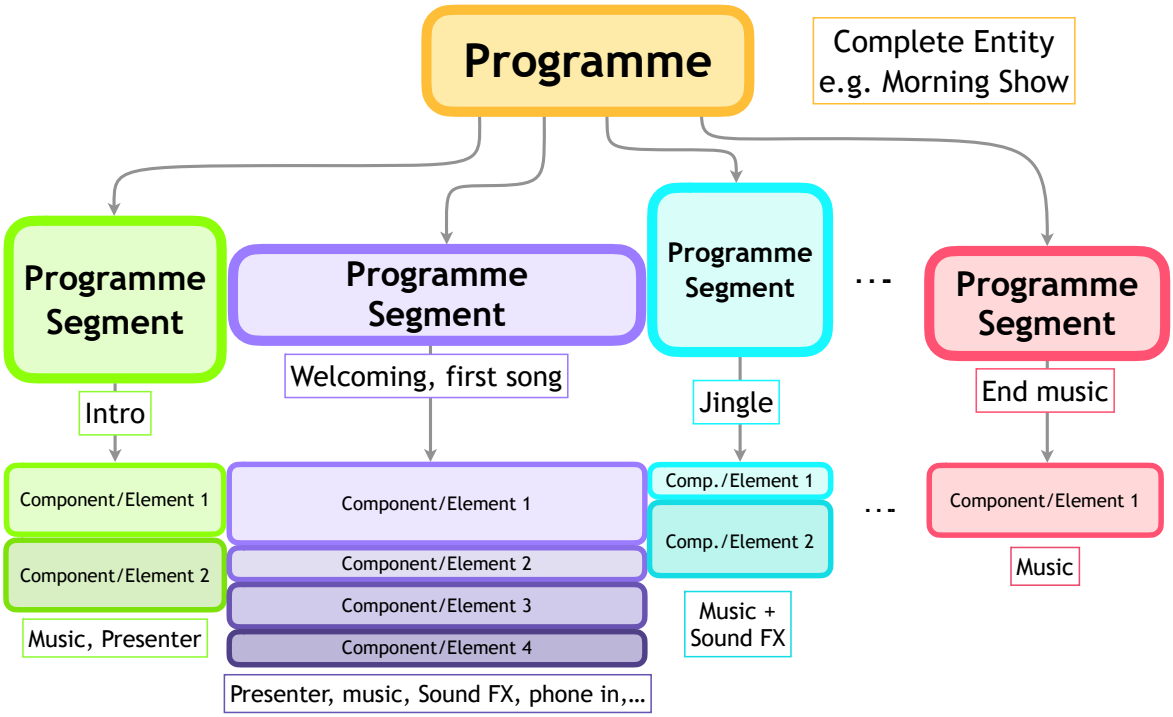

Figure 2: Definition of a Programme with its Segments and Components (or Elements)

1.3 Programme Loudness

The reader is advised to check EBU Tech 3343 for a detailed description of mixing a programme in the loudness world and its differences to peak normalisation. **Figure 3** illustrates the main concept, with Programme Loudness being the **average loudness**, measured over the whole duration of the programme. At any point during the programme, the current loudness (blue line) may deviate significantly (being higher or lower) from this average (orange line), resulting in a dynamic, open sound. The principal measuring device to achieve this goal are the **ears** and the **brain**.

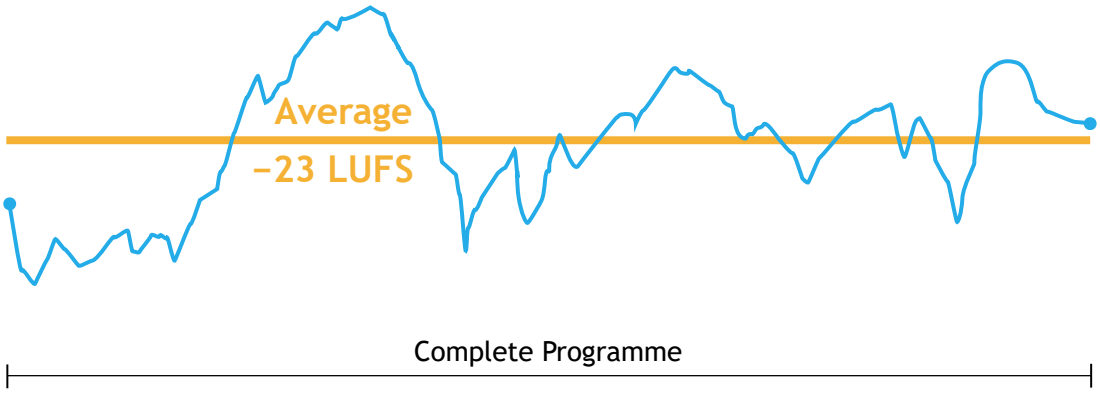

Figure 3: Programme Loudness of −23 LUFS, the average of the whole programme

2. Production and Distribution characteristics and workflows

When adopting EBU R 128 in Radio, it is recommended that the audio workflow be considered in *two separate stages*, **Production** and **Distribution** (see **Figure 4**).

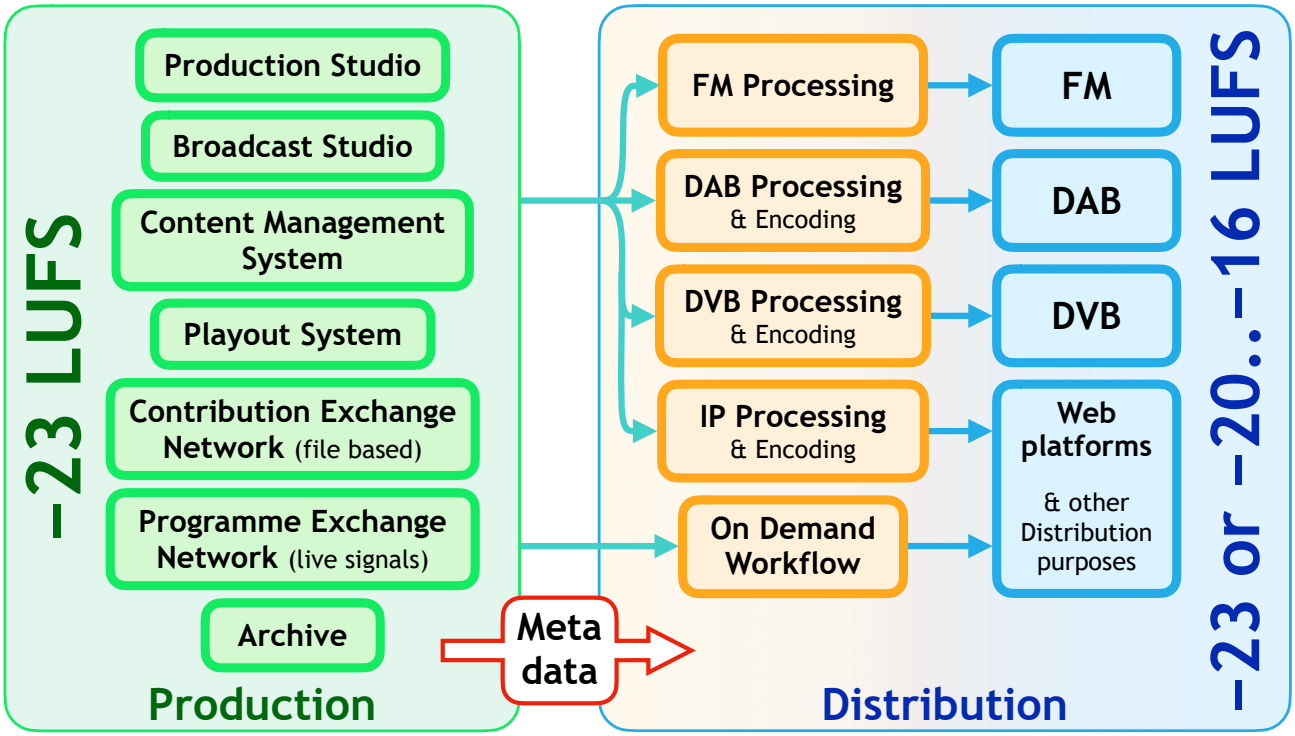

Figure 4: Two-stage structure of a Radio station workflow

During ***production*** the aim is to craft a generic, accurately controlled Radio programme with open dynamics, agnostic of the distribution channel. If signal compression is used, it is applied due to artistic reasons, fitting the content. The Peak-to-Loudness Ratio (PLR), a measure of micro-dynamics, is appropriate for the content and *not* restricted in advance for a certain distribution platform.

During ***distribution*** the programme may be adapted according to the specific requirements of different distribution channels. This includes the choice of a *Distribution Loudness Level* (see § 5) as well as dynamic processing to comply with specifications such as legal requirements for FM or different loudness targets for Web-platforms.

Frequently, ***Metadata*** links production and distribution, supporting, for example, automated workflows or audio processing in the consumer device.

3. The Loudness transition

There are a few prerequisites to facilitate the transition from legacy peak to loudness normalisation:

- Discuss the move towards loudness normalisation with the station's management. The switch to the loudness paradigm is a strategic decision that must be supported top-down;
- Measure Integrated Loudness (IL_K) and Loudness Range (LRA) at the output of the broadcasting studios and the output of the distribution stage to get a general idea of the current situation regarding the loudness level and the dynamic properties of the content;
- Measure IL_K and LRA of pre-produced programmes and programme segments;
- Identify the main areas where levelling is performed. Analyse the gain structure and check if simplifications are possible using a single loudness level for the whole production chain;
- Calculate the demand for loudness meters and processing devices (True-peak limiter, Loudness processor);
- Start loudness training for key production personnel which can then spread the word and disseminate the knowledge. Radio journalists, DJ Radio operators and other non-technicians should be trained right after the major equipment decisions have been taken and the technical transition has just been performed;
- Analyse your ingest workflows, check the loudness measurement and processing capabilities of your content database. If possible, expand its capabilities by adding loudness measurement and normalisation to the workflow. Process programme segments after adding them to the playlist, before they are played out;
- A best practice is to start introducing loudness normalisation with a single station at a single location, for example, a news station or a classical music station. The most noticeable improvements can be found in channels with subtle FM processing;
- The switch to Loudness normalisation should be a win-win-win situation: for the technicians, the content producers and of course for the audience!

4. Production

Beyond the generic mixing and production techniques described in EBU Tech 3343, there are many details to be addressed on a daily hands-on basis. Radio contains specific topics such as self-operated producing and broadcasting, a particular speech-music balance depending on the genre, automated workflows and especially dynamic processing for diverse reproduction environments (for example, car, smartphone, kitchen radio etc.) as Radio explicitly addresses mobile listeners as well.

4.1 Production principles and directives

With loudness normalising according to EBU R 128, unintended loudness jumps can be avoided between programmes. For higher intra-programme loudness consistency, to facilitate content production by non-technicians and to achieve a higher audio quality in distribution, **individual segments within a programme should generally be loudness normalised**. Nevertheless, the *dynamic properties* of content should be left intact during the production stage. Loudness normalisation breaks the vicious circle of hypercompressing content to beat the competition. Introducing loudness for Radio thus enhances the overall audio quality of the station.

With loudness normalisation dynamics is encouraged, compression loses.

The basis for manoeuvring in the loudness world is the **international loudness measurement standard** defined in **ITU-R BS.1770**. Every loudness meter used in a station must comply to this standard. Manufacturers of audio editing equipment and Content Management Systems are requested to implement the necessary loudness tools, such as automatic clip-based loudness normalisation.

EBU R 128 is the basis for the loudness normalisation of programmes.

Based on **EBU R 128**, the following **production directives** lead to a seamless switch to loudness normalisation in Radio:

- Produce programmes at a **Target Loudness Level of −23 LUFS** to encourage dynamics and foster audio quality. Programme exchange both in-house and with external partners is consequently much easier. Annoying loudness jumps between different contributions are avoided;
- **Normalise Programme Segments during production** instead of compensating an unbalanced mix with aggressive dynamic processing during *distribution*;
- Introduce **loudness measurement and metering** in all production and playout studios in accordance with EBU Tech 3341 and Tech 3343;
- Level **Compression** should be used as an artistic tool and not as an excessively used weapon in the loudness war;
- There is no technical reason to compromise the **dynamic properties** of a programme during production. The two main parameters to judge this are the *Peak-to-Loudness Ratio* (PLR) and *Loudness Range* (LRA) – see the following two bullet points;
- The **Peak-to-Loudness Ratio** is the difference between the Loudness Level and the True-Peak Level of the audio signal. It is a measure of micro-dynamics and thus of how much compression has been applied. Keep PLR at healthy levels and manipulate it only at the distribution stage to adapt to a specific distribution path if needed;

- Use the parameter **Loudness Range** (LRA) as a long-term **dynamics indicator** in **production**. It is also useful to check the robustness of the audio signal chain and can reveal unwanted dynamic processing. LRA provides comprehensive values for speech, also facilitating the judgement of its intelligibility. LRA may additionally help to steer dynamics processing for different Radio distribution platforms;
- Analyse the voice processing chain and adapt it to an average loudness value of **−23 LUFS (0 LU** on the relative scale) in self-op broadcasting and self-recording studios. Anticipate the consequences and adjust, for example, mix-minus or return signal levels accordingly;
- Loudness-normalise raw original sounds and/or studio recordings *only after* unwanted loudness inconsistencies and/or pauses are compensated or removed;
- For music, consider loudness normalisation either by processing the audio file itself or by using playback metadata;
- For *music with exceptionally high dynamics* (high Loudness Range and/or high Peak-to-Loudness Ratio) or with intentionally lower loudness, special solutions should be considered (for example, high quality dynamics processing, metadata to ensure a deliberately lower loudness level on playback);
- If the same content is used multiple times on different stations or platforms, keep the programme dynamics generic and appropriate for the genre during production. Apply the final processing individually at playout with regards to the specific station or platform;
- Continuously train technicians, editors, sound designers and journalists and communicate the advantages and simplifications that result from adopting EBU R 128 in Radio production.

Producing programmes at −23 LUFS encourages dynamics and facilitates content exchange.

4.2 Loudness Metering

Whereas ITU-R BS.1770 defines the loudness measurement algorithm, metering requirements are defined in EBU Tech 3341, introducing "EBU Mode". It is recommended to equip a Radio station with EBU Mode compliant loudness meters. While Tech 3341 contains the main ingredients of loudness metering (including test signals) it intentionally does not define the graphical user interface. For easier readability in general and for non-technical operators/producers in particular, this document recommends a **meter layout** as follows:

- The loudness meter shows the **relative scale** (−23 LUFS is displayed on the meter as **0 LU** according to EBU Tech 3341 and EBU R 128). It displays the "**EBU +9 scale**" for a higher resolution around the Target Loudness Level of 0 LU;
- The loudness meter uses a **contrasting colour** on either side of 0 LU to highlight the section where the anchor signal (speech, foreground music) should typically be. The meter's background colour and the highlighted section should be distinguishable by users with impaired colour vision (see **Figures 5 & 6** for examples);
- If the 400 ms time scale **Momentary Loudness** (M) is used for immediate visual feedback, the highlighted section of the display lies between **+3 LU** and **−3 LU** (see **Figure 5**);
- If the 3s time scale **Short-term Loudness** (S) is used for immediate visual feedback, the highlighted section of the display lies between **+2 LU** and **−2 LU** (see **Figure 6**);

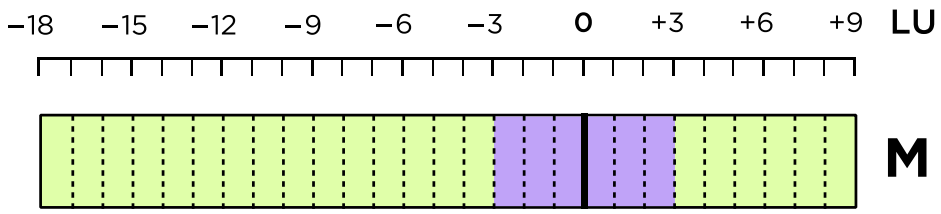

Figure 5: Highlighted range of ±3 LU for anchor signal levelling (Momentary Loudness)

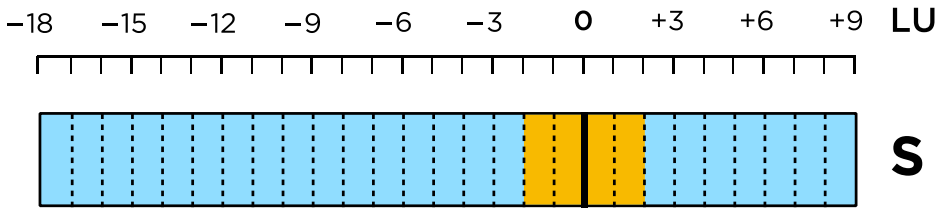

Figure 6: Highlighted range of ±2 LU for anchor signal levelling (Short-term Loudness)

- Especially (but not only) for self-operated studios, the **Short-Term Loudness** display is recommended. The display is more stable than Momentary Loudness and therefore easier to read at a glance. Experience of EBU Members has shown that Short-term Loudness correlates well with the perception of the loudness of speech, with the 3-second sliding window bridging the gap between words and sentences;
- The Loudness meter displays the **Programme Loudness Level** (ILK) as a clearly visible number with a precision of one decimal place;
- The loudness meter may be used in horizontal or vertical orientation. The display should be as big as practically possible to facilitate reading it within the peripheral field of vision.

5. Distribution

As stated in § 1, the **key concept** of Loudness in Radio is the clear distinction between the *production* side and the *distribution* side. In this section and in the Use Cases the term *Distribution Loudness* is used. It is the average Integrated Loudness of a day of linear broadcasting of a station, or the Integrated Loudness of a station's stream via online distribution. This term is used here for the sake of clarity and allows to talk of a Loudness value that includes all linear and non-linear listening.

The following factors should be considered for choosing a Distribution Loudness Level:

- For content produced and digitally distributed at **−23 LUFS**, many consumer devices (for example, smartphones) still have insufficient amplification to allow an adequate listening level in noisy environments[1];
- There still exists a culture in much of the Radio industry where being as loud, or louder, than other stations is believed to be a competitive advantage;
- DAB is received by the same devices as FM and AM analogue broadcasts whose Loudness is linked to transmitter power or controlled by government regulations. Nevertheless, most car radio

[1] It is anticipated that this situation will considerably improve in 2022 when CENELEC EN 50332 3 [10] and IEC 62368 1 [11] will effectively be enforced.

receivers automatically adjust the loudness of a station on different platforms to the same level;

- Radio exists in media devices that include audio streaming and download services, which may have chosen different (and also inconsistent) *Distribution Loudness* levels, for example Smart TVs and Smart Speakers;
- "Visual Radio" is consumed on smart devices via apps, leading to a direct level comparison with ubiquitous platforms such as YouTube and music streaming services;
- Radio exists in devices that also reproduce audio on physical formats such as CDs, where genres like Pop music still suffer from the "Loudness War". This decade-long race has systematically led to extreme loudness levels;
- Some broadcasters currently use DVB (digital satellite TV and digital terrestrial TV) as reserve feeds for Radio transmitters;
- Radio services share platforms with television services that are broadcast with a *Distribution Loudness* level of **−23 LUFS**;
- Work is continuously being done to harmonize loudness levels across platforms, resulting in, for example, EBU R 128 s2, AES71 [12], AES TD1004 [13] and CENELEC EN50332 [10].

Generally, Radio broadcasters should aim for a **Distribution Loudness Level** of **−23 LUFS**. As broadcast and other distribution platforms proliferate in parallel, −23 LUFS is the only common value that meets all the goals for a harmonious audience experience across all consumer devices and for all genres of audio content.

−23 LUFS is the goal for the *Distribution Loudness Level*.

A **Distribution Loudness Level** higher than −23 LUFS may be considered:

- if a Radio broadcaster does not want to change the alignment of the studio level to the nominal AM/FM carrier deviation;
- if a Radio broadcaster wants to compensate for still too low playback gain in mobile devices in challenging listening environments;
- if a Radio Broadcaster desires a loudness level closer to ubiquitous streaming services (Spotify, iTunes, Amazon Music etc.) and/or legacy and new media devices (CD, Smart Speakers etc.);

In such a case, the *Distribution Loudness Level* should be in the range of **−20.0** to **−16.0 LUFS**. This is compliant with the possible Streaming Loudness Level defined in EBU R 128 s2.

The case of **FM distribution** is covered in more detail in § 5.1.

A value between −20 and −16 LUFS may be considered as an intermediate Distribution Loudness Level.

5.1 FM Distribution

As analogue FM transmission is still widespread among EBU Members, a dedicated section is justified. EBU Tech 3344 describes in detail how to align a Radio station with production at −23 LUFS to the legacy FM stereo system. As the MPX (Multiplex) power limit specified in ITU-R BS.412 is based on an rms-measurement, a direct comparison with a loudness-based solution is not possible. Nevertheless, Radio broadcasters have performed long-term measurements of their legacy level alignment with a loudness meter. For countries using the MPX power limit, the loudness level has been often around −18 LUFS. The alignment of these stations[2] sets a Studio Level of −9 dBTP equal to a deviation of the FM carrier of ±40 kHz (for the audio part within the MPX, using a 400 Hz sine tone and no MPX power limiter in the signal chain).

Switching to the loudness paradigm, a Radio broadcaster has three options:

Option 1: to use a Distribution Loudness Level of −23 LUFS and change the alignment to the analogue FM carrier deviation to reach the same perceived loudness as before (illustrated in **Figure 7**);

Option 2: to leave the alignment as it is and change the Distribution Loudness level to reach the same perceived loudness as before (illustrated in **Figure 8**);

Option 3: to leave the alignment as it is and keep the Distribution Loudness level at −23 LUFS (illustrated in **Figure 9**).

Options 1 and 2 result in the same perceived loudness and the same headroom on the receiver side. Option 3 results in a lower loudness level but offers more headroom which, for example, makes it more appropriate for classical music.

EBU Tech 3344 recommends **Option 1** with the significant advantage of keeping the loudness level at −23 LUFS for all cases, also for DAB. The difference lies only in the Permitted Maximum True-peak Level (significantly higher for DAB, resulting in more headroom).

EBU Members having already switched to loudness normalisation in Radio have often chosen **Option 2**, keeping the current alignment of the FM distribution chain.

[2] For example, Radio stations in Germany and Austria

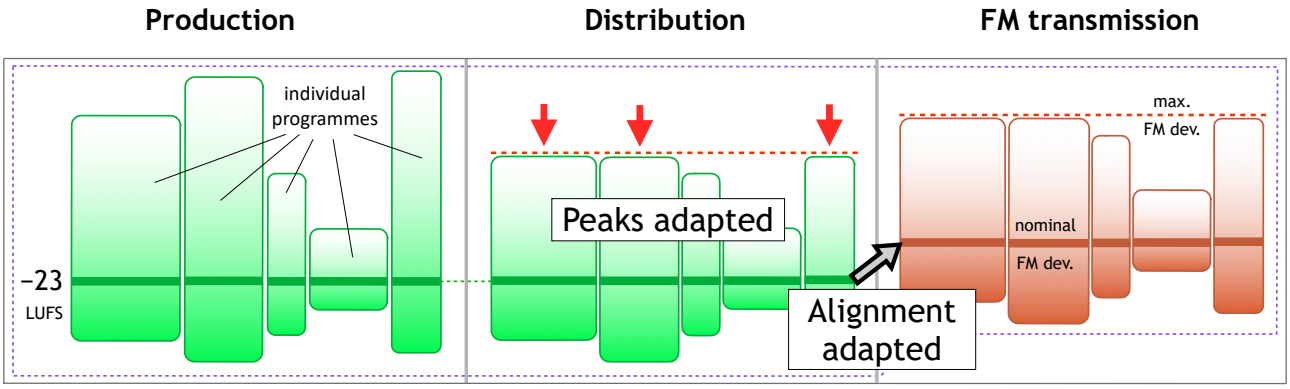

Figure 7 (Option 1): Adaption of the alignment from a Distribution Loudness Level of −23 LUFS to the analogue FM carrier deviation

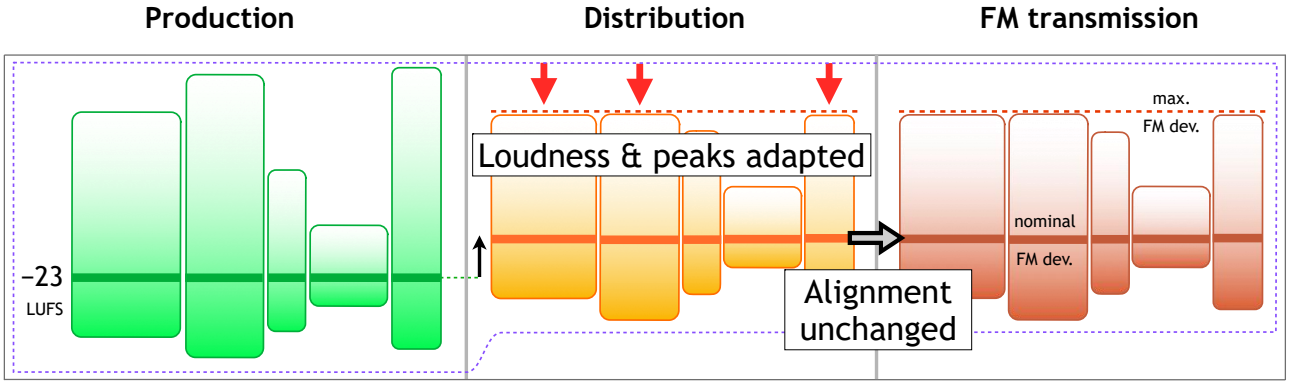

Figure 8 (Option 2): Adaptation of Distribution Loudness; legacy alignment to FM carrier deviation

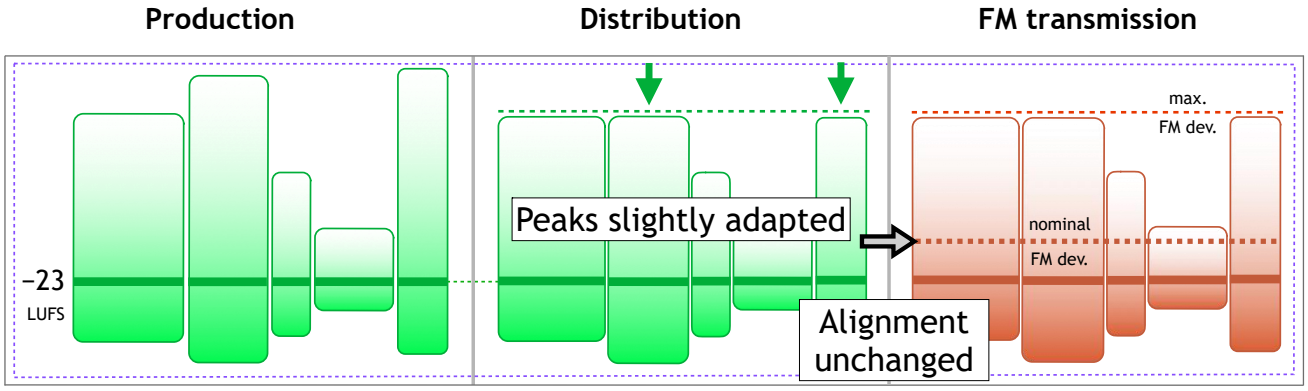

Figure 9 (Option 3): Distribution Loudness of −23 LUFS, legacy alignment to FM carrier deviation

As mentioned before, in most *car radios* the receiver chips perform so-called "blending"[3]. This aligns the loudness of an FM signal and a DAB signal of the same station to the same loudness level (measured according to ITU-R BS.1770). It is down to the individual car (radio) manufacturer to take advantage of this automatic loudness balance. As can be seen in the use case of NRK (§ 7.2), this was apparently not fully implemented in every car in 2012. The EBU has confirmation from several sources that the situation has considerably improved. To get evidence, the EBU will liaise with the WorldDAB forum for more details regarding the actual situation.

[3] According to CATENA, the R&D department of NXP Semiconductors, major manufacturer of car radio receiver chips.

6. Future developments

The media landscape is characterised by a wide variety of playback devices and listening environments. This leads to the often-found compromises concerning destructive dynamic processing in the distribution chain and to the associated inevitable loss of audio and signal quality.

In the future, the reproduction level and the dynamic properties of a programme will more frequently rely on *metadata-controlled* Playback Loudness and Dynamic Range Control (DRC). In a robust metadata system, this can provide a useful adaptation to the requirements of the listener's particular playback scenario.

The following factors should be considered:

- Developments in devices that require a *seamless transition* between different distribution platforms for the same audio service for mobile listening (for example, in-car listening);
- Advances in *non-linear* listening features. For example, a listener to DAB might choose to restart a programme, and that audio signal would then be delivered via the Internet;
- Advances in consumer devices allowing the dynamic properties of the audio signal (Loudness Range, Peak-to-Loudness Ratio) to be controlled by the user or automatically to better suit the listening environment. This may be achieved by metadata-controlled behaviour.

Radio broadcasters will then unanimously be able to distribute Radio programmes with the original dynamic properties and a Distribution Loudness level of −23 LUFS, as recommended in EBU R 128. By using comprehensive DRC metadata, broadcasters will be able to steer and control the dynamic processing parameters on the reproduction side.

The ANSI/CTA-2075 [14] standard describes metadata-based Loudness and DRC management in user video playback devices for a variety of codecs currently in use. These codecs are already supported by major manufacturers in many user playback devices.

Other standards concerning DRC are listed in § 8 (see [15] - [20]).

7. Use Cases

To demonstrate the practical application of EBU R 128 in Radio Production and Distribution, EBU Members already using loudness normalisation on their Radio services describe their experiences, successes and challenges. The common thread is the benefits of using loudness normalisation at −23 LUFS in production, vastly simplified delivery specifications, fewer operator errors and higher audience satisfaction.

7.1 BBC

7.1.1 BBC Radio delivery specification

Rough estimates suggest that in every hour, the BBC produces 57 hours of audio content. A lot of this content consists of live transmissions or repeats of live transmissions. A significant proportion of the content is pre-recorded.

The content is produced by BBC production departments across the globe and by over 400 independent production companies. The specification for programme levels used to say that audio levels should “peak to PPM 6” and this worked fine when all programmes were made in BBC studios that were maintained by BBC engineering teams. That is no longer the case and so the problem of not all programmes being made to the same levels has arisen. Many studios outside the BBC do not have a PPM, but furthermore, interpretations of what “peaking to PPM 6” actually means became

noticeably variable. New ways of listening on internet-connected devices mean that the audience experienced huge Loudness variations between programmes. “Peaking to PPM 6” on a sports commentary is different to “peaking to PPM 6” on a classical music concert, and different again to a popular music programme.

The solution:

- **EBU R 128** *is now the technical delivery specification for all BBC Radio*;
- Adopting EBU R 128 in Radio production has allowed the BBC to define a technical specification that can be met by anyone producing audio;
- The specification is *simple* to understand, and it produces *consistent* results;
- The delivery specification forms part of the contract between the BBC commissioners and the programme production company and using EBU R 128 has removed the potential for debate about compliance.
- The technical specification for programme delivery has gone from several pages of technical description to a just few sentences;
- On stations and on audience generated playlists, transitions between different genres of programme are now smoother and the differing sonic styles of the programmes have been maintained;
- For the studio operator, the playback of pre-recorded programmes is simpler and therefore there are fewer operational errors.

7.1.2 BBC Radio 4 Extra

BBC Radio 4 Extra is a UK national Radio station that broadcasts on DAB, DVB and online. It features documentaries, drama, panel shows and comedy from the BBC’s archive. The varied nature of the station means that recent content is often interspersed with recordings from as far back as the 1950s and occasionally, even earlier.

Although the archive recordings have been digitised, this has happened over a period of many years and there are inconsistencies in the way that level control has been applied. In some cases, peak normalisation to −10 or −6 dBFS has been used. In others, peak level has been allowed to approach 0 dBFS. The problem is illustrated by the DAW waveform display in **Figure 10**:

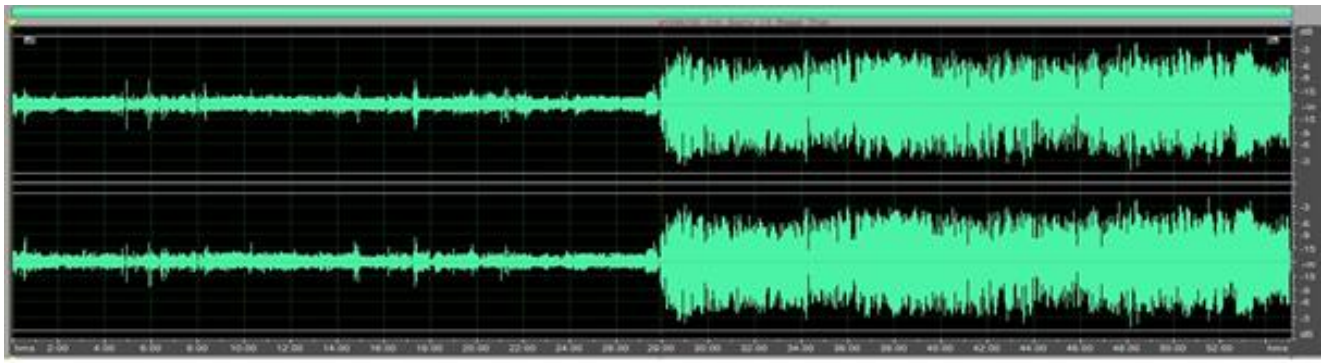

Figure 10: Audio waveform of two different programmes on BBC Radio 4 Extra

The left half of the display shows a recording of a drama *(‘Artists: The Inner Child’*) and the right half is a comedy show (*‘I’m Sorry I’ll Read That Again’*). These are as obtained from the archive prior to broadcast. The maximum peak levels for these items are −9 dBFS and −1.5 dBFS and the integrated loudness −33 LUFS and −17 LUFS respectively – a difference of 16 LU!

This is an extreme example, but significant loudness differences between archive recordings is common and must be compensated for, or it will result in unacceptable loudness jumps for the listener. BBC Radio 4 Extra uses automated playout from a schedule of programmes that are uploaded to the playout server in advance. To avoid gross loudness variations of the type shown above, programme staff had to preview each item and manually apply gain or attenuation based on what their ears and a PPM told them was necessary. Any remaining loudness discrepancies were then smoothed out by the transmission processor, which was set accordingly.

A pilot project showed that the above steps could be simplified by using loudness normalisation on the archive files. The programme scheduler was given access to a network folder into which they loaded the day's programmes. This batch-normalised each file to −23 LUFS and presented them to an output folder for uploading into the playout system. Not only did this remove the need for manual checking of loudness and level, but it also resulted in more consistent loudness at the input to the transmission processor. This made it possible to back-off the AGC and compression settings and improve the listening experience.

Following a successful pilot of this loudness normalisation process, it has become a permanent feature in the workflow of Radio 4 Extra. It has been found necessary to use normalisation software that includes a peak limiter: Although it was found that most of the archive content could be normalised to −23 LUFS without peak limiting, a proportion of dramas were found to contain transients that required controlling to prevent them from exceeding −1 dBTP.

7.1.3 BBC Distribution Loudness for UK national Radio services.

Table 1 shows the current Loudness values for BBC Radio services, by genre of programming. The values predate AES TD1004 'Recommendation for Loudness of Audio Streaming and Network File Playback' [13] and EBU R 128 s2 'Loudness in Streaming'. These numbers are averages and only roughly indicate the situation. All this is achieved with legacy processing technology using measurements such as Sample Peak in dBFS and other standard 1990s parameters of multiband compression.

Table 1: Loudness values for BBC Radio services, by genre

		Popular Music	Classical Music	Speech
Studio Output	**Loudness** (LUFS)	−19	−27	−23
	Max True Peak (dBTP)	−3	−6	−1
	Max LRA (LU; Guideline)	7	15	6
DAB Distribution	**Loudness** (LUFS)	−14	−23	−18
	Max True Peak (dBTP)	−4	−2	−5
	Max LRA (LU; Guideline)	3	15	6
DVB and Linear Online Distribution	**Loudness** (LUFS)	−14	−27	−18
	Max True Peak (dBTP)	−4	−6	−5
	Max LRA (LU; Guideline)	3	15	6
Non-Linear Online Distribution (Podcasts and web audio clips)	**Loudness** (LUFS)	−18	−18	−18
	Max True Peak (dBTP)	−2	−2	−2
	Max LRA (LU; Guideline)	7	15	6

The BBC will be looking for the next generation of processing tools to use parameters measured in dBTP and LUFS as defined in ITU-R BS.1770.

The values are **not** necessarily where the BBC wants to be, and several factors pull the decisions on distribution processing in different directions. The weight applied to each of these factors is different for each service and is a compromise between the engineering ideal and editorial desire.

The factors that pull on these decisions are:

- A desire that the listener should be able to move between different devices and listen to the same service, without large jumps in Loudness;
- A desire that a listener moving between services on the same device, won't experience large Loudness jumps between different services;
- A desire for high audio fidelity to the listener – particularly for Classical Music services and Arts & Drama speech programmes;
- A desire to be "Loudest on the dial" and for the sound to be "competitive" – particularly for popular music services;
- A desire to keep the engineering simple and thus have lower running costs.

7.2 NRK

The Norwegian Public Broadcaster NRK started Radio over DAB on a small scale in 1994. The DAB network has since been expanded through the years and two commercial broadcasters have joined. All national Radio services on FM were switched off in Norway during 2017. At the same time, NRK decided to switch to DAB+ as the emission format. Digital Radio is also available on all TV platforms and on the Internet, both "live" and "on demand".

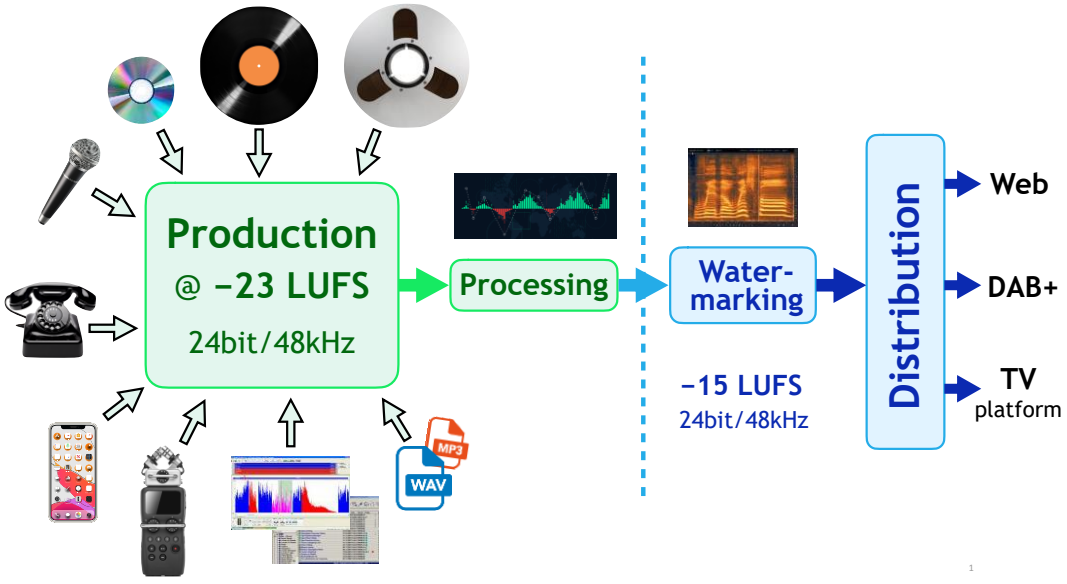

Figure 11: The NRK Radio chain

At NRK, four key elements in the total Radio chain (see **Figure 11**) that may harm audio quality were pinpointed: *Production faults - Transmission processing - Audio watermarking - Lossy emission codecs*.

As the audio quality reaching the listener is dependent on all four degrading elements, the focus is on minimizing the impact of each element. What may sound "great" in production may not sound that great to the listeners at the far end of the total Radio chain.

NRK's first digital Radio infrastructure was based on MPEG 1 layer II at 384 kbit/s. When the switch to DAB+ was made in 2017, the Radio production format was also changed to *linear PCM*, as listening tests at NRK showed that DAB+ demands audio that is free from any lossy bitrate reduction prior to final encoding. To preserve high audio quality throughout the complete production chain 24-bit linear PCM @ 48 kHz was chosen as the default production format. The audio signal is automatically scanned for loudness data and adjusted to −23 LUFS either by metadata or a static gain change.

All studios and production facilities are equipped with hardware loudness meters. Only a few daily live broadcasts are in the hands of audio engineers. Most live playouts are "self-op" by the presenters of a show. They may use anything from traditional mixers with physical faders to faderless operation on a computer screen. Loudness normalised audio levels are a must for "self-op" Radio.

In February 2012, NRK and the two commercial broadcasters with national services in Norway agreed on a *common Distribution Loudness Level* for Digital Radio, namely −15 LUFS. There were at least two arguments for choosing -15 LUFS. Firstly, -23 LUFS, as used in TV production, was too soft for portable players. The second argument is related to potential traffic hazards: back in 2012, the DAB networks had quite good coverage in metropolitan areas, but most road tunnels had FM coverage only. If you listened to DAB at −23 LUFS in your car and drove into a road tunnel, after 3 seconds or so the car Radio would switch to FM at an audio level about 8 dB louder than -23 LUFS. That's a big traffic hazard!

Figure 12 shows the level variation of 21 Radio stations in Norway after the switch to a common Distribution Loudness Level. The variability has been reduced dramatically!

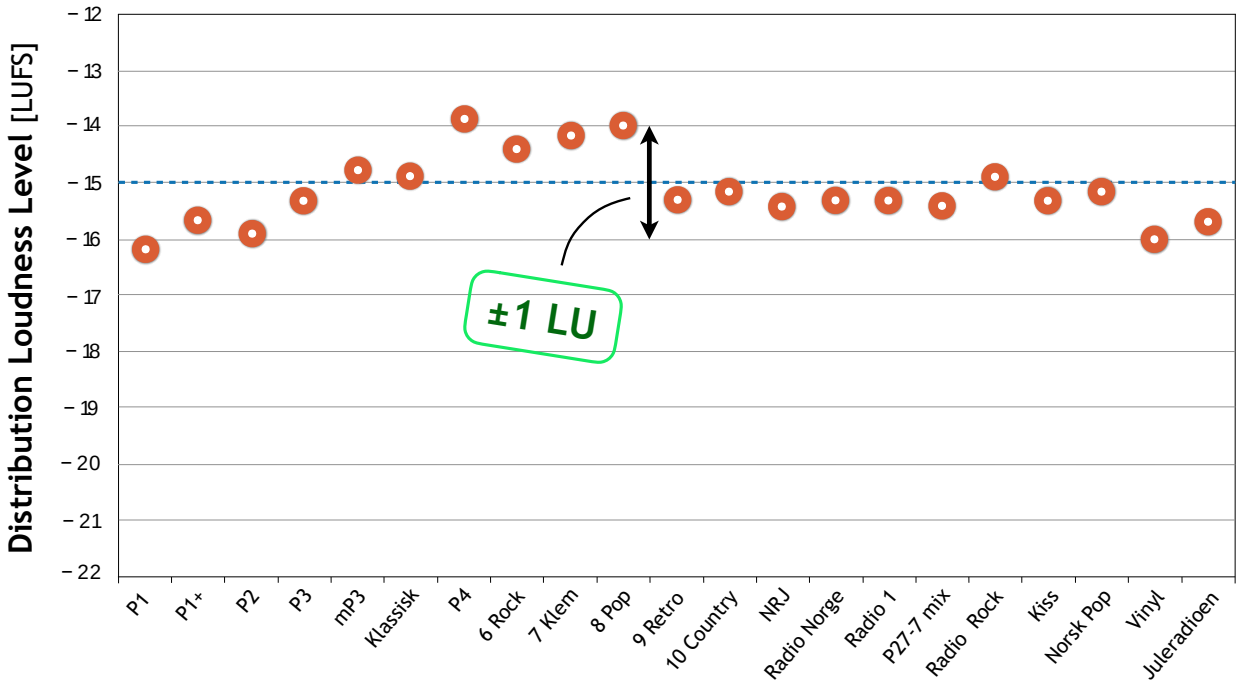

Figure 12: Spot check of Radio Distribution Loudness levels on DAB in Norway

In 2012, NRK had a plan to slowly back down to -23 LUFS as analogue Radio was switched off. But nowadays there exist many competitors in the form of new platforms such as YouTube and Spotify with target levels up to −16 LUFS. Digital Radio still sits at −15 LUFS in Norway, and NRK sees no immediate reason to change to a softer target level. The 8 dB gain change from −23 LUFS in production is done in transmission processors with presets to suit the content in each Radio service. Consistent levels in production greatly reduces audible artifacts in output processing.

For statistical purposes, all national Radio- and TV services in Norway have a watermark embedded in the audio (**Figure 13**). At NRK, the watermark is inserted between the output processors and emission coding.

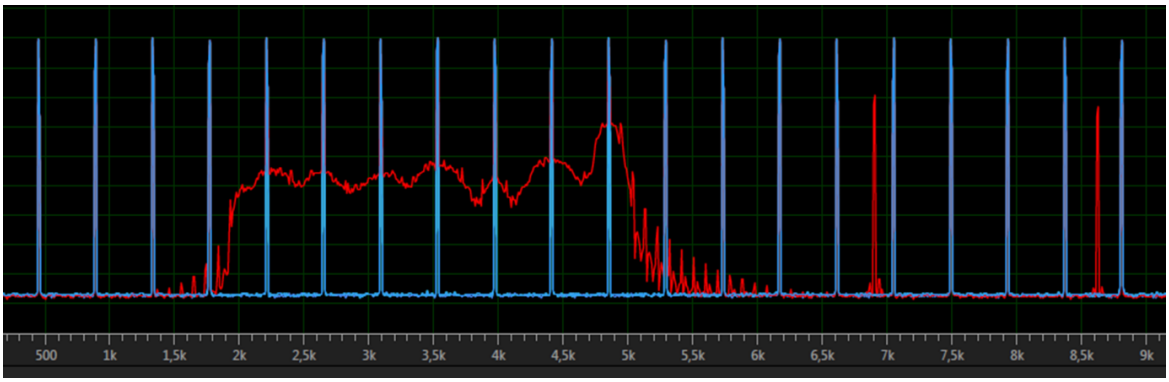

Figure 13: Typical watermark signal (red), "riding" on an audio test signal (blue)

Lessons learned:

- Introducing loudness normalisation is easier when all distribution/emission is in the digital domain;
- Loudness normalisation is only successful when all broadcasters agree to a common target level;
- Normalising the output level of most Radio services is quite easy: just measure and adjust;
- All Radio services need not be at a common loudness target level. People expect Radio drama and classical music to be somewhat softer;
- Radio listeners are not like TV viewers (on the sofa in their living room). Radio listeners are "all over the place" with very varying listening conditions. Audio levels must be chosen accordingly;
- 95% of the programmes produced can easily be levelled to −15 LUFS by adequate processing. Exceptions may be content such as classical music and Radio drama. Clever use of modern processing can raise levels to −15 LUFS4 while maintaining good audio quality.

Please judge for yourself: NRK Radio Klassisk (not geoblocked).

7.3 Rundfunk Berlin Brandenburg: rbbKultur

rbbKultur includes a daytime programme with a magazine character:

- Music, mainly classical in a broader sense;
- Short presentations between the music pieces;
- Several times a day one or two studio guests and/or phone-ins: discussions on current topics, mostly culture;
- Short pre-produced spoken-word contributions;
- News (takeover from our Radio station rbb-InfoRadio);
- Culture news.

The evening and weekend programmes consist mainly of:

[4] The EBU hopes that Norway will lower the Distribution Loudness Level to at least −16 LUFS, in accordance with EBU R 128 s3 ("Loudness in Radio") [4] and the present document.

- Concerts (live and recorded);
- Radio drama and documentaries;
- Music programmes of specific genres (classical, jazz, experimental pop music).

Production and broadcasting of rbbKultur was switched to loudness-based control to:

- Avoid loudness jumps within the programme;
- Have better control over the loudness relationship between the presentation and the music. The presentation was typically perceived as being too loud;
- Avoid the deformations of the dynamics of the broadcast signal caused by dynamic processing, which partially led to the inversion of intended volume variations, especially in classical music and Radio drama.

7.3.1 Loudness in production at rbbKultur

Before loudness normalisation was introduced, pre-produced programmes and music stored in the audio database had inconsistent loudness levels and inconsistent peak levels due to a variety of approaches to levelling during ingest procedures in the past. The files were played back unchanged. Before distribution, the broadcast signal was subjected to aggressive dynamic processing to compensate - among other purposes - for the inconsistent loudness levels.

Consequences:

- The loudness relationship between presentation and music could hardly be controlled;
- The aforementioned dynamic deformation up to the reversal of intended volume variations was profound.

The solution to overcome the above issues is the **adoption** of **EBU R 128** in *production*. The recommendation is well suited as a guideline for production, though it would be helpful to have even more headroom for some very dynamic programmes such as classical music and Radio drama. Ideally, if all programme segments have the same loudness of −23 LUFS according to R 128, there are no disturbing loudness jumps within the programme. Therefore, the goal was loudness level control at the source. Live sources and pre-produced material should be controlled according to R 128 as early as possible in the signal flow or production process.

This was achieved by the following measures:

- Installation of *loudness meters* in all locations where live broadcasts are made and where pre-production for the broadcast takes place;
- Training of staff in loudness-based production and broadcasting;
- Installation of loudness-based levellers for all suitable live sources, for example, for phone-in lines and take-over of non-R-128-compliant spoken word contributions, but not for pre-produced classical music and live concert broadcasts. For very dynamic programmes, manual control according to R 128 must be used, as an automatic system cannot know the artistic flow of the dynamic level variations;
- Automatic levelling of presentation and studio guest mic signals turned out to be problematic. The presenters felt restricted in their creative freedom, when it came to adapt the style of their speech to the character of the music presented. It was also impossible to find a universal setting for automatic levelling that worked for all presenters and all studio situations with one or two guest mics. Finally, rbb decided to work without automatic levelling for studio mics, except in the case of massively exceeding the target loudness level. The *presenters* are now in charge of *manual R 128-compliant levelling*;

- A presentation/music relationship of −3 to −1 LU turned out to be satisfactory for our daytime programme. rbb decided against the spoken word being the 0 LU anchor signal, because that would mean further limitation of headroom and consequently more dynamic processing for music. Thus, music and pre-produced programmes are played back at 0 LU while presentation is manually levelled by the presenters with a target level of 0 LU and then attenuated by 3 dB. This typically results in a loudness level of −3 to −1 LU for the presenter's voices;
- The audio database has been upgraded so that the loudness of all pre-produced programmes is automatically measured, and the values are stored as metadata. The playout system in the broadcast control room then uses the metadata to normalise the loudness of each programme by simply statically changing the playback gain – the audio files are not changed.

Successes:

- The presentation/music relationship can be controlled much better;
- Loudness jumps during the programme are avoided to a large extent;
- No more deformation of intended volume variations in the broadcast signal occurs because there is no longer a need to compensate for differences in loudness through aggressive dynamic processing of the transmitted signal.

7.3.2 Loudness in distribution at rbbKultur

The objective of introducing loudness-based control in *distribution* was to improve the dynamics processing. Especially in classical music and Radio drama, the deformation up to the inversion of intended volume variations in the broadcast signal was disturbing.

rbbKultur is distributed on four platforms:

- FM
- DAB+ (HE AAC @ 96 kbit/s)
- Live stream (AAC ABR up to 256 kbit/s)
- DVB (MPEG Layer II @ 320 kbit/s)

Target loudness

EBU R 128 with its Programme Loudness Level of −23 LUFS is well suited as a guideline in production. As long as metadata-based Dynamic Range Control in end user devices has not caught on, higher target values for the *Distribution Loudness Level* can be used in distribution to counteract insufficient electroacoustic gain in some end-user devices and for the sound to be "competitive".

For streaming, R 128 s2 was published in August 2020 and recommends −18 LUFS "unless metadata is employed to manage device gain and the dynamic properties of content". For (usually pop) music programmes "with a Peak-to-Loudness Ratio (PLR) lower than 15 dB, a PL value of −16.0 LUFS is allowable." The AES had already recommended a programme loudness between −20 LUFS and −16 LUFS, and the public broadcasters in Germany had recommended −20 LUFS for "special content such as culture channels".

Implementation

By introducing R 128 in the production domain, the need to compensate for loudness jumps through dynamics processing in the distribution domain was eliminated. *The processing could then be far less aggressive*.

After intensive listening tests of highly dynamic programmes (classical music and Radio drama), rbb decided to completely remove automatic (loudness) levelling from the signal chain. It had become

evident that current automatic levelling cannot deal with intended large loudness variations in classical music and Radio drama in an adequate way. That is because the machine has no information about the intention and the temporal behaviour of these variations.

Instead, rbb introduced compression and limiting with moderate settings for evening and weekend programmes and less moderate settings for the daytime programme. Lacking automatic levelling, a target loudness level is not defined. Instead, a long-term loudness measurement is conducted and manual adaptation of the settings of the processing is performed if necessary.

rbb's compromise between competitive programme loudness and high-quality dynamics processing results in a *Distribution Loudness* level of about −20.2 LUFS for the daytime programme and about −20.9 LUFS for the evening programme, which is comparable to other classical music Radio stations in Germany.

A potential problem caused by our more-or-less moderate processing could be that very soft parts of classic music recordings with very high dynamics would be too soft on air and would be "lost below the noise floor". To compensate, rbb is prepared to introduce dynamics processing based on Loudness Range (LRA) at the source, that is, batch Loudness Range processing of music pieces with very wide dynamic properties. So far there has been no negative feedback, so this LRA processing step has not been necessary.

As the four distribution platforms have their specific requirements, rbb has introduced separate dynamics processing for each of them.

For legal reasons, rbb must limit the FM signal at −6.3 dBTP and use MPX limiting. As the listening environments for FM and DAB+ are usually the same, it was decided to use the same dynamic compression for FM and DAB+. Nevertheless, rbb wanted to take advantage of the benefits that DAB+ offers to reduce limiting, so a higher True-peak limiting threshold (−3.0 dBTP) was chosen. Furthermore, MPX limiting for DAB+ is not used. For rbb, this constitutes such a great advantage that the rare inconvenience of minor loudness jumps when Radio tuners are switching between FM and DAB+ due to DAB+ signal loss was readily accepted.

For streaming, the final decision has not yet been taken. Therefore, for the time being, the same processing as for DAB+ is used.

For DVB-S, the unprocessed signal is used, a gain of 1 dB is added and True-peak limiting at −2 dBTP is applied. The 1 dB gain exists because the average programme loudness of culture channels on DVB-S in Germany is roughly −22 LUFS.

For the future, rbb counts on ARD, the EBU and world-wide common solutions for the loudness level of all distribution channels, either using loudness and DRC metadata or recommendations accepted by all broadcasters.

7.4 SWR

7.4.1 Loudness profiles

In a highly automated content management and playout system loudness and dynamics of individual programme segments should be managed. The balance of programme segments determines the overall impression of a programme. There is no generally applicable balance. The key is to find the best balance for individual programme types. The "programme-based loudness profiles" approach presented here offers a general and easy-to-implement solution based on metadata.

7.4.1.1 Semantic Labels (SLb)

In Radio, it is common to define the presenter's voice as the loudness anchor. This provides a reference to which all other programme segments can refer to. The voice anchor is set to 0 LU (−23 LUFS). The loudness of all other components can be specified as an offset to the anchor. Values of ±3 LU are common. This way, for example, the loudness relationship of speech to music can be defined in a programme.

For the management of different programme components, semantic labels (SLb) are introduced to distinguish between different types of programme segments. Supplemented by specifications for loudness and dynamics, or more generally by audio processing parameters, programme-based loudness profiles are created that characterise the aural impression of a programme.

Eight labels form a main set of categories (Numbers 0 to 7 in **Table 2**). Additional subcategories can be used to specify content even more precisely, for example, "Music of the 60s" (SLb 4.1) or "Music of the 70s" (SLb 4.2). This allows automatic audio processing to be individually applied to programme segments.

Content management systems already use similar metadata. It is desirable to extend them to the semantic categories listed in Table 2. For the exchange of contributions in media networks, it should be possible to reset the subcategories to their respective main category to ensure a seamless exchange of metadata beyond one's own channel.

Table 2: Semantic labels used to distinguish between different types of programme components

	Loudness Profiles ➜	**News Magazine**		**Concert Replay**		**News & Classics**		
No.	**Semantic Label**	**PL**	**LRA**	**PL**	**LRA**	**PL**	**LRA**	
0	Raw audio	Not for playout						
1	Speech, news	0 LU	3 LU	0 LU	3 LU	0 LU	3 LU	
1.1	Lively discussion	0 LU	5 LU					
2	Speech, arts & drama	0 LU	7 LU	0 LU	9 LU	0 LU	7 LU	
3	Magazine (speech & music)	0 LU	No pr.	0 LU	No pr.	0 LU	No pr.	
4	Music	0 LU	8 LU	+3 LU	No pr.	0 LU	15 LU	
4.1	60s music	0 LU/8 LU, EQ						
4.2	70s music	0 LU/8 LU, EQ						
5	Advertising	0 LU	No pr.	0 LU	No pr.	0 LU	No pr.	
6	Layout/Signets/SFX	No processing at all						
7	Others	Not for playout						

7.4.1.2 Programme-based Loudness Profiles

By adding loudness and dynamic parameters to each Semantic Label, a *Programme-based Loudness Profile* is created. Loudness profiles fully describe the acoustic impression of a programme. Table 2 starts with the simplest profile type **"News Magazine",** a Radio magazine with news reports and popular music: All programme segments have the same loudness. Different LRA values for speech and music on the one hand ensure the intelligibility of spoken word contributions and on the other hand provide enough dynamics for music. Popular music stations may only need this one profile.

Music programmes require a different profile. The next Loudness Profile shown in Table 2 after "News Magazine" is **"Concert Replay"**, representing a classical music programme. For a good mix of

announcements and music, music should be played about 3 LU louder. This profile can be used as a template for all programmes where music is the main signal, from Radio hit parades to the Eurovision Song Contest.

Programme-based Loudness Profiles in a content management system can be realised via plug-in processes that render all programme segments to their respective target parameters before play-out. This way, temporary versions of an original file are created for one-time broadcast. Within the same day, a Mozart movement can be played early in the morning in a "News Magazine" with reduced dynamics, but in the evening as part of a "Concert Replay" with full dynamics.

7.4.1.3 Processing

Loudness processing in modern DAWs is performed about 20 times faster than real time. This is fast enough to process Breaking News contributions while they are already announced.

Using a DAW, additional audio processing beyond loudness normalisation can also be defined for SLbs, for example, adaptive equalizing. In SWR's workflow the audio processing parameters are stored in an XML file that represents the entire processing. Many DAWs already offer these possibilities natively.

To separate the original audio file from temporary versions defined by the Loudness Profile, a suffix can be defined for each XML, which is appended to the file name when the different versions are saved.

7.4.1.4 Practical application

When a contribution is ingested into the playlist of a programme, the processing XML is identified in accordance with the Semantic Label and the processing is executed. The processing should be fast enough to process contributions even during an ongoing programme. In the unusual case that the processing is not completed in time, the playout system should fall back on the unprocessed original file to avoid broadcast blackouts.

Show opener, sound effects and other layout elements are often short and distinctive. It is necessary to produce layout elements appropriate for the specific use and then to exclude them from loudness processing within the framework of SLbs. For this purpose, layout elements get their own SLb.

Figure 14 shows the audio workflow at SWR using Loudness profiles (SLP – "Sendungs-Lautheitsprofil" (Programme Loudness Profile)).

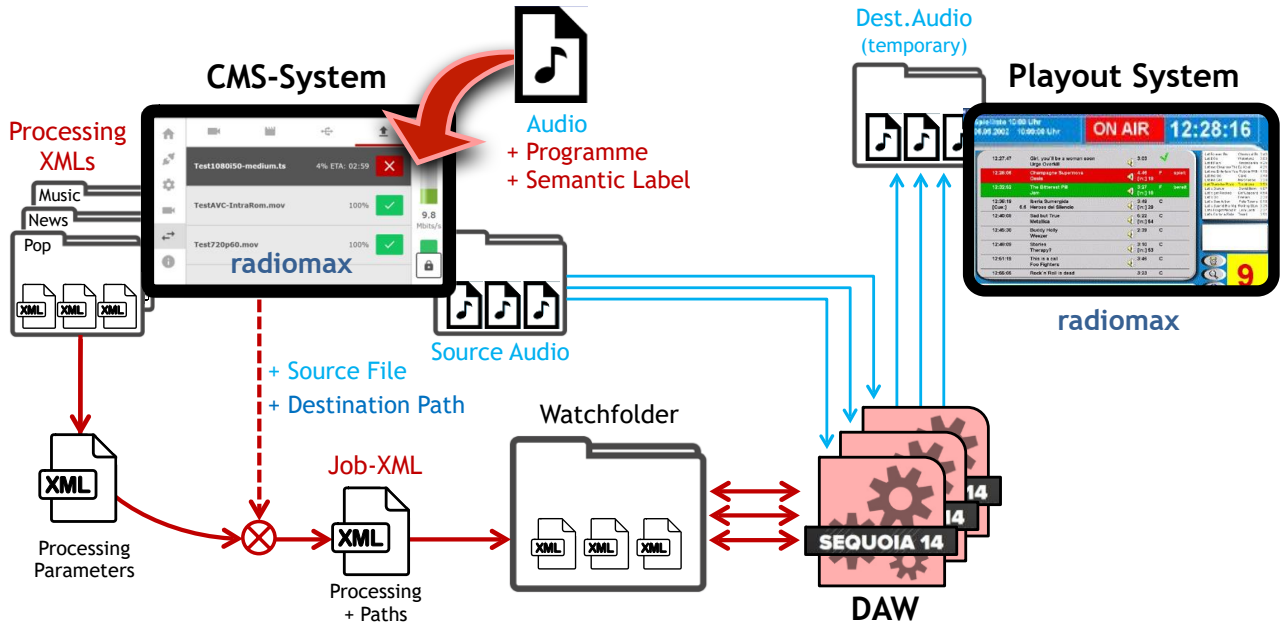

Figure 14: Audio workflow at SWR using programme loudness profiles (SLPs)

The individual audio segments and components are ingested into the Content Management System (CMS) with their respective Sound Label and Programme destination. Based on this information, an XML file stored in the Programme Loudness Profile activates the particular processing of the audio file in one of three Digital Audio Workstations (DAW). The temporary loudness-processed audio file is forwarded to the Playout System. The component or segment is thus specifically tailored to the respective programme.

7.4.2 Youth channel with popular music

"DASDING" is a multimedia youth channel from the German Broadcaster SWR. Based on the web portal *DASDING.de*, a 24-hour Radio channel is distributed via audio streaming, DAB+ and DVB-S. FM is available in a few cities. The channel is characterised by modern pop music. A video livestream from the studio webcam can be watched via a smartphone app.

Before the introduction of loudness normalisation, there had been complaints about insufficient speech intelligibility of some presenters' voices, especially when talking over music. Furthermore, many audio contributions had to be levelled manually to simplify self-op broadcasting.

DASDING was the fourth programme of the SWR where loudness control was introduced. It was possible to build upon the experience gained from the other programmes.

7.4.2.1 Live audio workflow

At SWR it is agreed that all live signals from professional sources must be compliant to EBU R 128. Non-compliant signals, for example, telephone, web applications or microphones, which usually contain spoken word, are routed separately to the mixing console to be normalised by channel processing. Level-based processing such as AGC and compressors respond quickly but achieve an accuracy of only ±3 LU. R 128-based AGCs located directly pre-fader are responsible for the rest. The presenter's microphone gets a small boost of up to +1 LU.

Special attention has been paid to crosstalk between microphones. Side chains of compressors are linked, however, R 128-AGCs remain unlinked, but gain is limited to 3 LU to avoid build up. Channels for file-based audio signals do not contain any processing.

7.4.2.2 Metering and Signalling

Traditional QPPM level meters are replaced by loudness meters. For self-op, *Short-term loudness* is displayed on the *'EBU +9 scale'*. In studios operated by technicians, *Momentary* and *Short-term* loudness are displayed. To emphasize the target loudness level, a range of ±3 LU is highlighted on Momentary and ±2 LU on Short-term loudness meters (see § 4.2, Figures 5 & 6).

The on-air fader switch for microphones is assigned two additional functions. The R 128-based AGC freezes when the fader is closed. It also controls the start-stop measurement of integrated loudness. An announcement triggers a measuring interval from reset&start to stop. The integrated loudness of the latest announcement is displayed. This is because presenters can hardly check loudness during speaking or even reading.

7.4.2.3 File-based audio workflow

At SWR, Loudness Profiles for stations or programmes have been introduced. Different programme components are distinguished by Semantic Labels to be processed individually. Main categories are spoken-word contributions, music and sound elements. Each category is assigned values for the loudness target level and the permitted maximum LRA value. Processing is done via DAW, requested by the content management system – see previous use case "Programme-based Loudness Profiles". **Table 3** shows the target parameters for the playout of programme items for the youth channel "DASDING". The system is powerful enough to process files in parallel to be resilient to, for example, breaking news.

Table 3: Loudness target values for different programme components based on −23 LUFS

Loudness Profile for the youth channel "DASDING"		
Category	**Loudness**	**Max. LRA**
Spoken word	0 LU	2.5 LU
Music	0 LU	6 LU
Sound element	No processing	

7.4.2.4 Production

As very few loudness tools were available in production when loudness normalisation was introduced, the waveform display of the audio editing software is shifted up by 5 dB. This way, the established visual habits are used to keep the loudness at a reasonable level. For this workflow, Loudness normalisation is not necessary during production since the Integrated Loudness Level as well as LRA are corrected before playout anyway.

7.4.2.5 Instructions and training

For presenters, a studio instruction session lasting 30 minutes turned out to be sufficient. Technicians received a half-day theoretical loudness training followed by practical exercises.

7.4.2.6 Further Improvements

Using clip-based loudness normalisation during editing would result in better-balanced contributions. It would be possible to set the LRA process to a higher LRA, which would reduce the workload of LRA processing and lead to more lively results.

7.4.3 SWR Distribution Loudness for Radio

In 2014, SWR introduced loudness normalisation to its news channel SWR-Info. As a result, levelling errors could be eliminated in the broadcast studio and subsequent correction became obsolete. The distribution processing was redefined, no longer as a correction instance, but as a channel-related

processing regarding the specific distribution platform. This allowed the signal quality to be significantly increased, especially for classical music and arts & drama stations.

The basic decisions of SWR were to:

- use a Distribution Loudness Level of −18 LUFS combined with a True Peak level of −6 dBTP as intermediate delivery for the FM radio network;
- broadcast on DAB and the internet at −16 LUFS/−1 dBTP for all channels;
- broadcast arts & drama as well as classical music additionally at a competitive loudness of approximately −16 LUFS;
- apply soft audio compression instead of levelling to classical music, arts & drama.

Table 4 shows the current values for three stations of SWR and four distribution platforms.

Table 4: Target values for SWR's different distribution platforms for three stations

		SWR Aktuell (News)	**SWR2 (Classical Music, Arts & Drama)**	**DASDING (Popular Music)**
Broadcast Studio Output	**Loudness** (LUFS)	−23	−23	−23
	Max True Peak (dBTP)	−1	−1	−1
	Max LRA (LU)	~4	~25	~6
FM	**Loudness** (LUFS)	−18	(−20…) −19	−18
	Max True Peak (dBTP)	−6	−6	−6
	Max LRA (LU)	~3	~16	~3
DVB-S	**Loudness** (LUFS)		−23	
	Max True Peak (dBTP)		−1	
	Max LRA (LU)		~25	
DAB & WEB	**Loudness** (LUFS)	−16	−16	−16
	Max True Peak (dBTP)	−1	−1	−4
	Max LRA (LU)	~3	~19	~3
Audio on Demand (WEB)	**Loudness** (LUFS)	−16	−16	−16
	Max True Peak (dBTP)	−1	−1	−1
	Max LRA (LU)	~3	~20	~3

8. References

[1] **EBU R 128** *'Loudness normalisation and permitted maximum level of audio signals'* https://tech.ebu.ch/publications/r128

[2] **EBU R 128 s1** *'Loudness parameters for short-form content (adverts, promos, etc.)'* https://tech.ebu.ch/publications/r128s1

[3] **EBU R 128 s2** *'Loudness in Streaming'* https://tech.ebu.ch/publications/r128s2

[4] **EBU R 128 s3** *'Loudness in Radio'* https://tech.ebu.ch/publications/r128s3

[5] **EBU Tech 3341** *'Loudness Metering: 'EBU Mode' metering to supplement EBU R 128 loudness normalisation'* https://tech.ebu.ch/publications/tech3341

[6] **EBU Tech 3342** *'Loudness Range: A descriptor to supplement EBU R 128 loudness normalisation'* https://tech.ebu.ch/publications/tech3342

[7] **EBU Tech 3343** *'Guidelines for Production of programmes in accordance with EBU R 128'* https://tech.ebu.ch/publications/tech3343

[8] **EBU Tech 3344** *'Guidelines for Distribution and Reproduction of programmes in accordance with EBU R 128'* https://tech.ebu.ch/publications/tech3344

[9] **ITU-R BS.1770** *'Algorithms to measure audio programme loudness and true-peak audio level'* https://www.itu.int/rec/R-REC-BS.1770/en

[10] **CENELEC EN 50332-3** *'Sound system equipment: headphones and earphones associated with personal music players - maximum sound pressure level measurement methodology - Part 3: measurement method for sound dose management'* *https://www.en-standard.eu/csn-en-50332-3*

[11] **IEC 62368-1** *'Audio/video, information and communication technology equipment – Part 1: Safety requirements'*

[12] **AES71-2018** *'Recommended Practice Loudness Guidelines for Over the Top Television and Online Video Distribution'* https://www.aes.org/tmpFiles/aessc/20210225/aes71-2018-i.pdf

[13] **AES TD1004** *'Recommendation for Loudness of Audio Streaming and Network File Playback'* https://www.aes.org/technical/documents/AESTD1004_1_15_10.pdf

[14] **ANSI/CTA-2075** *'Loudness Standard for Over the Top Television and Online Video Distribution for Mobile and Fixed Devices'* https://shop.cta.tech/products/loudness-standard-for-over-the-top-television

[15] **ETSI TS 102 366** *'Digital Audio Compression(AC-3, Enhanced AC-3) Standard'* https://www.etsi.org/deliver/etsi_ts/102300_102399/102366/01.04.01_60/ts_102366v010401p.pdf

[16] **ETSI TS 103 190-1** *'Digital Audio Compression (AC-4) Standard; Part 1: Channel based coding'* https://www.etsi.org/deliver/etsi_ts/103100_103199/10319001/01.03.01_60/ts_10319001v010301p.pdf

ETSI TS 103 190-1 *'Digital Audio Compression (AC-4) Standard; Part 2: Immersive and personalized audio'*
https://www.etsi.org/deliver/etsi_ts/103100_103199/10319002/01.02.01_60/ts_10319002v010201p.pdf

[17] **ISO/IEC 23003-3** *'Information technology — MPEG audio technologies — Part 3: Unified speech and audio coding'* https://www.iso.org/standard/76385.html

[18] **ISO/IEC 23003-4** *'Information technology — MPEG audio technologies — Part 4: Dynamic range control'* https://www.iso.org/standard/75930.html

[19] **ISO/IEC 23008-3** *'Information technology — High efficiency coding and media delivery in heterogeneous environments — Part 3: 3D audio'* https://www.iso.org/standard/74430.html

[20] **ETSI TS 103 491** *'DTS-UHD Audio Format; Delivery of Channels, Objects and Ambisonic Sound Fields'*
https://www.etsi.org/deliver/etsi_ts/103400_103499/103491/01.01.01_60/ts_103491v010101p.pdf

All links were current at publication.