Levelling and Loudness

— in radio and television broadcasting

Gerhard Spikofski and Siegfried Klar
Institut für Rundfunktechnik GmbH (IRT)

Sudden differences in loudness between – and even within – radio and television programmes have been well known for a long time. With the more-recent introduction of digital techniques, combined with the parallel transmission of digital and analogue broadcasts, this problem is again becoming highly significant.

This article presents some solutions for avoiding loudness differences in radio and television broadcasting, based on levelling recommendations and a newly-developed loudness algorithm.

Listeners and viewers are becoming increasingly concerned over sudden variations in programme *loudness*. These loudness "jumps" are most apparent when zapping through European DVB television and radio channels. The loudness differences between film dialogues and highly-compressed commercial breaks (adverts) are perceived as being particularly jarring. Both under-levelling and over-levelling can be observed, resulting in level differences of more than 15 dB.

The reasons for such programme *level* **and** *loudness* variations are, among others:

- apparent inexperience in the levelling of sound channels;
- the use of different and sometimes non-standardized programme *level* meters;
- no standardized *loudness* meter has been available until now;
- archive material (both analogue and digital) has not, to any great extent, been adapted to the types of sound channel being used today.

In FM radio broadcasting, loudness is mainly balanced or controlled by means of compressors and limiters that prevent the frequency deviation of the transmitter from exceeding the permissable limits.

In the case of digital broadcasting, it should also be possible to achieve balanced loudness profiles – by following the existing international recommendations of the ITU and the EBU. These profiles should be met not only when comparing different programmes / channels, but also between different contributions within any single programme.

Characteristics of radio programme meters

Alignment level

ITU Recommendation ITU-R BS.645-2 [1] defines the programme level of radio channels by means of an *alignment signal* (1 kHz sine wave). The specified level of the sine wave corresponds approximately to full-scale programme level, in terms of *loudness*.

Table 1
Audio levels in studio and transmission environments

Recommendations for analogue & digital audio levels	Alignment Level (AL) –9 dB (35%)	Nominal Level (PML) [a] 0 dB (100%)
ITU-R BS.645-2 Transmission Level (international)	0 dBu [b]	+9 dBu
ARD HFBL-K Studio Level (national)	–3 dBu (adaptation)	+6 dBu (adaptation)
US Reference Level (national)		+4 dBu (adaptation)
EBU digital Transmission & Studio Level (international)	–18 dBFS	–9 dBFS [c]

a. PML = Permitted Maximum Level
b. 0 dBu = 0.775 V rms (sine wave) = 1.1 V peak
c. dBFS = Clipping Level (FS = Full Scale)

As the alignment signal is “static”, it can be measured by means of typical RMS meters as well as specific programme meters.

It should be noted that the analogue *alignment level* (AL), and the “nominal” or *permitted maximum level* (PML), are specified diversely due to the different national and international recommendations in use *(see Table 1)*.

In the case of digital audio channels, the relationship between the alignment signal and the full-scale or clipping level was already specified in 1992 (EBU Rec. R68 [2]). When following this recommendation, the difference between full-scale (or clipping) level and the alignment level is 18 dB *(Table 1)*. In other words, the alignment level should be –18 dBFS.

Audio programme meters for broadcasting

Today, many different programme meters are in use at professional studios, with widely varying ballistical features *(see Table 2 and Fig. 1b)*.

Table 2
Programme meters used in international transmission and studio environments

Programme Meter Type	Recommendation	PML [a] 100%	Limit Level	Scale	Attack time (integration)	Decay time (fall-back)	Invisible peaks
VU Meter	ANSI C 16.5 IEC 268-17	0 VU +0 dBu		–20 to +3 [dB]	300 ms / 90%	300 ms / 10%	+13 ... +16 dB
DIN PPM (QPPM)	DIN 45406 IEC 268-10/1 ARD Pfl.H.3/6	**0 dBr** +9 dBu	+16 dBr +25 dBu	–50 to +5 [dB]	10 ms / **90%** **5 ms / 80%**	20 dB / 1.5 s = 13 dB/s	+3 ... +4 dB
BBC PPM (QPPM)	IEC 268-10 / IIa	‘6’ +8 dBu		1 to 7 [ ]	**10 ms / 80%**	24 dB / 2.8 s = 8.6 dB/s	+4 ... +6 dB
EBU PPM Std (QPPM)	EBU 3205 E IEC 268-10 / IIb	**+9 dB** +9 dBu		–12 to +12 [dB]	10 ms / 80%	24 dB / 2.8 s = 8.6 dB/s	+4 ... +6 dB
EBU Digi PPM (QPPM)	EBU IEC 268-18	**–9 dBFS**	0 dBFS	–40 to +0 [dB]	≤ 5 ms / 80%	20 dB / 1.7 s = 12 dB/s	+3 ... +4 dB
IRT Digi PPM (QPPM)	IRT proposal	**0 dBr** **100%**	≤ +10 dBr	–50 to +10 [dB]	**5 ... 10 ms / to 80%**	20 dB / 1.7 s = 12 dB/s	+3 ... +4 dB

a. Permitted Maximum Level (PML): 100% Modulation = +9 dBu = –9 dBFS for transmission lines [1][2]
≡ +6 dBu ARD Nominal Studio Level [3].

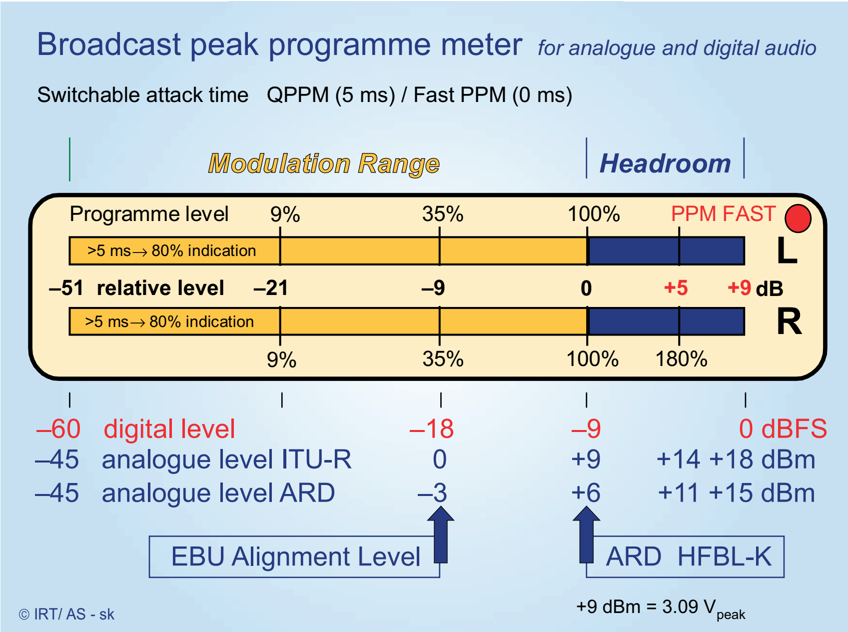


Figure 1a
Recommended broadcast peak-programme meter

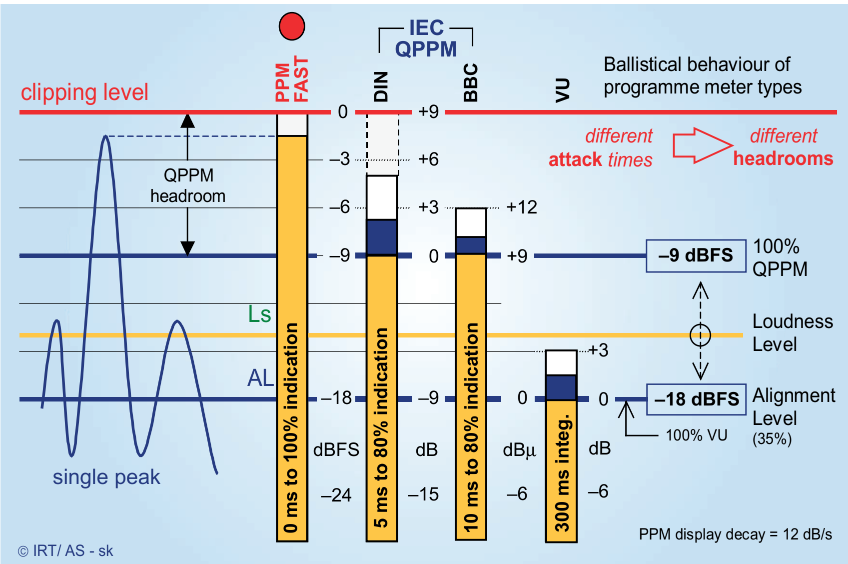


Figure 1b
Ballistical characteristics of different broadcast programme meters

Whereas in America and Australia, VU meters [4] are mainly used, the *peak programme meter* (PPM) is recommended by the EBU [5] for use in European countries. They are specified in the following IEC recommendations:

- IEC 268-10 [6] (analogue PPM);
- IEC 268-18 [7] (digital PPM).

The IEC category of PPM is the so-called *quasi-peak programme meter* (QPPM) which neglects any short-duration signal variations. For digital PPMs, the EBU recommends almost the same ballistical characteristics as that described in IEC 268-10 (Type 1).

Since the introduction of digital audio techniques in broadcasting, additional – but not precisely specified – PPMs have caused some confusion. Besides their different scale layouts, these PPMs primarily vary in their ballistical features – described by parameters such as *attack time* or *integration time*, and *fall-back time* or *decay time*.

Table 2 shows the PPMs that are currently used in Europe. Regarding the layout of the scale, the full-scale tag (100% tag = 0 dB) – and also the specified *headroom* – should take into account the “attack time” of the programme meter.

As an example, the VU meter – which can be considered as relatively slow – obviously needs an appropriate headroom because of the “invisible” signal peaks. Consequently, the difference between the 100% tag and the alignment level has to be smaller than in the case of other meter types.

> **Note:** The attack time of the PPM used by the German broadcasters ARD and ZDF [8] is specified as 10 ms / 90%. This means that it takes 10 ms to reach the 90% tag. The IEC meter type which is used by the BBC is specified slightly differently (10 ms / 80%).

In the case of the fast digital *sample programme meter* (SPPM), theoretically no headroom is needed. These meters are appropriate for controlling signal peaks with respect to clipping but they are not as suitable as QPPMs for normal programme levelling. For example, signals with a high proportion of peaks tend to be under-levelled whereas heavily-compressed signals with limited peaks tend to be over-levelled. This can result in huge jumps in loudness, which seem to be more intensive than when using a QPPM.

The use of unspecified level meters is widely observed in the digital audio field. If sound engineers are reasonably familiar with a particular level meter, the use of an unspecified device could result in severe levelling mistakes such as clipping and loudness jumps. Because unspecified instruments offer a wide gamut of characteristics, it is difficult to become familiar with them and gain sufficient experience in levelling.

Digital programme meters are frequently software applications. As is well known from such applications, there are "infinite" error sources. Because the "attack time" tends towards 0 ms, this means that the peak samples are indicated correctly. However, there are wide variations in the decay time. Those effects can result in different displays as well as differences in level.

In Germany, the QPPM is precisely specified in ARD-Pflichtenheft 3/6 [8]. This meter is recommended for the levelling of both analogue and digital audio signals. Additional PPMs – with attack times shorter than 10 ms – are also specified here, but should only be used for monitoring – not for levelling.

In order to avoid confusion, the IRT recommends that the scale layout of the digital PPM is adapted to that of the analogue QPPM [6] *(Fig. 1a, Table 2)*. That means that the 100% tag has to be 9 dB below full scale.

Dynamic range of digital audio systems

Programme levelling and headroom

As already mentioned, the levelling range and the necessary headroom depend on the ballistical features of the meter in use. Whereas VU meters need up to 18 dB headroom, PPMs only require 9 dB [1][9][10].

The 9 dB headroom of the EBU PPM is strictly linked to QPPMs that accord with [1] and the alignment level specified in [2]. Using instruments with different ballistical features obviously results in other headroom recommendations.

Headroom has to be considered as a buffer range between the nominal and clipping levels. If the European recommendation is followed, the exchange of programme material is guaranteed to have no levelling problems. German broadcasters have accepted this recommendation and the headroom is specified in document ARD HFBL-K Rec. 15 IRT [3], which accords with the EBU recommendation. In the case of analogue signals and also devices that involve A/D and D/A conversions, the absolute audio limit at German broadcast studios is +15 dBu (100% tag = +6 dBu, plus another 9 dB headroom) *(see Fig. 1a)*.

Usable dynamic range – objective and subjective considerations

When discussing headroom and footroom [10], the question always arises – whether the resulting system dynamics are sufficient to accommodate the full dynamic range of the human ear. In other words, which quantization level or how many bits are necessary to guarantee the transmission of music signals without any perceivable noise.

One answer to this question was given in a paper published in 1985 [11]. In the following section, the conditions and results of this 1985 study are presented.

This investigation was conducted before the era of bitrate reduction systems such as MiniDisc (ATRAC), MPEG-1 Layer 2 (mp2) and Layer 3 (mp3). Bitrate reduction systems are therefore not considered in this context. Compared to PCM (Pulse Code Modulation) systems, the aforementioned bitrate reduction systems evidently need less quantization. The fact that they allow noise-free recordings nevertheless shows that, in these cases, other quality features have to be considered.

In PCM systems, the dynamic range of a system is defined as the level differences between full-scale programme level and the inherent noise level of the system.

The dynamic range, the signal-to-noise ratio and the quantization noise can be calculated by means of the following formula:

$$S/N\ [dB] = 6n + 2$$

... where n = quantization level (number of bits).

Table3
Achievable signal-to-noise ratios for different quantizations and noise-level measurements

Noise voltage level	16-bit	20-bit	24-bit
RMS (dB)	–98	–122	–146
DIN 45 405 (dB) [12]	–90	–114	–138
ITU 468 (dBqps) [13]	–86	–110	–134

The calculated value – with a negative sign – corresponds to the RMS value of the quantization noise, relative to 0 dBFS programme level (the Full Scale / Clipping Level of a digital system). *Table 3* shows the RMS noise values for three typical quantizations. These absolute values represent the maximum dynamic range (in dB) for each of the three quantizations shown in the table.

If we consider a headroom of 9 dB [2] and a footroom of 20 dB [10], the derived values for dynamic range are shown in *Fig. 2* as a function of quantization.

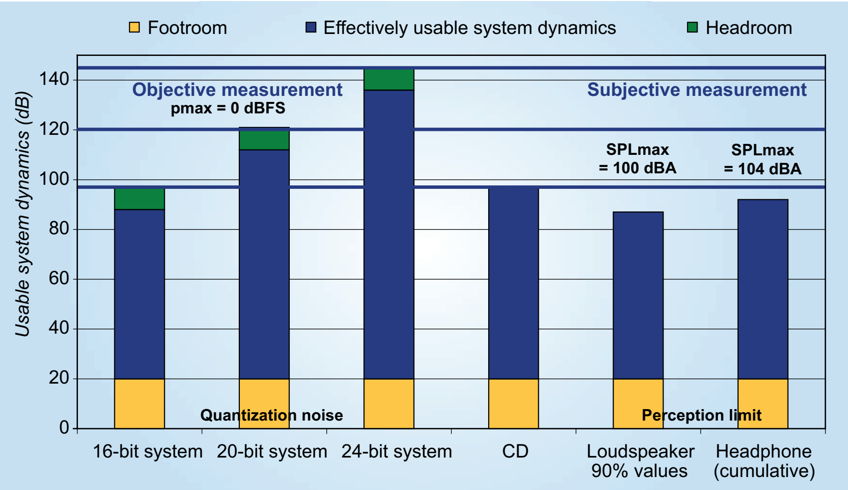


Figure 2
Usable dynamic range of digital PCM systems – both subjective and objective

In principle, the reference values for the dynamic range – the maximum full-scale programme level on the one hand and the system noise level on the other – correspond to certain sound pressure levels in the reproduction of music signals. The relevant sound pressure levels are the ***maximum listening level*** and the ***just imperceptible noise level***.

These two levels were actually investigated separately by the IRT, despite the fact that they are linked together as system features. The five selected test items (female speech, male speech, orchestral, string quartet and rock music) were only used to determine the *maximum listening level*. The representative noise signals (idle channel noises, white noise, etc.) were investigated in the absence of programme signals. That meant that the disturbing noises were only assessed during music pauses – without having to consider the masking effect that would occur in the presence of programme signals.

Subjective experiments were carried out with 20 normal “listeners” in individual sessions. The listening set-up met the requirements for professional listening evaluations, including stereo loudspeaker and headphone reproduction [14][15].

The results of the investigation are presented in *Fig. 2*. The two aforementioned reference values correspond to (i) the 90% value of the cumulative frequency distribution of the maximum listening levels (dBA) and (ii) the average value of the individual perception limits for the system noises that were investigated.

In the left part of *Fig. 2*, the relationship between quantization and system dynamics is shown for three linear PCM systems (16-bit, 20-bit and 24-bit). In each case, the recommended headroom of 9 dB and footroom of 20 dB have been included. The results show that a linear 16-bit system, such as CD, just meets the requirements of the human ear for loudspeaker reproduction. In the case of headphone reproduction, the human requirements are only met if the headroom allowance is relinquished – which is normally the case with CD production today. Consequently, for digital audio studio production, where headroom and footroom are essential, the test results presented show that professional audio production needs at least 18-bit systems.

Programme levelling and loudness

Programme levelling

Level adjustments are controlled by means of a level meter (e.g. QPPM) such that the maximum programme levels almost meet but do not exceed the 100% tag. In German broadcasting, the level meter QPPM accords with IEC 268-10 [6] and is standardized for both analogue and digital signals. Meeting the 100% tag, which implies a 9 dB headroom, guarantees transmissions that are free of distortions. This does not mean that no amplitudes greater than 100% occur. Any short-term peaks that are invisible to the sound engineer should not generally produce clipping – because a sufficient headroom of 9 dB is provided, as a result of extensive programme signal analysis [9].

Programme loudness

As is generally known, the same *levelling* applied to different programme signals does not normally result in the same *loudness* impression. This discrepancy is especially evident when comparing music and speech. In order to reach a uniform *loudness* balance in mixed broadcast programming, special levelling recommendations have been defined following detailed investigations [17][18].

Meeting these recommendations in situations where speech is more important (e.g. magazines, motoring programmes and commercials), the speech should be levelled to 0 dB and the music to between –8 dB and –4 dB.

Those recommendations are useful for avoiding extreme loudness differences between and within broadcast programmes. However, adapting the programme loudness to suit the requirements of the human ear cannot always be achieved by this means alone. This is particularly true when using special audio processors. In this case, when adapting the loudness of broadcast programmes to the characteristics of the human ear, an additional **loudness meter** is necessary along with the **level meter** which is controlling the technical levels.

Although some investigations had been carried out in this field [16][19][20][21], no standardized loudness meter is available at the moment. Loudness corrections today still have to be done manually by the control engineer. This, of course, is not practicable when most of the control functions are handled automatically.

However, new investigations have shown that a studio loudness meter may be realizable [21], using new loudness algorithms based on measuring both the signal level and the signal power.

The following methods were tested:

- loudness measurement – RTW [16];
- loudness measurement – EMMETT [19];
- signal level – QPPM;
- signal power level – PWR.

The study dealt with both the subjective and objective aspects of loudness measurements. In the former case, psychoacoustic measurements were carried out to determine the subjectively-perceived loudness of the selected broadcast programme material. In the latter case, objective measurements were aimed at deriving rel-

Abbreviations

A/D	Analogue-to-Digital	**ISO**	International Organization for Standardization
ADR	Astra Digital Radio	**ITU**	International Telecommunication Union
AL	Alignment Level	**MPEG**	(ISO/IEC) Moving Picture Experts Group
dBFS	dB relative to Full-Scale reading	**PCM**	Pulse Code Modulation
D/A	Digital-to-Analogue	**PML**	Permitted Maximum Level
DAB	Digital Audio Broadcasting (Eureka-147)	**PPM**	Peak Programme Meter
DSR	Digital Satellite Radio	**QPPM**	Quasi-Peak Programme Meter
DVB	Digital Video Broadcasting	**RMS**	Root-Mean-Square
FM	Frequency Modulation	**SPPM**	Sample Peak Programme Meter
IEC	International Electrotechnical Commission	**VU**	(Audio) Volume Units

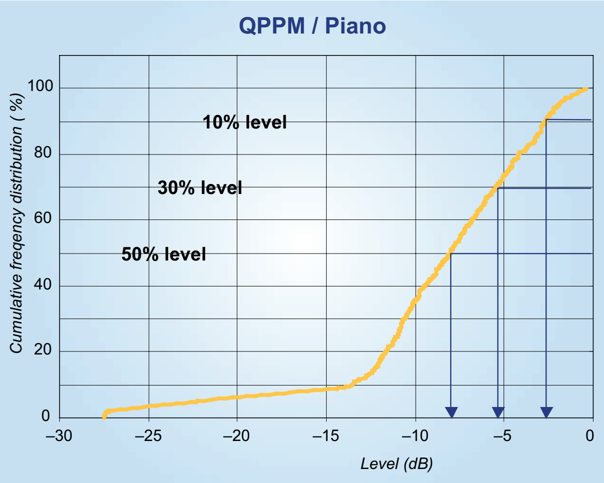


Figure 3
Principal analysis of the cumulative frequency distribution of objective loudness levels

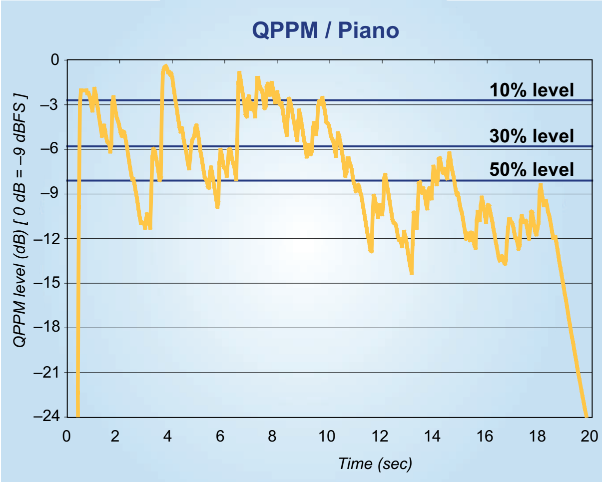


Figure 4
QPPM level and cumulative frequency rates under test (test item = piano)

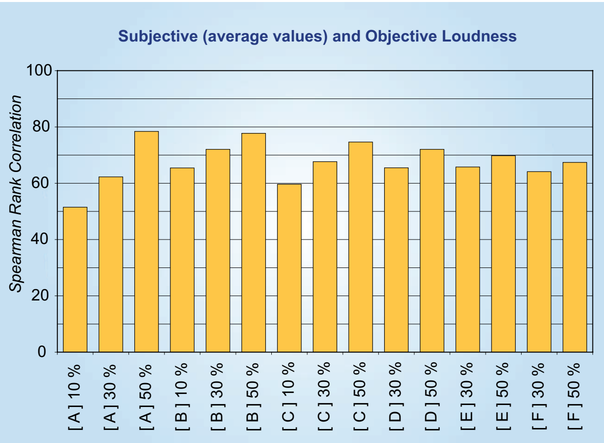


Figure 5
Spearman Rank Correlation between subjective (average values) and objective loudness parameters (A - F, not specified, are the six loudness algorithms under test)

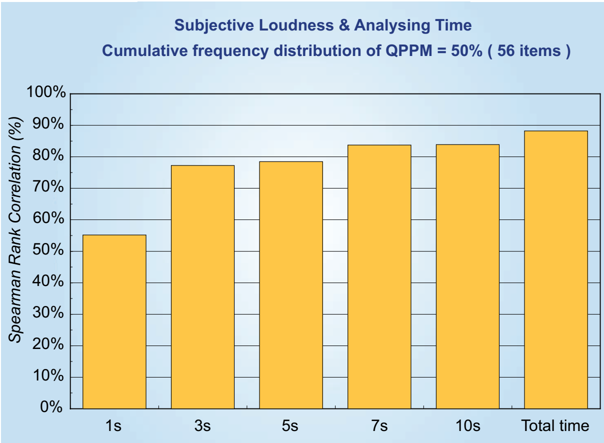


Figure 6
Correlation between subjective and objective QPPM loudness – variation of analysing time

evant signal parameters which would allow us to define *objective loudness*. The performance/accuracy of the objective parameters was assessed by correlating them with the associated subjective loudness values.

The test material comprised recordings of DSR (Digital Satellite Radio) with 16 stereo radio programmes, recorded in 1984. Each of the 16 programmes was represented by audio clips of about 15 s duration. The 56 clips eventually chosen contained announcements, orchestral, chamber, piano, vocal and pop music. This selection of clips was considered to be representative of actual radio programming at the time, especially with respect to levelling and audio processing.

In order to derive relevant objective parameters for each of the loudness algorithms and programmes, audio-level histograms (frequency of specific level values within the item duration) were analysed. In each case, the *cumulative frequency distribution* was plotted, to illustrate how programme levels were being exceeded for 10%, 30% and 50% of the time *(Fig. 3)*.

As an example, the measurement of QPPM vs. Time (incorporating the analysed cumulative frequency distribution) is presented in *Fig. 4*. The measurements were made using the ARD-Pflichtenheft Nr. 3/6 level meter, with 10 ms integration time and a release time of 1.5 s [8].

The criterion used for assessing the performance of these loudness algorithms is the *Spearman Rank Correlation* between the subjective and objective loudness measurements. Whereas subjective loudness is represented

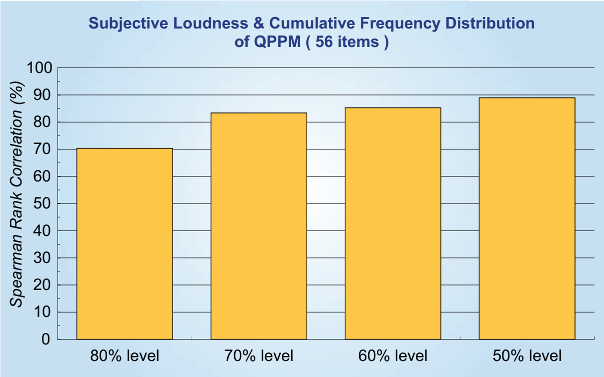


Figure 7
Correlation between subjective and objective QPPM loudness
– variation of cumulative frequency rate

by the average values of the subjective loudness assessments, the corresponding objective parameters are the levels that were exceeded for 10%, 30% and 50% of the time.

With reference to *Fig. 5*, it can be seen that the 50% level displays the highest correlation, for all the algorithms tested.

If we consider just the 50% values in *Fig. 5*, a correlation > 67% is achieved with each algorithm (labelled A - F), whereas algorithms A and B display the highest correlation (78%).

Because of the high correlation coefficients of algorithms A and B, and because of the relatively small deviations between the subjective and objective loudness parameters [21], these two algorithms form a good basis for developing a studio loudness meter. It can be stated that these programme meters are in accordance with the meter specified in [8], with an integration time of 10 ms.

In order to optimize the loudness algorithm with the level meter specified in [6] (with 10 ms integration time and 1.5 s release time), additional measurements were carried out. Among other parameters, the cumulative frequency distribution (60%, 70% and 80%) and the *analysing time* (1 s, 3 s, 5 s, 7 s and 10 s) were tested. The corresponding results are presented in *Figs 6 - 8*.

After optimizing the parameters under test, the resulting correlation between subjective and objective loudness amounts to 90%. The individual results of subjective and objective loudness are presented in *Fig. 8* with additional indication of the average values and the 95% confidence intervals of the subjective loudness levels.

Based on these results, a loudness algorithm was defined and a prototype of the studio loudness meter was developed. At the moment, this prototype is undergoing tests – with special emphasis being given to practical performance problems.

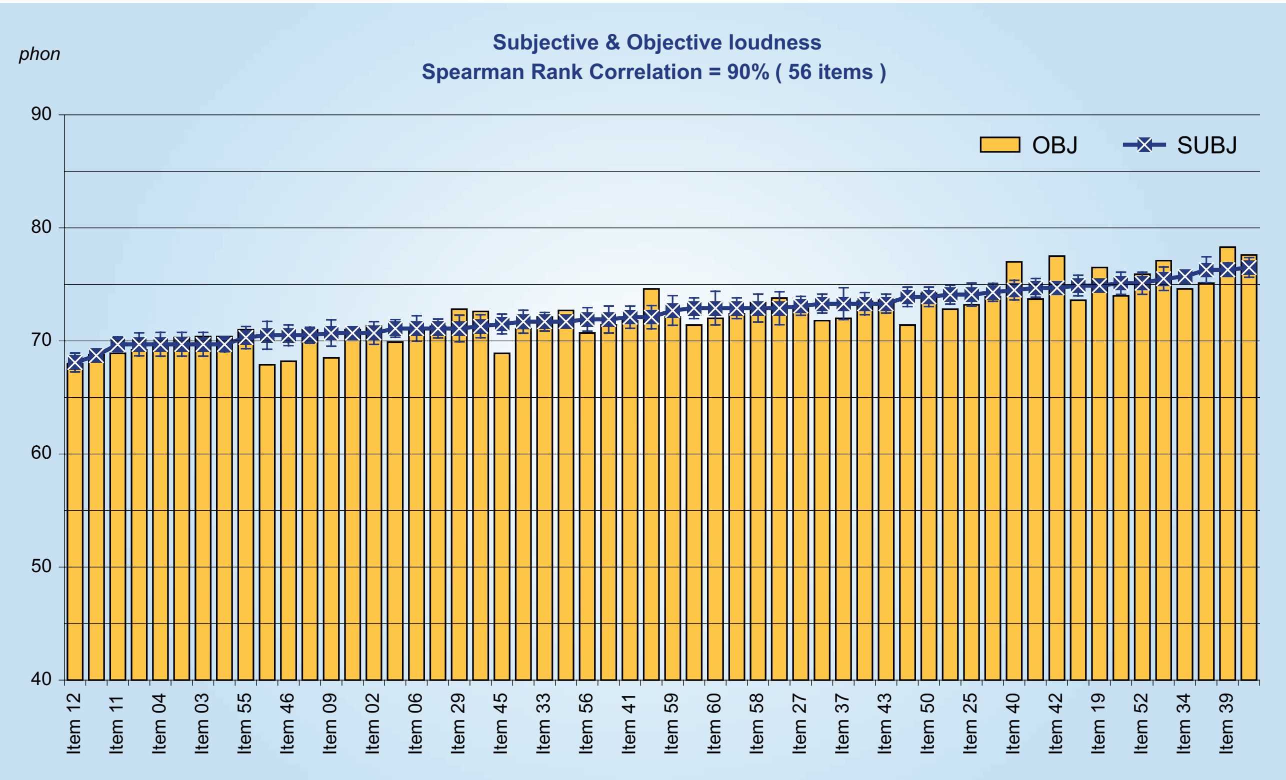


Figure 8
Subjective (averages and 95% confidence intervals) and objective QPPM loudness

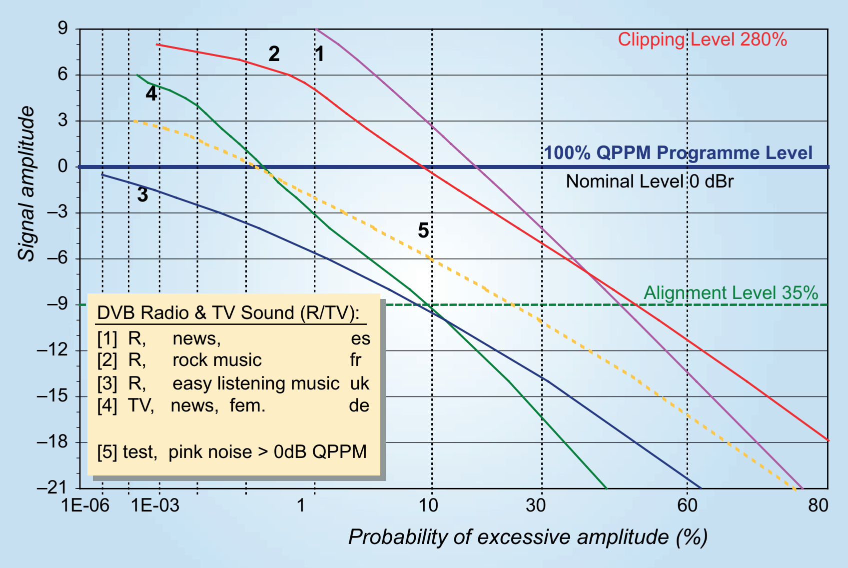


Figure 9
Amplitude statistics of DVB radio and TV signals

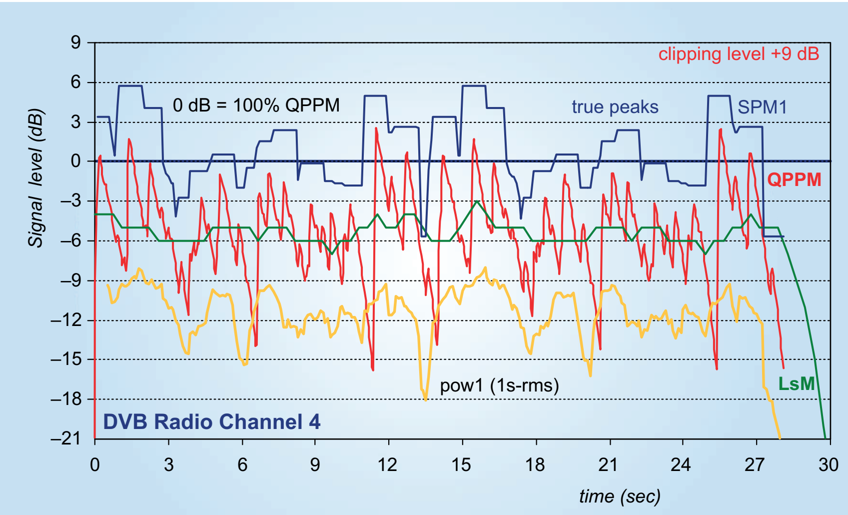


Figure 10
Programme levels (QPPM, SPPM, PWR 1s, QPPM Loudness LsM) – female speaker (no compression)

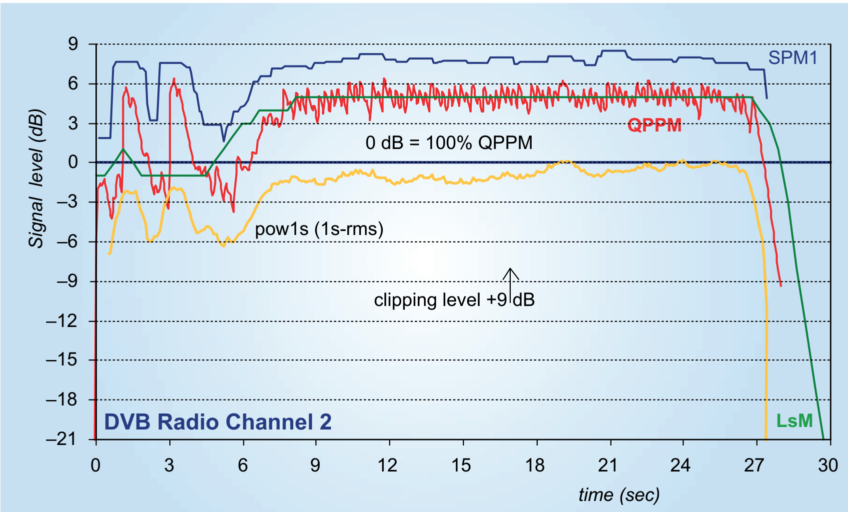


Figure 11
Programme levels (QPPM, SPPM, PWR 1s, QPPM Loudness LsM) – pop music (high compression)

Programme analysis of DVB channels

In order to gain experience with this newly-developed studio loudness meter, audio measurements were carried out on different European DVB channels. Besides the loudness levels (LsM) and the signal levels (PPM, QPPM), the signal amplitudes were also included in the measurement campaign. The following two methods of analysing the derived data were considered as appropriate:

- **Amplitude statistics** – analysis of the cumulative frequency distribution of the audio samples. The form of the diagrams presented here (signal amplitude vs. probability of exceeding the amplitude) yield interesting information about loudness and compression features of the analysed signals *(Fig. 9)*.
- **Level registration vs. time** – recording the normally displayed levels (e.g. QPPM, SPPM, PWR 1s, QPPM-Loudness LsM) for later evaluation of the programme signals *(Figs 10 - 11)*.

The measurement results presented in *Figs 9 - 11* show beyond doubt that there are tremendous differences between the DVB channels under test, when considering amplitude statistics, QPPM, PPM and LsM. In other words, the results clearly display non-adherence to the relevant ITU levelling recommendations [1].

Programme and loudness levelling in digital sound broadcasting

General aspects

Digital Radio offers the chance to get rid of those constraints that are well known in analogue FM radio. In Digital Radio, there is no relationship between loudness and transmission range that requires audio processing. Therefore, the wide dynamic range of Digital Radio can be used to good effect, e.g. to broadcast the full dynamic range of top-quality CD recordings.

First of all, the transmitters have to be levelled correctly according to the relevant ITU/EBU Recommendations [1][2]. This should prevent the occurrence of extreme variations in programme loudness. In today's European radio channels (DVB, DAB and ADR), programme signals equivalent to 20-bit PCM quantization can be transmitted – with a headroom of 9 dB and without compromising the perceived audio quality. These arguments support the 9 dB EBU headroom as well as the use of QPPM in the broadcast studios, and should result in a much-needed homogenisation of engineering operations and maintenance.

With respect to manual levelling, only specified and correctly calibrated IEC instruments (QPPM) should be used *(see Table 2)*. In order to control the loudness profile within a single programme, an additional loudness meter – such as the algorithm proposed in this article – should be used. The proposed loudness meter, moreover, offers the opportunity to control the loudness profile automatically.

Automatic pre-fading ... and adjusting archive programme material

Because of level and loudness differences in archive material, an accompanying archive (database) of level and loudness correction values would be useful for automatic broadcast operations.

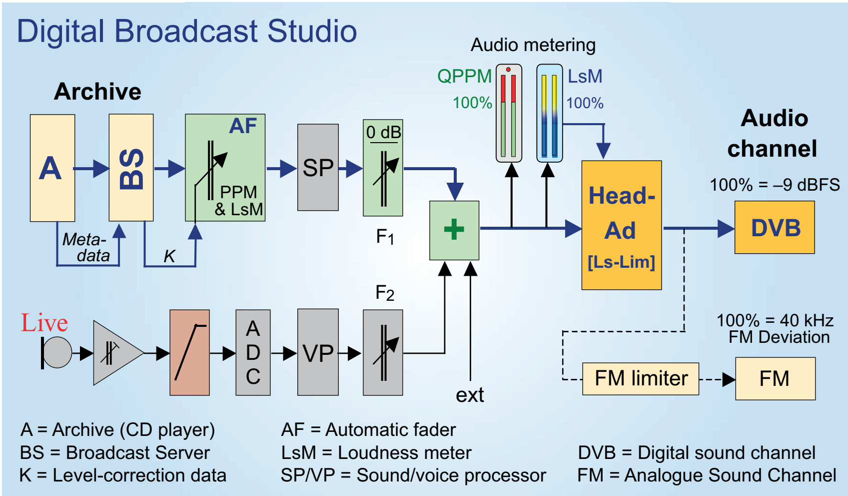


Figure 12
Proposed levelling scheme for digital sound broadcasting

Fig. 12 shows a possible signal processing scheme for *computer-aided radio* (CAR). The archive material is pre-levelled by means of an "automatic fader" (AF). The archive contribution on the "broadcast server" (BS) can be levelled optimally before broadcasting, by means of "level correction" (K) and "loudness correction" (LsM), which is realized by the automatic fader. Controlling all the contributions in the sum channel are the QPPM and the proposed loudness meter (LsM).

Loudness metering

In addition to the 100% tag of QPPM, the loudness meter (LsM) also needs a 100% tag. For optimal levelling of digital sound channels, an additional limit value has to be defined along with the headroom. Unwanted high-level signals could be controlled by means of loudness limitation (Ls-Lim).

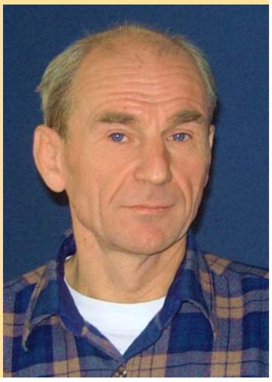
The loudness limiter can be realized by means of an automatic fader that is controlled by the proposed loudness meter. By ensuring that the velocity of the loudness fading matches that of manual fading by a sound engineer, audible distortion could be avoided. This operation could be described as "headroom adaptation".

Conclusions

If the relevant recommendations of the ITU [1] and EBU [2] are met, and the broadcast signal is levelled optimally by means of QPPM [6], a certain loudness balance could be achieved – thus avoiding extreme jumps in loudness. Nevertheless, loudness differences will remain because of diverse recording and audio processing

Gerhard Spikofski studied electrical engineering at Berlin Technical University, one of his main areas of study being technical acoustics. Since 1980, he has been on the scientific staff of the Institut für Rundfunktechnik, Munich (IRT). His field of interest covers development and optimization of audio systems in broadcasting, with special reference to the psychoacoustic aspects.

Dipl.-Ing. Spikofski has published many articles in national and international specialist journals and is a regular speaker at national and international technical conferences. He is also a member of various national and international standardization bodies.



Siegfried Klar studied communications engineering at the academy of Giessen (Germany). Since 1978, he has been on the scientific staff of the Institut für Rundfunktechnik, Munich (IRT). After dealing with video measurement engineering, he changed to the radio broadcast department. In this new working field, he concentrated on problems addressing analogue and digital audio processing and the broadcasting of radio and TV signals.

Dipl.-Ing. Klar's current area of activity covers the analysis and optimization of digital audio broadcasting systems.

techniques. These remaining loudness differences can be controlled by an additional *loudness meter* at the studio output.

In order to achieve loudness balancing of digital audio broadcasts (such as DVB, DAB and ADR), the first step is to meet the 9 dB headroom proposal. The resulting reduction of the available dynamic range is of no consequence to current Digital Radio and TV sound channels – with their quasi 20-bit resolution. As high audio levels cannot be avoided in practice and, at the same time, in order to guarantee an agreed loudness limit, an automatic loudness limiter is suggested – a so called *headroom adapter*. This solution (to avoid clipping of the signal) seems to be preferable to that of using limiters. The automatic controlling of both **level** and **loudness** is achieved by the proposed loudness meter.

Because of the different requirements of archive and broadcast material, it is advisable to distinguish between the levelling of archive and broadcast material. In the case of archive material that will be broadcast, it is highly recommended that this programme material is properly adjusted to suit the new transmission channels available today.

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