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On levelling and loudness problems at television and radio broadcast studios

Siegfried Klar, Gerhard Spikofski

Institut für Rundfunktechnik GmbH

Research and Development Institute of ARD, ZDF, DLR, ORF and SRG/SSR), Munich, Germany
klar@irt.de spikofski@irt.de

ABSTRACT
The problem of partially extreme loudness differences in radio and television programmes has been well known for a long time. With respect to the introduction of new digital techniques combined with parallel transmission of digital and analogue signals the problem of loudness differences again is especially significant. Based upon relevant levelling recommendations and a newly developed loudness algorithm solutions avoiding loudness differences in radio and television are presented.

1. PREFACE

Loudness leaps in radio and television sound increasingly bother listeners. Those loudness leaps are extremely obvious when zapping through radio and television programmes of European digital DVB channels. Within one programme, the transition from a film dialog to a strongly compressed commercial is perceived as particularly bothering. Both under-levelling and over-levelling are observed, resulting in level differences of more than 15 dB.

Among others, the reasons for such levelling and loudness leaps are as follows:

- Apparent obscuration concerning levelling of sound channels
- Usage of particular different and partly unspecified programme level meters
- No standardized studio loudness meter is available until now
- archive material (analogue and digital) is partly not adapted to the regarding sound channel

In most of the analogue radio FM-channels the loudness is mainly balanced by means of compressors and limiters meeting the permissible maximum frequency deviation. This situation seems to be tolerated momentarily.

Considering digital broadcasting it should be possible to achieve balanced loudness profiles when regarding the existing international recommendations of ITU and EBU. This target should be aspired regarding the comparison of different programmes as well as different contributions within one programme.

2. CHARACTERISTICS OF AUDIO PROGRAMME METERS

2.1 ALIGNMENT LEVEL

ITU Recommendation ITU-R BS.645-2 [3] defines the levelling of radio channels by means of the "Alignment signal (1 kHz sinus)". The actual specified level of the sinus signal refers approximately to a full scale programme level regarding loudness.

As the alignment signal has to be considered as a "static" signal, it can be measured by means of usual RMS-meters as well as specified programme meters.

It has to be considered that the analogue "alignment level" and respectively the "nominal" or "permitted maximum level (PML)" is specified diversely considering national and international recommendations **(Tab. 1)**.

Recommendations for analogue & digital Audio Levels	Alignment Level **AL** -9 dBr (35%)	Nominal Level PML*** 0 dBr (100%)
ITU-R BS.645-2 Transmission Level international	**0 dBu***	+9 dBu
ARD HFBL-K Studio Level national	-3 dBu (adaption)	**+6 dBu** (adaption)
US (UK) Reference Level national		+4 dBm (dBu) (adaption)
EBU international digital Transmission & Studio Level	**- 18 dBFS**	- 9 dBFS

***) PML = Permitted Maximum Level

*) 0 dBu = 0.775 V rms (sine) = 1.1 V peak

0 dBFS = Clipping Level (FS = Full Scale)

Tab. 1: Audio levels in transmission and studio environments

Regarding digital audio channels, the relation between the alignment signal and the full scale or clipping level was already specified in 1992 EBU-Rec. R68 [8]. **(Tab. 1)**. Following this recommendation the difference between full scale or clipping level and alignment level has to amount to 18 dB. In other words the alignment level has to be –18 dBFS.

2.2 AUDIO PROGRAMME METERS FOR BROADCASTING

Today a lot of different programme meters are in use at professional studios with widely varying ballistically features.

Whereas in America and Australia mainly VU-meters [5] are used, for European countries "peak programme meters (PPM)" are recommended by the EBU [10]. Those PPM are specified in corresponding IEC-recommendations IEC 268-10 [4] (analogue PPM) and IEC 268-18 Digital PPM [6]. The IEC-category of PPM are
so called "Quasi-peak programme meter QPPM" which neglect short signal attacks due to the human ear. For digital PPM EBU recommends the same ballistcs as described in IEC 268-10 (Type 1).

Since the introduction of digital audio techniques in broadcast, additional but not precisely specified PPM's have caused some confusion.
The following table shows those PPM's which are in use in Europe at the moment.

Besides their different scale-layouts, the PPM primarily vary with respect to their ballistic features described by parameters like "attack time" or "integration time" and "fall back time" or "decay time" **(Tab. 2)**.

Considering the scale-layout the full scale-tag (100%-tag = 0 dB) as well as the specified headroom should accord to "the attack time" of the regarding 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 alignment level has to be smaller than in other cases. Therefore recommendation ITU-R BS.645-2 [3] specifies the 100%-tag to be equal to the alignment level. In contrast to VU-meter the 100%-tag amounts to a 9 dB higher level regarding QPPM.

Remark: The attack time of the PPM, which is used in German broadcast (ARD and ZDF) [15], is specified as 10 ms / 90%. That means it takes 10 ms to reach the 90%-tag. The IEC-type, which is used by the BBC, is slightly different specified as 10 ms / 80%.

Regarding fast digital “sample programme meter SPPM” theoretically no headroom is needed. Those meters are appropriate to control signal peaks with respect to clipping but they are not as suitable as QPPM regarding adequate programme levelling. For example, signals with high proportion of peaks tend to be under-levelled whereas strongly compressed with limited peaks tend to be over-levelled. This can result in grave loudness leaps, which seem to be more intensive than using a QPPM.

The use of unspecified level meters is widely observed in digital audio fields. Mainly if sound engineers are familiar with specified level meters, the use of unspecified devices could result in severe levelling mistakes like clipping and loudness leaps. Because of the wide spread of characteristics of unspecified instruments it is difficult to get familiar and gather experience in levelling.

Digital programme meters frequently are software applications. As known from those applications there are ”infinite” error sources. Whereas the “attack time” normally is almost 0 ms, that means the peak samples are indicated correctly, but there is a wide variation concerning the decay time. Those effects can result in different display as well as in different levelling.

In Germany the QPPM is precisely specified in ARD-Pflichtenheft 3/6 [15]. The specified meter is recommended for levelling analogue and digital signals. Three additional PPMs with 0 ms, 0.1 ms and 1 ms attack time, which are also specified here, should only be used for controlling and not for levelling.

In order to avoid confusion the IRT suggests to adapt the scale layout of digital PPM to the scale layout of the analogue QPPM [4] **(Tab. 2)**. That means the 100%-tag is e. g. 10 dB below full scale.

3. DYNAMIC RANGE OF DIGITAL AUDIO SYSTEMS

3.1 PROGRAMME LEVELLING AND HEADROOM

As already mentioned, the levelling range and necessary headroom depends on the ballistical features of the meter in use. Whereas VU-meters need up to 18 dB headroom, corresponding PPMs only require 9 – 10 dB [3].

The EBU-headroom of 9 dB is strictly combined with QPPM according to [3] and it’s reference to the alignment level specified in [8]. Using instruments with different ballistical features obviously result in other headroom recommendations.

Headroom has to be considered as a buffer range between nominal and clipping level. Meeting the European recommendation, the exchange of programme material is guaranteed without any levelling problems. German broadcasters agreed to this recommendation and specified the headroom in ARD HFBL-K Rec. 15 IRT [16] according to the EBU-recommendation. Regarding analogue signals and devices including A/D and D/A-conversion, the absolute level limit at German broadcast studios is +15 dBu (100%-tag = +6 dBu + 9 dB headroom) **(Fig. 2)**.

3.2 USABLE DYNAMIC RANGE - OBJECTIVE AND SUBJECTIVE CONSIDERATION

Discussing the specifications of headroom and footroom naturally the question whether the resulting system dynamic is sufficient with respect to the human ear has to be answered. In other words, which quantization or how many bits are necessary to guarantee a transmission of music signals without perceivable noise.
To answer this question it can be referred to a responding paper from 1985 [25]. In the following the conditions and results of this study are shortly presented.

Audiobitrate reduction systems, as for example Minidisc (ATRAC), MPEG 1, Layer 2 (MP2) and Layer 3 (MP3), not being introduced during the above mentioned investigation, are not considered in this context. Compared to linear PCM-systems (Pulse Code Modulation) these systems need evidently less quantization. The fact that they nevertheless allow noise-free recordings show that in the cases of bitrate reduction systems other quality parameters have to be considered.

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

The system dynamic, the signal-to-noise ratio or quantization noise can be calculated by means of the following formula

$$S / N\ [dB] = 6n + 2 \quad (n = \text{quantization / bit}).$$

The calculated value – with negative sign – corresponds to the RMS-value of the quatization noise related to 0 dBFS programme level (Full Scale – clipping level of the digital system).

The relation between different measurements are given in the following table. The corresponding absolute values represents the maximum system dynamic in dB related to the given quantizations (reference level = 0 dBFS).

	16 Bit	20 Bit	24 Bit
RMS noise voltage level / dB	-98	-122	-146
Fremdspannungpegel DIN 45 405 / dB [13]	-90	-114	-138
Noise voltage level ITU 468 / dBqps [1]	-86	-110	-134

Considering a headroom of 9 dB [8] and a footroom of 20 dB, as recommended in [24], the determined values of technically usable system dynamic are shown in **Fig. 9** as a function of quantization.

In principle the reference values of the system dynamic – the maximum full scale programme level on one hand and the system noise level on the other – correspond to certain sound pressure levels of the reproduction of music signals. With respect to the necessary quantization, the relevant sound pressure levels are the maximum listening level and the just unperceptible noise level.

In order to control the great number of influencing factors of the psychoacoustic experiments the following boundary conditions were defined.

Both aspects – determination of the maximum listening level and noise perception limits – were investigated separately in spite of the fact that they are coupled as system features. That means the selected music items (female / male speaker, orchestra, string quartet, rock music) were only used to determine the maximum listening level. On the other side representative noise signals (idle channel noises) (PCM 1610 – CD-Mastering (Sony), PCM-Tonkanalsystem MSt13 (Siemens) and white noise) were investigated without programme signal. That means that the disturbing noise was only assessed in music pauses without considering cover up effects caused by the programme signal.

The experiments were carried out with 20 normal hearing subjects as individual single sessions. The listening set-up meet the corresponding requirements on professional listening conditions including headphone reproduction [2, 12].

The results of the investigation, which included stereo loudspeaker and headphone reproduction are presented in **Fig. 9** The determined reference values correspond to the 90% values of the cumulative frequency distribution of the maximum listening level / dBA and the average values of the individual perception limits of the investigated system noises.

In the left part of **Fig. 9** the relation between quantization and system dynamic regarding linear PCM-systems (16 bit, 20 bit, 24 bit) is shown. In each case the recommended headroom of 9 dB and footroom of 20dB is considered. If also reserving a useful footroom in the case of reproduction in order to consider for example individual deviations, the results in **Fig. 9** show that a linear 16 bit-system like CD just meets the requirements of the human ear regarding loudspeaker reproduction. In the case of headphone reproduction the requirements are only met when relinquishing a headroom – as however usual in todays CD production. As a consequence for digital audio studio production where headroom and footroom is essential the presented results show that professional audioproduction needs at least 20 bit systems.

4. PROGRAMME LEVELLING AND LOUDNESS

4.1 PROGRAMME LEVELLING

Levelling is widely understood as optimal programme signal adjustment to the corresponding transmission channel. This adjustment is controlled by means of a level meter e. g. QPPM aiming that the maximum programme levels almost meet but don't exceed the 100%-tag. In German broadcast the level meter QPPM according to IEC 268-10 [4] is standardized regarding analogue and digital signals. Meeting the 100%-tag, which implies a 9 dB headroom, guarantees a transmission free of distortions. That naturally doesn't mean that no amplitudes greater than 100% occur. Those short level peaks, which are kept invisible by the sound engineer, won't produce clipping because a sufficient headroom of 9 dB is provided. The recommended levelling reserve results from extensive programme signal analysis [23]. It was also observed that the characteristics of QPPM and identically levelling result in a certain loudness balance however not in loudness identity.

4.2 PROGRAMME LOUDNESS

As generally known the same levelling of different programme signals does not normally result in the same loudness impression. This discrepancy is especially evident considering music and speech. In order to reach a certain balance of mixed broadcast programmes with respect to loudness special levelling recommendations were defined based on detailed Investigations [18, 21, 22].

Meeting these recommendations in mixed speech/music programmes (magazines, car programmes, commercials) when speech is most important, speech should be levelled to 0 dB and music between –8 dB to –4 dB.

Those recommendations are useful to avoid extreme loudness differences of broadcast programmes. Loudness adaptation due to the human ear can't therefore be achieved. This is particularly true when using special audio processors. Adapting the loudness of broadcast programmes to the human ear, an additional loudness meter is necessary besides the level meter controlling the technical levelling.

Although some investigations had been carried out in this field [18, 19, 20, 26] no standardized loudness meter is available at the moment. Loudness corrections today therefore have to be done manually by the control engineer. This is of course not practicable whereas most of control functions are done automatically.

New investigations however show, that a corresponding studio loudness meter seems to be realizable [26].

In the mentioned study the performance of available studio loudness meters and specially new loudness algorithms based on measuring the signal level and signal power were investigated.

In detail the following methods were tested:

- Loudness measurement RTW [18]
- Loudness measurement EMMET [19]
- Signal level PPM (=QPPM)
- Signal power level PWR

The study deals with the subjective and objective aspects of loudness measurements. On one side psychoacoustic measurements were carried out to determine the subjectively perceived loudness of the selected broadcast programme material. On the other side objective measurements were aimed for deriving relevant signal parameters which allow to define objective loudness. The performance of the tested objective parameters were assessed referring to the correlation between subjective loudness an the tested objective parameters.

The used test material were recordings of DSR (Digital Satellite Radio) with 16 stereo radio programmes in 1984. Each of the 16 programmes was represented by typical items of about 15 s length. The presented 56 items contained announcements, orchestra, chamber music, piano, vocal music, pop music. This digitally broadcasted and received material can be considered as representative for the actual radio programme especially with respect to levelling and audio processing.

In order to derive relevant objective parameters for each of the loudness algorithms and programmes, item level histograms were analysed. In each case the cumulative frequency distribution was considered specifying the exceeded level with respect of 10%, 30% and 50% of the individual item time **(Fig. 10)**.

In the following each of the loudness algorithms are shortly characterized by means of the corresponding level versus time diagrams.

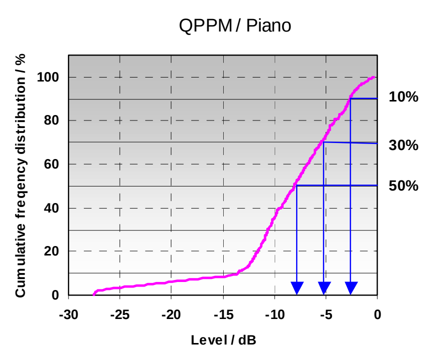

Fig. 10: Analysing cumulative frequency distribution of objective loudness measures

LOUDNESS MEASUREMENT RTW

The loudness measurement RTW [18] is based on filtering of the programme signal (comparable to 80 dB curve of recommendation ISO R26) and measuring of the filtered signal by means of the studio level meter according to [15] with 10 ms integration time **(Fig. 11)**.

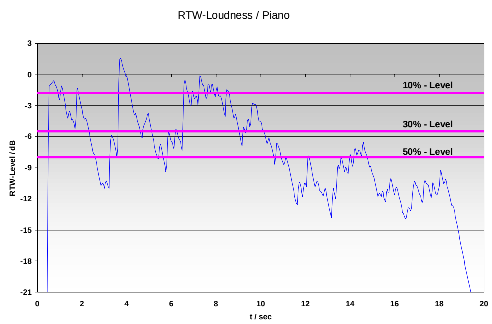

Fig. 11: RTW-loudness and considered cumulative frequency rate

LOUDNESS MEASUREMENT EMMET

The measurements were done with the loudness meter “Chromatec AM-2L” (Michael Stevens & Partner LTD). The algorithm itself is based on EMMET’s investigations [19, 20] including level analysis and human ear adapted filtering **(Fig. 12)**.

SIGNAL LEVEL QPPM

The measurement of the QPPM is done with the level meter according to ARD-Pflichtenheft Nr. 3/6 with 10 ms integration time and a release time of 1.5 s [15] **(Fig. 13)**.

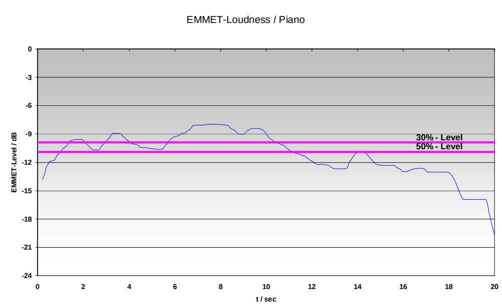

Fig. 12: EMMET-loudness and considered cumulative frequency rate

SIGNAL POWER LEVEL PWR

The signal power level was calculated samplewise by means of corresponding computer software. The tested parameters are the average values of the power level considering time intervals of 100 ms, 1 s and 2 s **(Fig. 14)**.

The results of the study are presented in **Fig.15** including all programme items (anonymously indicating the methods by the letters A-F).

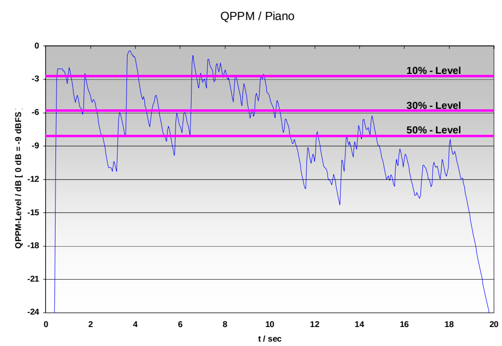

Fig. 13: QPPM-Level and considered cumulative frequency rate

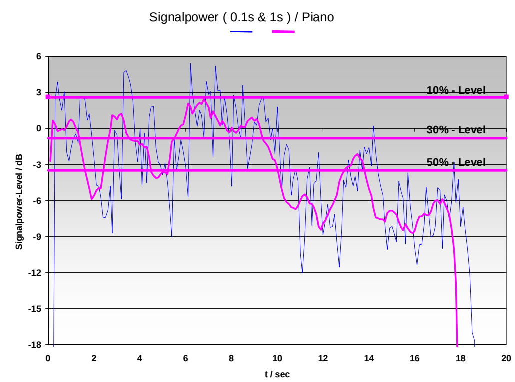

Fig. 14: Signalpower (averaging time 0.1ms, 1s) and considered cumulative frequency rate

The criterion for assessing the performance of the tested loudness algorithms is the Spearman-Rankcorrelation between subjective and objective loudness measurements. Whereas the subjective loudness is represented by the average values of the psychoacoustic loudness experiments, the corresponding objective parameters are the exceeded levels with rates of 10%, 30% and 50% in each case.

Considering the results in **Fig. 15** with respect to the cumulative frequency distribution, it can be shown that the 50% level – the levels being exceeded in 50% of the time – displays the highest correlation in each case.

Considering only the 50% values a correlation >67% is achieved in each case, whereas method A and B display the highest correlation (78%).

Because of the high correlation coefficients regarding method A and B and because of relatively small deviations between subjective and objective loudness parameters with respect to the individual results [26], these two methods are a good basis for developing the studio loudness meter. It is interesting that both methods work according to the level meter specified in [15] with 10 ms integration time.

In order to optimise the defined loudness algorithm based upon the recommended level meter [4] with 10 ms integration time and 1.5 s release time additional measurements were carried out. Among others the "exceeding level frequency" (60%, 70% and 80%) and "analysing time" (1 s, 3 s, 5 s, 7 s and 10 s) were tested. The corresponding results are presented in **Fig. 16-18**.

After optimising the tested parameters, the resulting correlation between subjective and objective loudness amounts to 90%. The individual results of subjective and objective loudness are presented in **Fig. 18** with additional indication of the average values and 95% confidence intervals of the subjective loudness levels.

Based upon the presented results the corresponding loudness algorithm was defined and the prototype of the defined studio loudness meter was developed. At the moment this prototype is under test with special respect of praxis performance problems.

5. PROGRAMME ANALYSIS OF DVB-CHANNELS

In order to analyse audio signals the following methods are considered to be appropriate:

AMPLITUDE STATISTICS

Analysis of the cumulative frequency distribution of audio samples. The form of the presented diagrams (Signal amplitude versus probability of amplitude exceedings) give interesting information about loudness and compression features of the analysed signal **(Fig. 3-4)**.

LEVEL REGISTRATION VERSUS TIME

Registration of normally displayed levels e. g. QPPM, SPPM, PWR 1s, QPPM-Loudness LsM and later evaluation of programme signals **(Fig. 5-7)**.

LEVEL STATISTICS

Analysis of the cumulative frequency distribution of various displayed levels.

The presented results refer to recordings of DVB audio-channels during January and February 2002 and are representative for all digital broadcast channels.

The main results of the carried out measurements are as follows **(Fig. 3-4)**:

- Extreme differences between radio and tv sound-channels and mismatching relevant ITU-recommendations [3]
- Clipping including audible distortions regarding one specific radio channel
- In few cases of radio channels the available headroom is not used because unadjusted limiter features
- Regarding one specific tv-channels peaks incomprehensively are not limited to the 100%-tag but to the alignment level.

6. PROGRAMME AND LOUDNESS LEVELLING IN DIGITAL SOUND BROADCAST

6.1 GENERAL ASPECTS

Digital radio offers the chance to get rid of those constraints which are well known from analogue FM-radio. In digital radio there is no relation between loudness transmission range requiring corresponding audio procession. Therefore the provided dynamic range of digital radio can be reasonably used, e. g. to broadcast full dynamic range of excellent CD-recordings.

First of all transmitters have to be levelled correctly according to ITU-Recommendation [3]. Thus extreme loudness leaps could be avoided. In todays European radio channels DVB, DAB and ADR, programme signals with equivalent 20 bit PCM-quantization can be transmitted so that a headroom of 9 dB practically doesn't mean any perceivable quality restriction. These arguments underline the 9 dB EBU-headroom as well as the use of QPPM in the broadcast studios and would result in a particulary desired homogenisation of engineering operations and maintenance.

With respect to manual levelling, only specified and correctly calibrated IEC-instruments QPPM should be used **(Tab. 2)**. In order to control the loudness profile within one programme an additional loudness meter according to the proposed algorithm of this paper should be used. The proposed loudness meter moreover gives the opportunity to control the loudness profile automatically.

6. 2 AUTOMATIC PRE-FADING ADJUSTING ARCHIVE PROGRAMME MATERIAL

Because of level and loudness difference of archive material, an accompanying archive of level and loudness correction values would be useful regarding automatic broadcast operations.

Fig. 8 shows possible signal processing of “computer-aided-radio (CAR)”. As shown in **Fig. 8** the archive material is pre-levelled by means of an “automatic-fader (AF)”. The archive contribution on the “broadcast-server (BS)” can be optimally levelled before broadcasting by means of the level correction (K) and loudness correction (LsM), which is realized by an automatic fader (AF). Controlling all contributions besides the QPPM the proposed loudness meter is provided (LsM) in the sum channel.

6.3 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 besides the headroom. Unwanted high level could be controlled by means of loudness limitation (Ls-Lim).
The loudness limiter can be realized by an automatic fader, which is controlled by the proposed loudness meter. Ensuring that the velocity of loudness fading matches manually fading of a sound engineer no signal audible distortion could be avoided. The presented operation could be described as “headroom adapter” **(Fig. 8)**.

7. CONCLUSION

If the corresponding recommendations of ITU [3] and EBU [8] are met and the broadcasted signal is optimally levelled by means of QPPM [4] a certain loudness balance could be achieved avoiding extreme loudness leaps. Nevertheless loudness differences will remain because of diverse recording and audio processing techniques. These remaining loudness differences can be controlled by an additional loudness meter at the studio output.

Achieving loudness balancing of digital audio broadcast such as DVB, DAB, and ADR, the first step is to meet the proposed headroom agreement of 9 dB. Considering todays digital radio and television sound channels with quasi 20 bit resolution the resulting reduction of available dynamic range has practically no evidence. As high levelling can not practically be avoided in broadcast operations, and at the same time to guarantee an agreed loudness limit, an automatic loudness adaptation is suggested by means of a so called “headroom adapter”. This solution in order to avoid clipping of the signal seems to be preferable compared with using limiters. This automatically controlling of level and loudness is achieved by the proposed loudness meter. Because of different requirements concerning archive and broadcast material it is consequently advisable to distinguish between levelling of broadcasting and production archiving respectively. It is urgently recommended in the case of broadcasting archive material to adjust the programme to the transmission channel.

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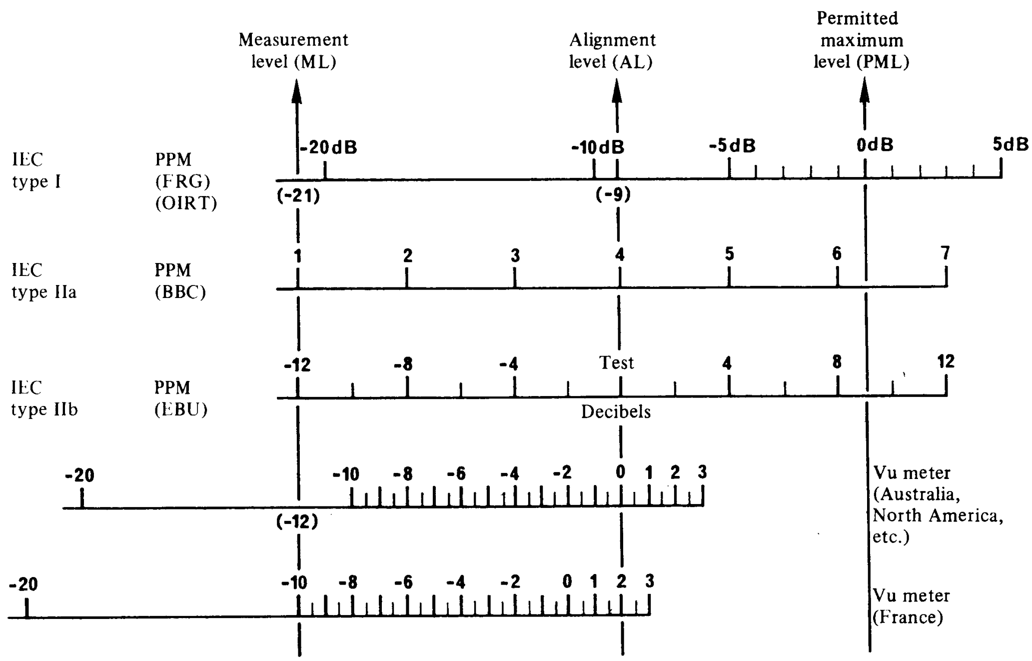
FIGURE 1

Indications produced by various types of programme meter with the recommended test signals



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Note 1 – Meter indications are schematic – not to scale.

Fig. 1: Indications produced by various types of programme meters

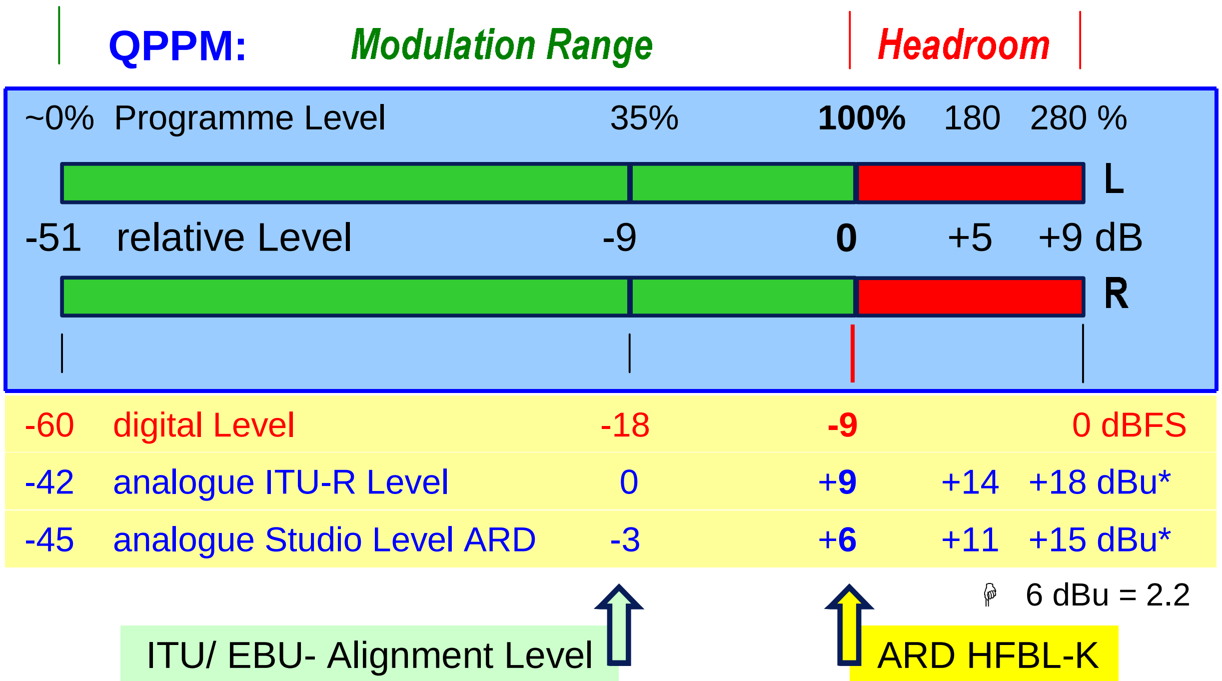

Fig. 2: Proposed Peak Programme Meter (PPM) scale for analogue and digital audio

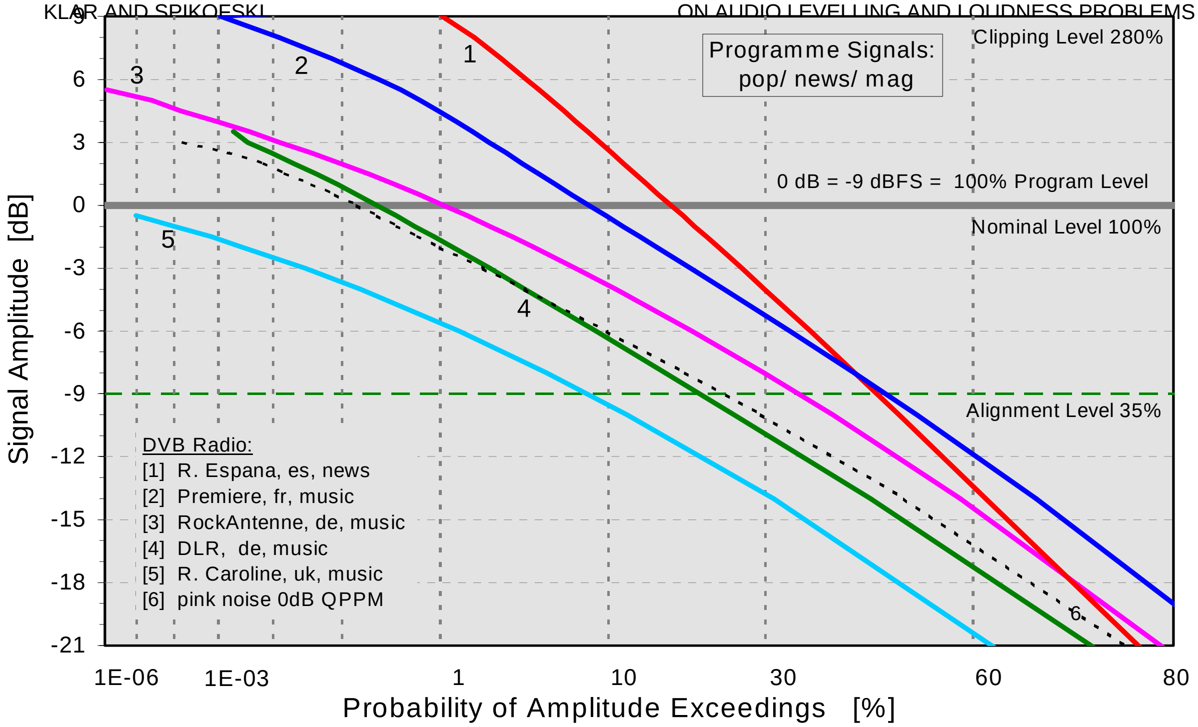

Fig. 3 : Amplitude statistics of DVB-Radio

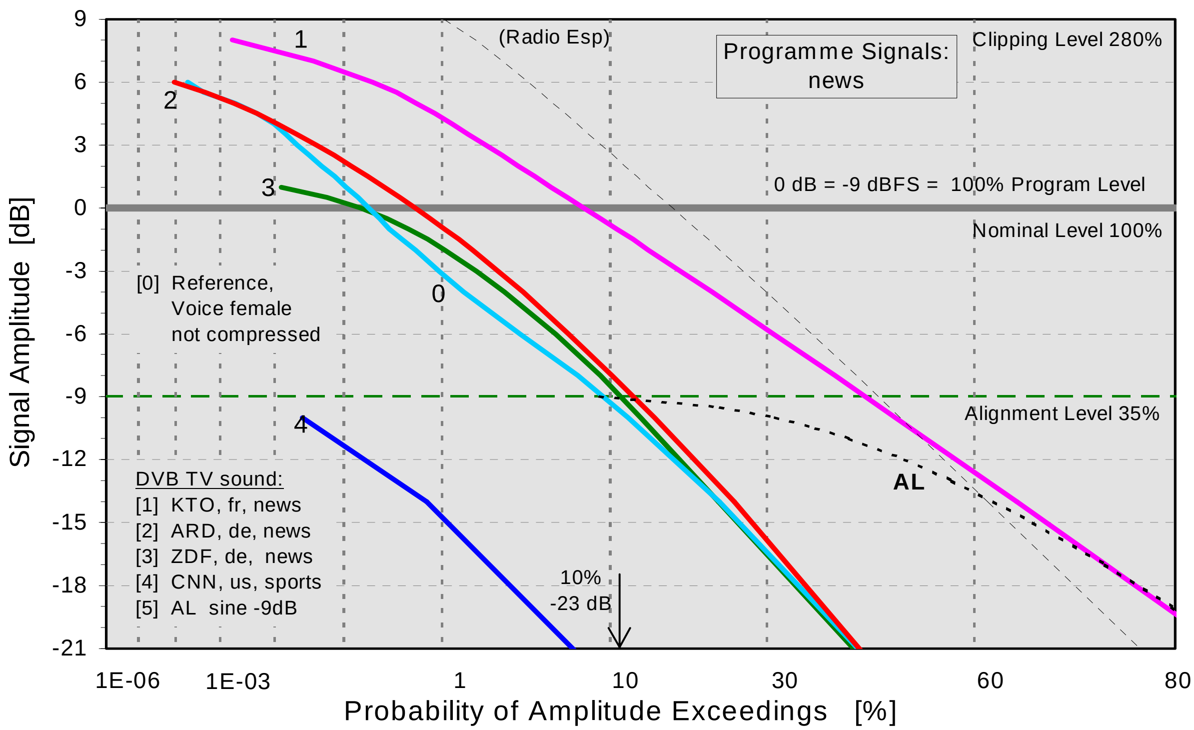

Fig. 4: Amplitude statistics of DVB TV-Sound

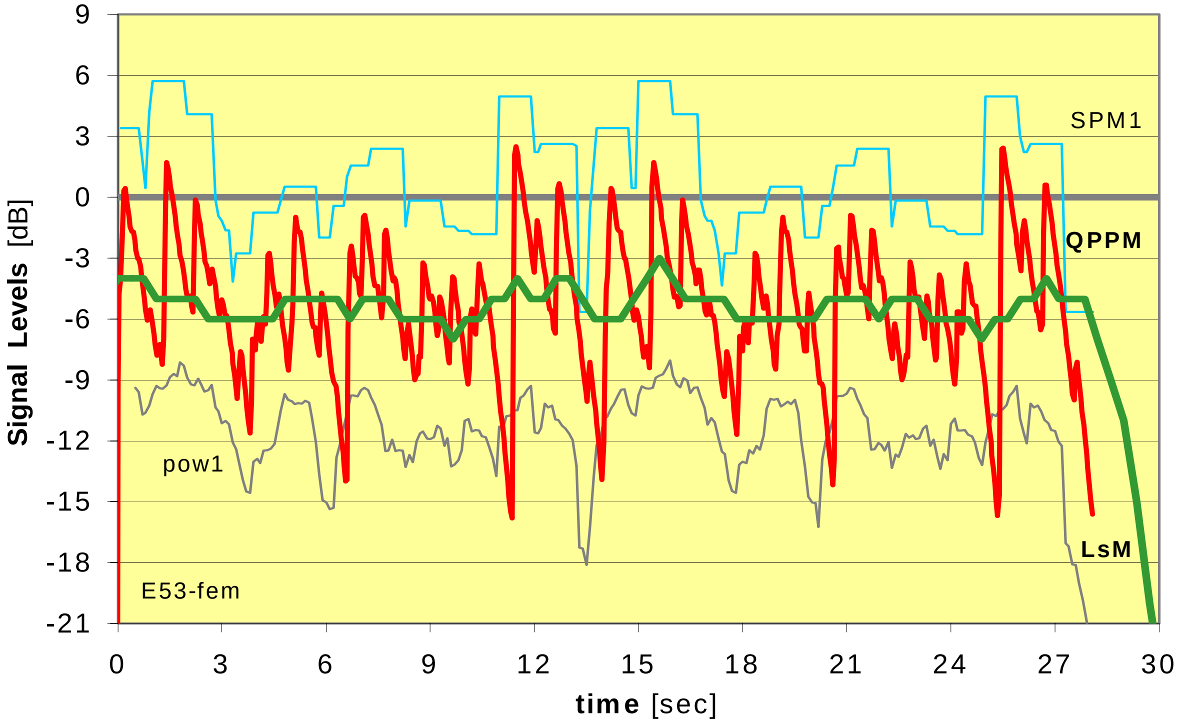

Fig. 5: Programme Levels (QPPM, SPPM, PWR 1s, QPPM-Loudness LsM) - Female speaker (no compression)

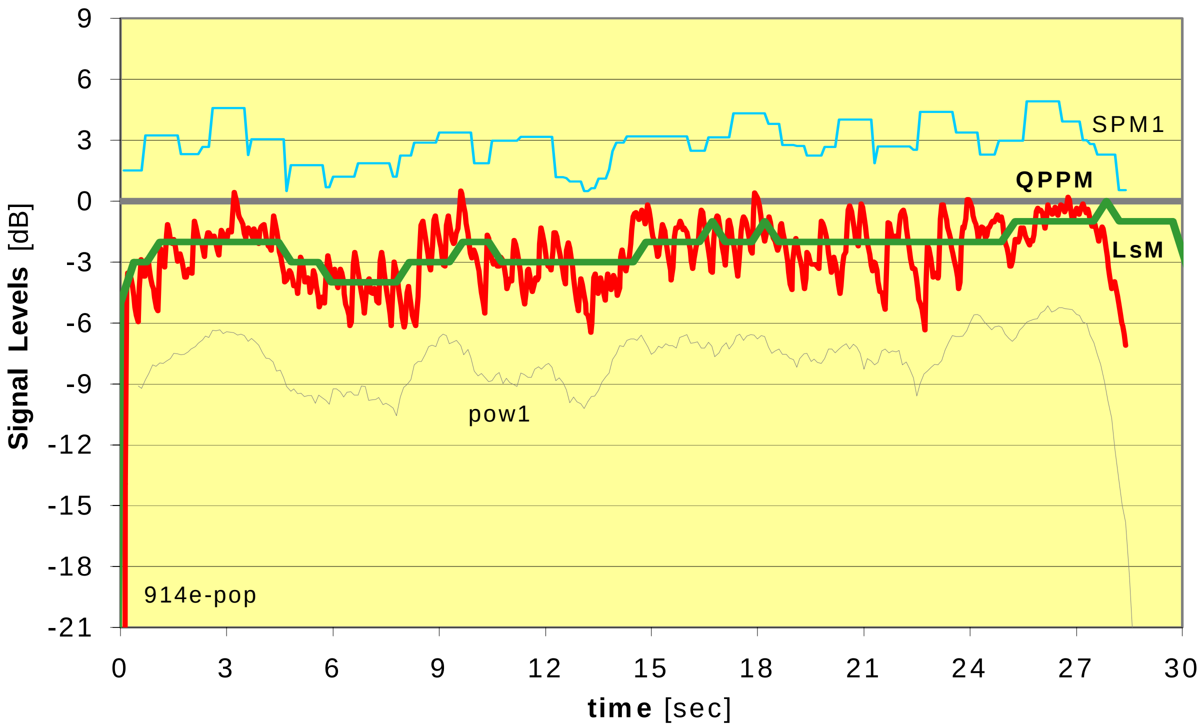

Fig. 6: Programme Levels (QPPM, SPPM, PWR 1s, QPPM-Loudness LsM) - Pop music (light compression)

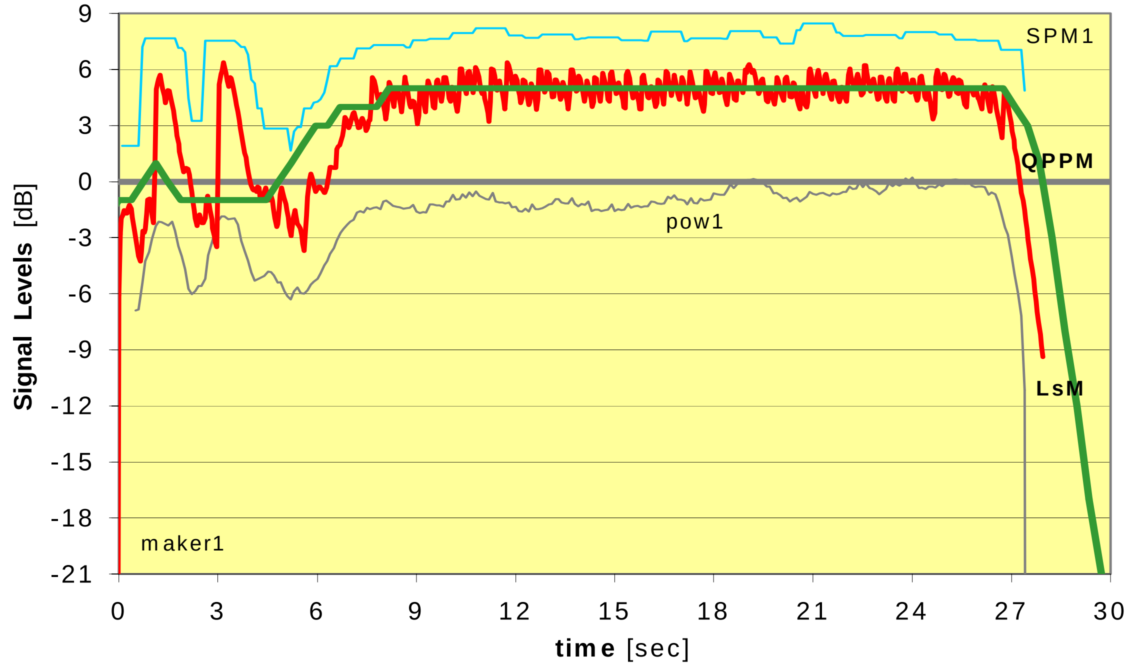

Fig. 7: Programme Levels (QPPM, SPPM, PWR 1s, QPPM-Loudness LsM) - Pop music (high compression)

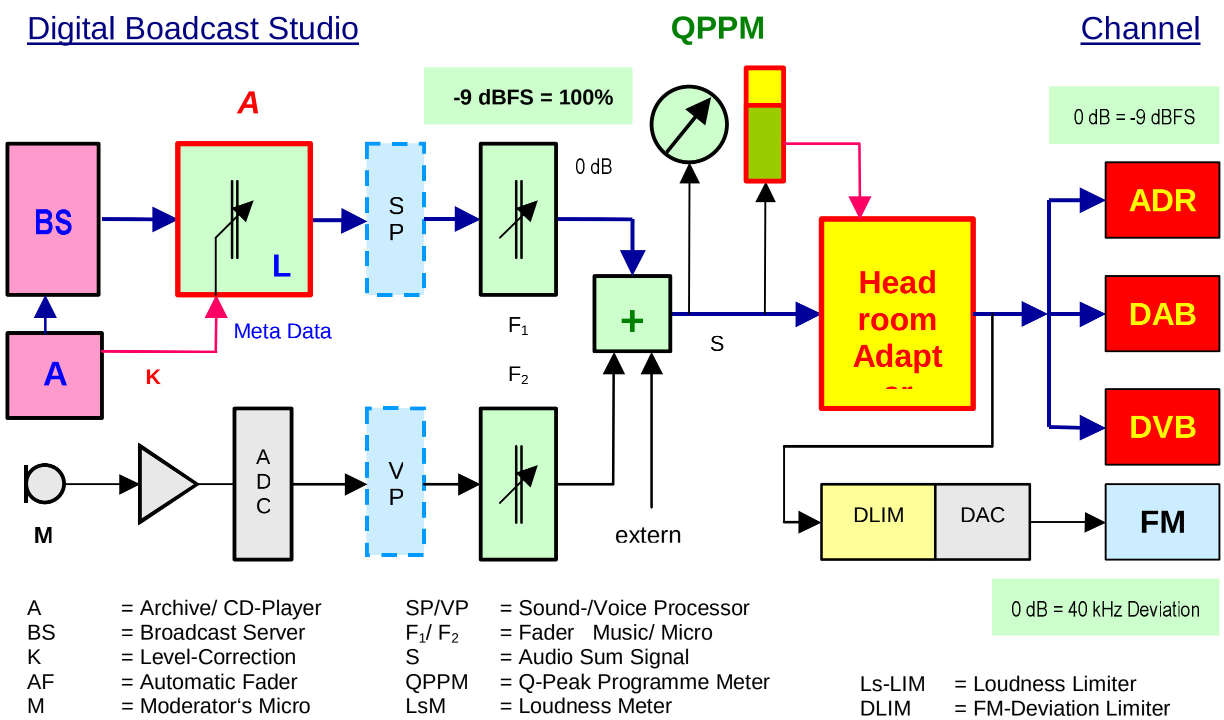

Fig. 8: Proposed levelling scheme of digital broadcast sound channels

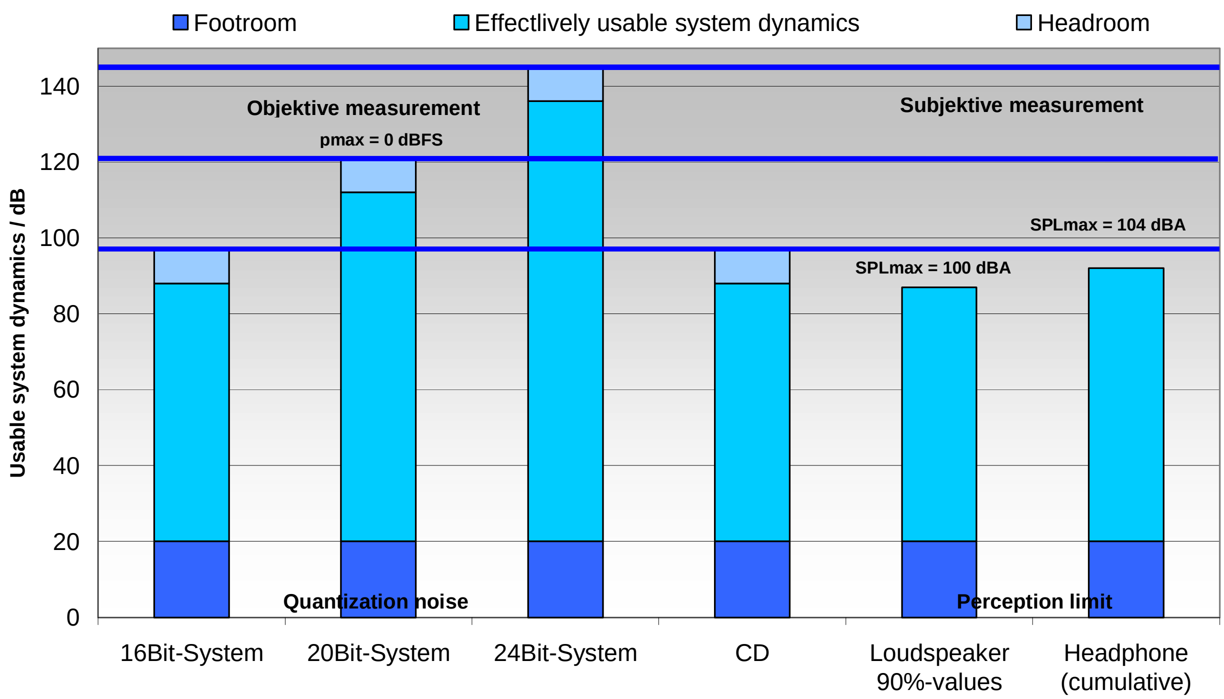

Fig. 9: Usable dynamic range of digital PCM-systems - Subjective and objective consideration

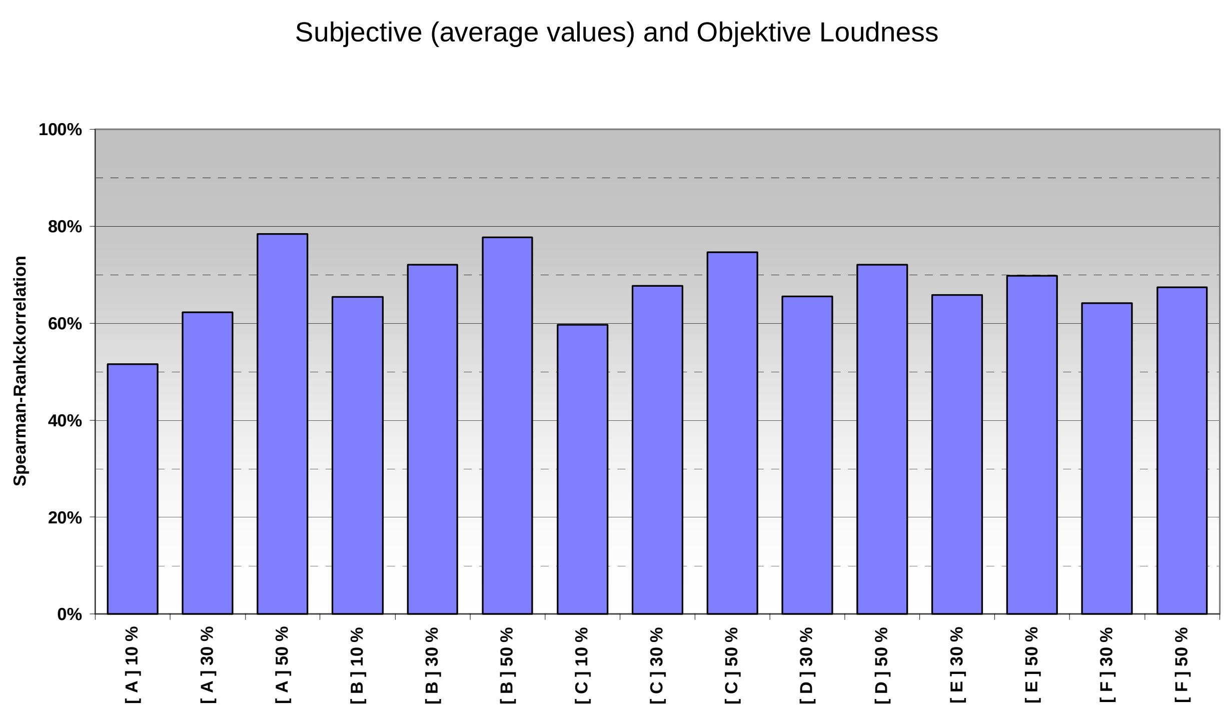

Fig. 15: Spearman-Rankcorrelation between subjective (average values) and tested objective loudness parameters

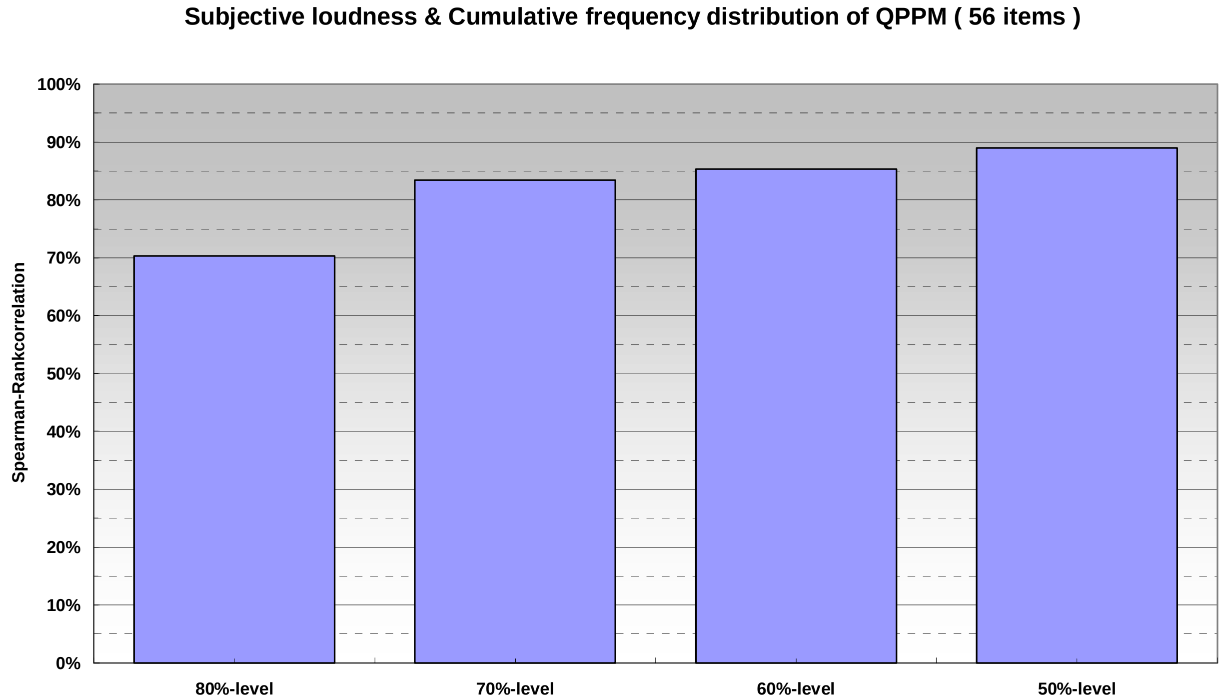

Fig. 16: Correlation between subjective and objective QPPM-Loudness - Variation of cumulative frequency rate

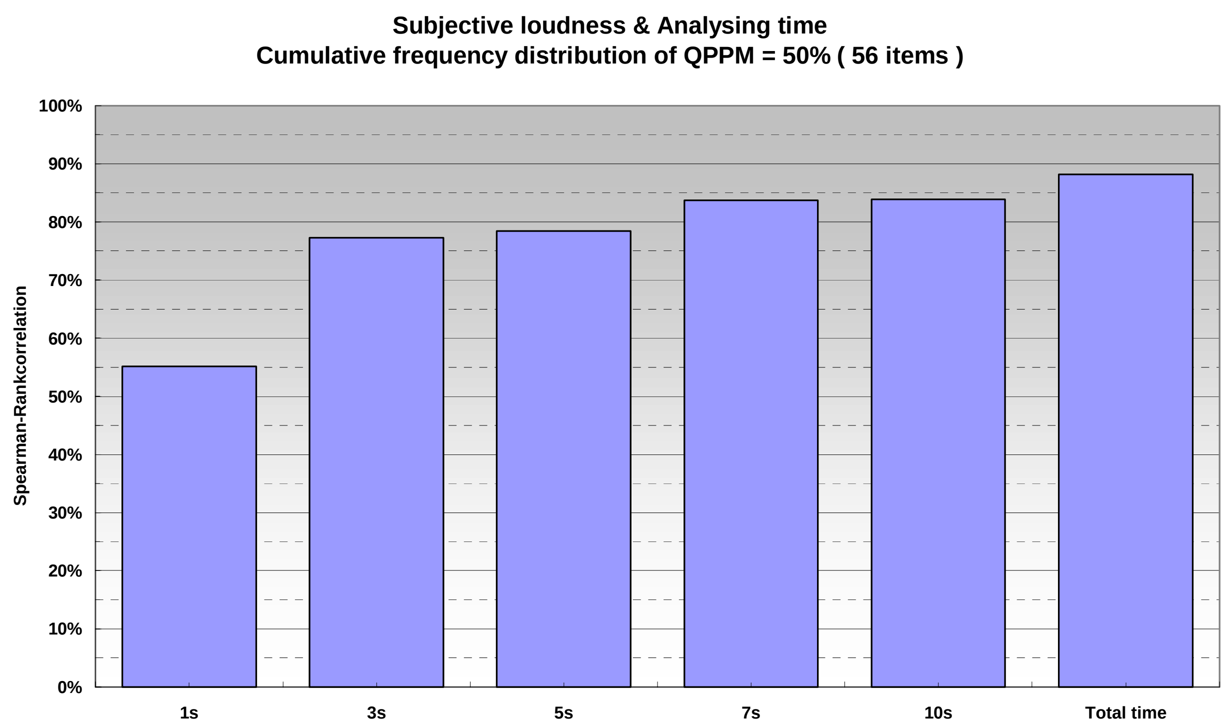

Fig.17: Correlation between subjective and objective QPPM-loudness – Variation of analysing time

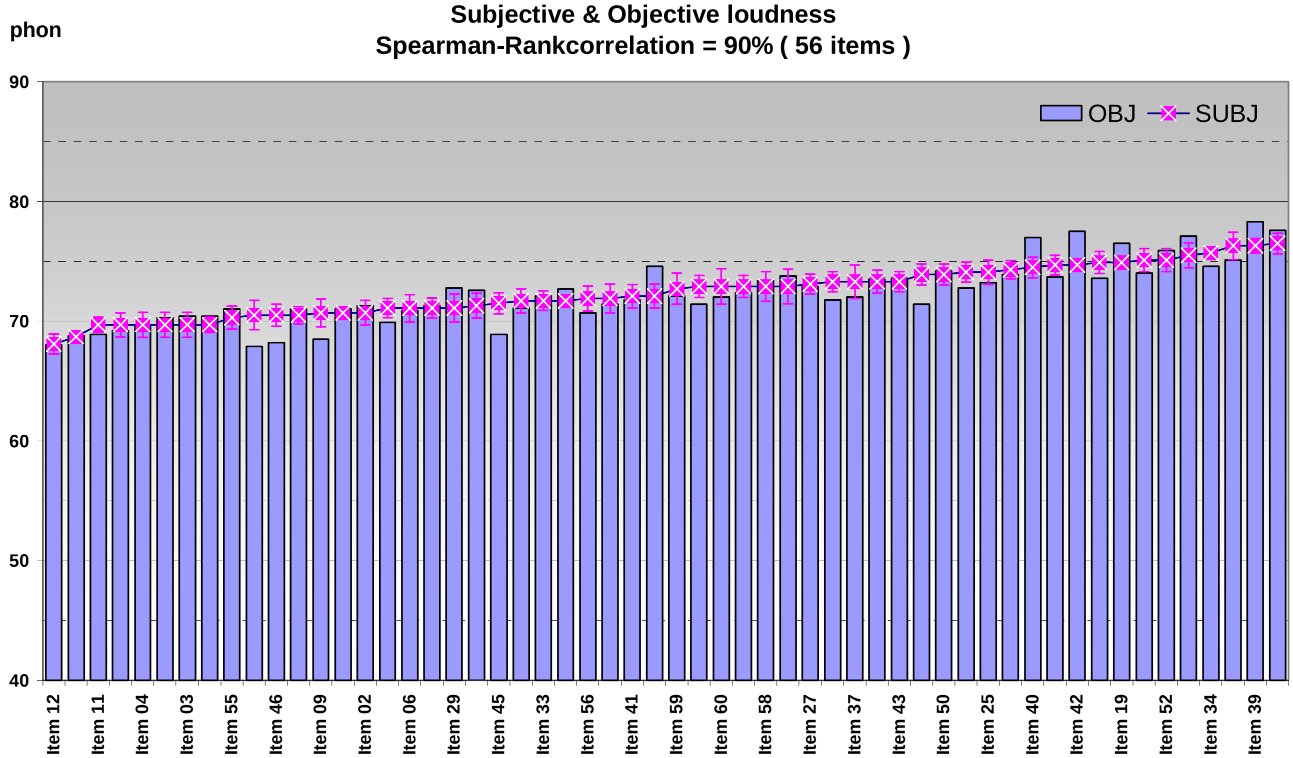

Fig. 18: Subjective (average and 95%-confidence interval) and objective QPPM-loudness

Programme Meter Type (PM)	Recommen- dation	AL (35%)	**PML***** **100%**	Clipping Level	Scale	unvisible peaks	attack time (integration)	decay (fall-back
VU Meter	ANSI C 16.5 IEC 268-17	0 VU = +0 dBu	0 VU		-20 to +3 [dB]	+13...+16 dB	300ms/ 90%	300ms/ 10%
DIN PPM (QPPM)	DIN 45406 IEC 268-10/ I	-9 dBr = +0 dBu	**0 dBr** = +9 dBu	+16 dBr =+22	-50 to +5 [dB]	+3...+4 dB	10 ms/ 90% **5 ms/ 80%**	20 dB/ 1.5s = 13 dB/s
BBC PPM (QPPM)	 IEC 268-10/	'4' = +0 dBu	'6' = +8 dBu	 =+24	1 to 7 [ ]	+4...+6 dB	10 ms/ 80% 20 ms/ 90%	24 dB/ 2.8s = 8.6 dB/s
EBU PPM (QPPM)	EBU 3205 E IEC 268-10/	+0 dB = +0 dBu	**+9 dB** = +9 dBu		-12 to 12 [dB]	+4...+6 dB	10 ms/ 80%	24 dB/ 2.8s = 8.6 dB/s
EBU PPM digital (QPPM)	EBU IEC 268-18	-18 dBFS = +0 dBu	**-9 dBFS**	0 dBFS	-40 to +0 [dB]	+3...+4 dB	5 ms/ 80%	20 dB/ 1.7s = 12 dB/s
digi PPM IRT Proposal	IRT/ IEC 268-18	-9 dBr 35%	**+0 dBr** **100%**	+9 dBr	-50 to +10 [dB]	+3 ...+4 dB	**5 ms/ 80%**	20 dB/ 1.7s = 12 dB/s

***) Permitted Maximum Level (PML): 100% Modulation = +9 dBu = -9 dBFS for Transmission Lines (ITU-R BS.645-2 & EBU-R.68) ≡ +6 dBu ARD Nominal Studio Level (ARD HFBL-K Rec.15 IRT)

Table 2: Types of programme meters in international transmission environments