EBU – TECH 3344

Practical guidelines for distribution systems in accordance with EBU R 128

Supplementary information for EBU R 128

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Conformance Notation

This document contains both **normative** text and **informative** text.

All text is normative except for that in the Introduction, any § explicitly labelled as 'Informative' or individual paragraphs that start with 'Note:'.

Normative text describes indispensable or mandatory elements. It contains the conformance keywords 'shall', 'should' or 'may', defined as follows:

'Shall' and 'shall not':	Indicate requirements to be followed strictly and from which no deviation is permitted in order to conform to the document.
'Should' and 'should not':	Indicate that, among several possibilities, one is recommended as particularly suitable, without mentioning or excluding others. OR indicate that a certain course of action is preferred but not necessarily required. OR indicate that (in the negative form) a certain possibility or course of action is deprecated but not prohibited.
'May' and 'need not'	Indicate a course of action permissible within the limits of the document.

Default identifies mandatory (in phrases containing "shall") or recommended (in phrases containing "should") presets that can, optionally, be overwritten by user action or supplemented with other options in advanced applications. Mandatory defaults must be supported. The support of recommended defaults is preferred, but not necessarily required.

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A conformant implementation is one that includes all mandatory provisions ('shall') and, if implemented, all recommended provisions ('should') as described. A conformant implementation need not implement optional provisions ('may') and need not implement them as described.

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Practical guidelines for distribution systems in accordance with EBU R 128

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1. Scope

This document represents practical guidelines for broadcast distribution. The guidelines are meant:

- To specify relevant settings and processing in the signal chain from the studio to consumer equipment.
- To encourage the adaptation of non-compliant broadcasts to become compatible with EBU Recommendation R 128 [1].

The following parties are encouraged to comply with this set of guidelines in order to allow interoperability between EBU Members' broadcasts that follow the EBU R 128 loudness normalisation recommendation and to allow consistency of playback on consumers' equipment:

- Content distributors — Companies that transmit radio and television via cable, satellite, terrestrial, IPTV or other means.
- Broadcast electronics equipment industry — Makers of audio and video distribution devices, professional integrated receiver decoders, measurement tools and loudness level adaptors.
- Consumer electronics equipment industry — Manufacturers of home theatre playback devices (such as AV-receivers), television sets and set-top boxes.

2. Guidelines for broadcasting and rebroadcasting of television and radio services

2.1 Objectives and basic principles

The goal of these guidelines is to achieve loudness normalisation between services in the distribution stage of the radio and television broadcast chain and to achieve loudness level equalisation between systems and interfaces of consumer equipment for radio and television reproduction. The objective is to scale down loudness differences to a level where consumers can safely switch from one channel to another and from one audio system to another without annoyance. To achieve consistent loudness levels throughout facilities, distribution and transmission networks and, ultimately, for the listener, the whole broadcast chain of production, play-out and distribution needs to be included in the scope. This document includes requirements and recommendations for set-top boxes, television sets and home theatre equipment, levels for encoding, decoding and modulation and it introduces lossless loudness normalisation in the distribution stage. The aim is to provide a consistent, higher quality of sound and thus a more pleasant listening experience for the audience.

The following basic principles apply to loudness normalisation in accordance with EBU R 128:

- Determination and consistency of the dynamic range properties of a radio or television service for specific transmission platforms are considered to be the responsibility of the broadcast stations.

  Consequently, a major restriction is that the dynamic range properties of a rebroadcasted service shall not be changed in the distribution stage unless it is strictly necessary for technical reasons, specifically to adapt the signal to the limitations of the distribution system, for example to prevent overloads.

- Adaptation for frequency-modulated and other analogue or pre-emphasis based transmission systems is considered to be the responsibility of the distribution companies.

  Consequently, it is recommended that processing for these kinds of systems be moved from the studio to the distribution stage of the broadcast chain, which thereby avoids needless limitation of the digital transmission systems.

- Specifications for set-top boxes and integrated digital television sets that are required to assure optimum and undistorted performance of EBU Members' broadcasts are considered to be the joint responsibility of the organisations which specify the requirements for the distribution and reception systems.

  Consequently, distribution companies are encouraged to follow the guidelines which support EBU R 128 in their own specifications and operational workflow and to actively support the aim to achieve loudness normalisation in radio and television broadcasting.

- Loudness equalisation between services is considered to be the objective of the distribution companies.

  Consequently, it is recommended that active normalisation be applied in distribution systems. It must be emphasised that it is essential to apply loudness normalisation on all services. By doing this, the quality of experience for all distributed content is improved for the benefit of the audience, while the EBU process for loudness normalisation is actively and sustainably supported. The added significance of loudness normalisation in the distribution stage is that it eliminates the motive for broadcast stations to compete on loudness and that it protects the position of the stations which have implemented EBU R 128 in their workflow.

Distribution companies intending to apply EBU R 128 in their internal workflow shall contact EBU headquarters in Geneva (please check http://tech.ebu.ch/loudness for details). The EBU shall subsequently consult its local member (where available) and may give access to support information. This procedure is essential to ensure that there is mutual understanding about the correct implementation of EBU Tech 3344. In this document it is assumed that this process is followed.

2.2 Loudness normalisation of television services

It is assumed in this guideline document that EBU Members' broadcasts will be transmitted in accordance with EBU R 128. However, other broadcast stations within and from outside Europe, thematic services, play-out systems, locally-inserted advertisements, set-top box interactive applications and local broadcasting stations can use other Target Levels that result in annoying inconsistencies in loudness when switching from one service to another. To overcome this problem, lossless loudness normalisation shall be applied in the distribution stage. For file-based systems this can be done by, for example, an algorithm implemented in software. For live broadcasting, normalisation shall be based on continuous loudness measurements over a full day (24 hours).

In the following cases a correction shall be applied:

- If the long-term loudness level(1) of MPEG-1 Layer II services of a particular broadcast station deviates from the Target Level of -23 LUFS described in EBU R 128.
- If the long-term loudness level of HE-AAC services of a particular broadcast station deviates from the Target Level of -23 LUFS described in EBU R 128, based on the application of the Decoder Target Level descriptor (target_level, specified in ISO/IEC 14496-3 [2]) at a level of -23 LUFS.
- If the long-term loudness level of Dolby Digital (Plus)(2) services of a particular broadcast station deviates from the Sound Reproduction Level as described in ETSI TS 102 366 [3], based on the assumption3 that the Decoder Target Level is equivalent to a loudness level of -31 LUFS.

Note 1: For a description of 'long-term loudness level' see § 3.2.

Note 2: Dolby Digital (Plus) is also identified in this document as DD/DD+.

Note 3: This assumption is also applied in the rest of this document. Instead of referring to a signal reference level of -31 dBFS as described in section E.4.3.1 of ETSI TS 102 366, this document refers to a loudness level of -31 LUFS as Decoder Target Level. This level is achieved if the absolute value of the 'Dialnorm' descriptor of the DD/DD+ bitstream equals the Programme Loudness Level according to EBU R 128.

Services compliant with EBU R 128 will consequently experience no adjustment and may, after consideration, even be excluded from active compensation. It is recommended, however, to leave active compensation enabled. If not, it is recommended that continuous measurements be performed on these services in order to monitor if the situation has changed. Direct consultation between the distributor and the broadcast station, particularly in case of deviations or complaints, is always recommended.

2.3 Loudness normalisation of radio services

Distributors that transmit radio services as well as television services shall apply the same normalisation and alignment process to radio as for television. In that way, loudness inconsistencies are avoided when switching between radio services and also when switching from television to radio mode and vice-versa on a set-top box or similar receiver device. By applying loudness normalisation for radio and television, loudness inconsistency is also solved for operators that use a radio service as an added audio description service to a television programme for visually-impaired people.

2.4 Analogue television and radio transmission via cable networks

Traditionally, preparation for radio and television audio broadcast has been done by the content provider taking into account the pre-emphasis gain and limitations for audio bandwidth and dynamic range. However, the principal means of distribution has changed to digital transmission. In general, audio codecs incorporated in these systems do not have to cope with analogue-based limitations. Therefore it is recommended that the responsibility for pre-emphasis processing be moved from the content provider to the distribution companies that supply analogue-modulated television and radio services, usually cable operators. This change is intended to encourage operators to recognise that digital distribution in all its forms, including digital television on cable networks, can take advantage of the full capabilities of being digital. This approach applies to the FM radio system specified in ITU-R BS.450-3 [4] and to the television systems using FM, AM and NICAM audio carriers specified in ITU-R BS.707 [5] with the Note that for the AM system L according to ITU-R BT.2043 [6] no pre-emphasis is used. For FM radio, EBU Tech 3344 introduces an unambiguous loudness reference, independent from stereo or mono modulation and irrespective of the amount of bandwidth used for additional signals in the FM multiplex.

For FM modulated radio and television systems it is recommended that pre-emphasis limiting be applied in accordance with, or compatible with, ITU-R BS.642 [7]. This processing could be done with specific equipment or it could be built into the RF modulator itself. Modern modulation equipment based on digital generation of the analogue composite signal offers opportunities to integrate digital pre-emphasis limiting and 15 kHz low pass filtering. Alternatively, the broadcast station could supply a separate audio signal for analogue distribution in addition to audio meant for digital transmission. In DVB systems, this can be done by generating an additional audio stream. Because of reasons described in § 5.2, it must be noted that in this approach the pre-emphasis levels in the modulators can peak significantly higher than the attack level of the pre-emphasis limiter in the studio, which decreases headroom and can cause audible distortion and other artefacts. Incorporating pre-emphasis limiting in the distribution stage is therefore the preferred approach, also taking into account that the number of services processed by cable operators that have not been pre-processed by broadcast stations is increasing.

2.5 Analogue terrestrial television and FM radio transmission

Traditionally, analogue transmitters for terrestrial radio and television transmission have been fed directly from the broadcast studio by using a high quality line. Usually, broadcast stations apply (pre-emphasis) limiting at a central location. As long as the analogue terrestrial transmission remains active, it is recommended that this signal chain be separated from the audio output that supplies digital transmission systems such as DVB, IPTV and DAB(+), or to follow the approach for analogue cable distribution described in the previous section. By doing this, audio quality for digital distribution will not be degraded due to the limitations of the analogue transmission system. Alignment and modulation levels are as for those of cable networks and can be found in § 5.

The loudness level paradigm for FM radio on cable networks described in this document can also be used for terrestrial FM radio transmission, although additional research is recommended for determining the correct modulation levels of a broadcast specification as an alternative for those set out in terrestrial planning standard ITU-R BS.412 [8]. Future legislation shall not only be based on maximum total FM deviation or maximum bandwidth (which includes additional signals such as pilot tone and RDS), but also on the long-term loudness level. The concept of EBU Tech 3344 complies with this requirement.

2.6 Loudness consistency in set-top boxes, television sets and home theatre devices

Whenever signals, codecs and interfaces come together in an audio device, there is a potential risk that differences appear regarding loudness levels. The set-top box, the television set and the AV-receiver are examples of such equipment. Figure 2.6.1 shows a graphical representation of a distribution network feeding consumer equipment. As shown, there are numerous ways to make a connection between equipment and several options to apply internal processing, each introducing risks of experiencing level uncertainties. After transmission over a distribution network, the television signal is received by an Integrated Digital Television (IDTV) and/or an Integrated Receiver Decoder (IRD) which can have several built-in decoders and interfaces. The dotted lines in the IDTV block indicate that some of these devices are able to apply internal decoding of codec bitstreams supplied to the HDMI output whilst others are not. The radio signal is received by the home theatre device or by a separate tuner.

Even if the channels leave the studio with the correct loudness levels, it is difficult to maintain these levels through the chain. Due to a multiplicity of working methods, playback by home theatre equipment such as an AV-receiver can be spoiled by loudness jumps of 11 dB or more when switching from one service to another using the set-top box connected to a home theatre device. In practice these variations in the broadcast chain can increase or decrease existing loudness differences between channels. Because the outcome can also differ between set-top box brands and models, it is impossible for a broadcast station to transmit with a guaranteed result in such a situation. To make matters even worse, as AV-receivers are also manufactured in different ways, mismatches of 4 dB or more can occur between brands and between models of the same brand. To counteract these problems, this document contains extensive guidelines for consumer equipment that include maximum backward compatibility with the installed base of devices.

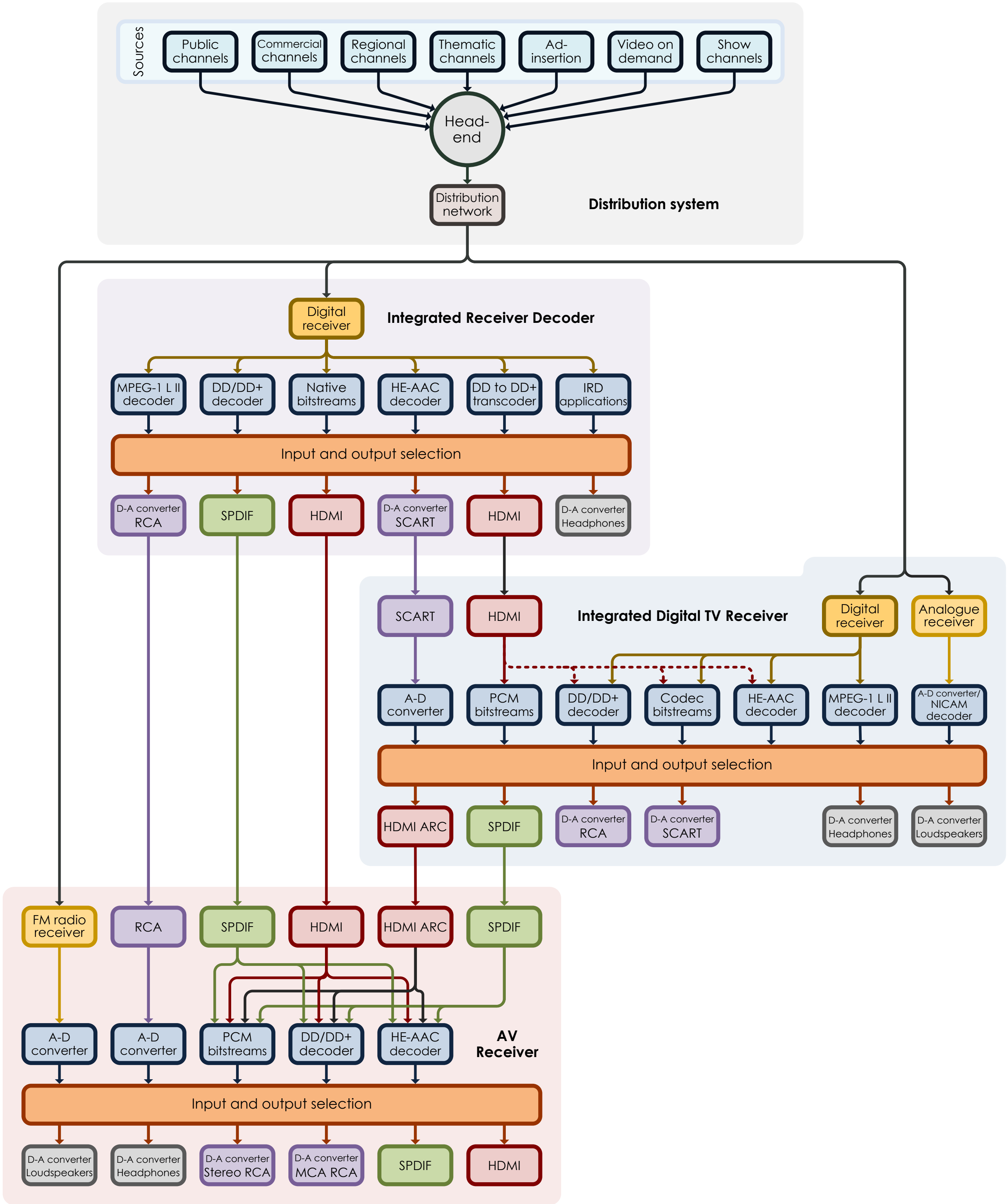


Figure 2.6.1: Various ways of interconnecting consumer equipment, each introducing risks of experiencing level uncertainties.

It is strongly recommended that the industry standardises the level-alignment and loudness level structure for consumer playback equipment. Basically, it is recommended that systems primarily designed for stereo reproduction be standardised to the EBU R 128 Target Level of -23 LUFS. For multi-channel systems it is recommended that equipment be standardised to the equivalent of the Sound Reproduction Level of -31 LUFS, the internal loudness level used in DD/DD+ codecs.

Adaptation of the set-top box is the best way to solve the loudness inconsistencies in the short term, as it is the central device for television broadcast and because it is often accessible remotely by means of a software update. Nevertheless, improvements for other consumer equipment are also included in this document so that in the future problems can be solved on both sides.

2.7 Loudness normalisation in portable and battery powered devices

Radio and television services are available on many distribution platforms, including the Internet and networks for mobile telephony. To prevent each distribution platform requiring its own Target Level based on limitations such as the maximum audio output level of portable and battery-powered devices, it is strongly recommended that all audio-reproduction devices, including battery-powered ones, have a volume control stage with a range that includes sufficient positive gain to allow a Target Level of -23 LUFS. Distortion due to overloads can be prevented by design, for instance by controlling the maximum gain or by including a digital limiter after the positive gain circuit but before the digital to analogue converter and headphone/loudspeaker output. Digital and analogue line outputs from these devices shall be compliant with the specification of media players, which can be found in § 9. If portable devices comply with this requirement, loudness normalisation as described in this document can be used and programmes, streams or files can be exchanged on a broad range of distribution platforms, while direct connections can be made with other equipment without undergoing jumps in loudness in comparison with other sources.

2.8 Note about Dolby Technical Bulletin 11

The document *"Requirement Updates for Dolby Decoders in DVB Consumer Broadcast Receivers, Technical Bulletin 11, 2010 Update"* [#1] contains information based on an old draft version of this Tech Doc. This information, specifically parts which refer according to the handling of the E-EDID query, is out-dated and must be ignored. It is hoped that EBU Tech 3344 can be taken into account for future revisions of that guideline document.

3. Loudness normalisation in digital distribution systems

3.1 Loudness level differences in distribution

Experience of international television broadcasting has shown that loudness levels have been very different in several territories for quite a long time. While some territories may have had their own problems with loudness within their own boundaries anyway, the rise of satellite (cross-territory) broadcasting and the attendant increase in stations playing pop videos put those stations in the same place as commercial radio stations — and the way to compete was to be louder than the competition. This state of affairs degrades the quality of experience and gives rise to customer complaints. Loudness normalisation in ***distribution*** removes the ***long-term differences*** so that the viewer can comfortably switch over from one service to another, so long as the services themselves have a consistent loudness over time. Simultaneously it eliminates the motive for competition on the basis of loudness.

Figure 3.1.1 indicates the effect of loudness normalisation in the distribution stage. Four services are shown with different characteristics. The left-hand scale in each image shows the studio signal level in the digital domain. The right-hand scale shows the loudness level. The tops of the red bars represent the maximum peak hold digital True Peak levels. True Peak level is the maximum level of an audio signal measured with an oversampling True Peak meter. The tops of the yellow bars correspond with the maximum peak hold level measured with a Quasi Peak Programme Meter (QPPM) according to IEC 60268-10 [9]. The tops of the blue bars show the integrated loudness levels measured according to EBU R 128. All levels are measured over one full day (24 hours). The range of average loudness levels is shown by 'Δ'. After the normalisation process, the long-term loudness levels are equal. Maximum True Peak levels and QPPM levels of several services may end up quite different, which is a known and harmless aspect of loudness normalisation, so long as the average target is such that digital clipping under normal conditions does not occur. The Target Level described in EBU R 128 complies with that requirement.

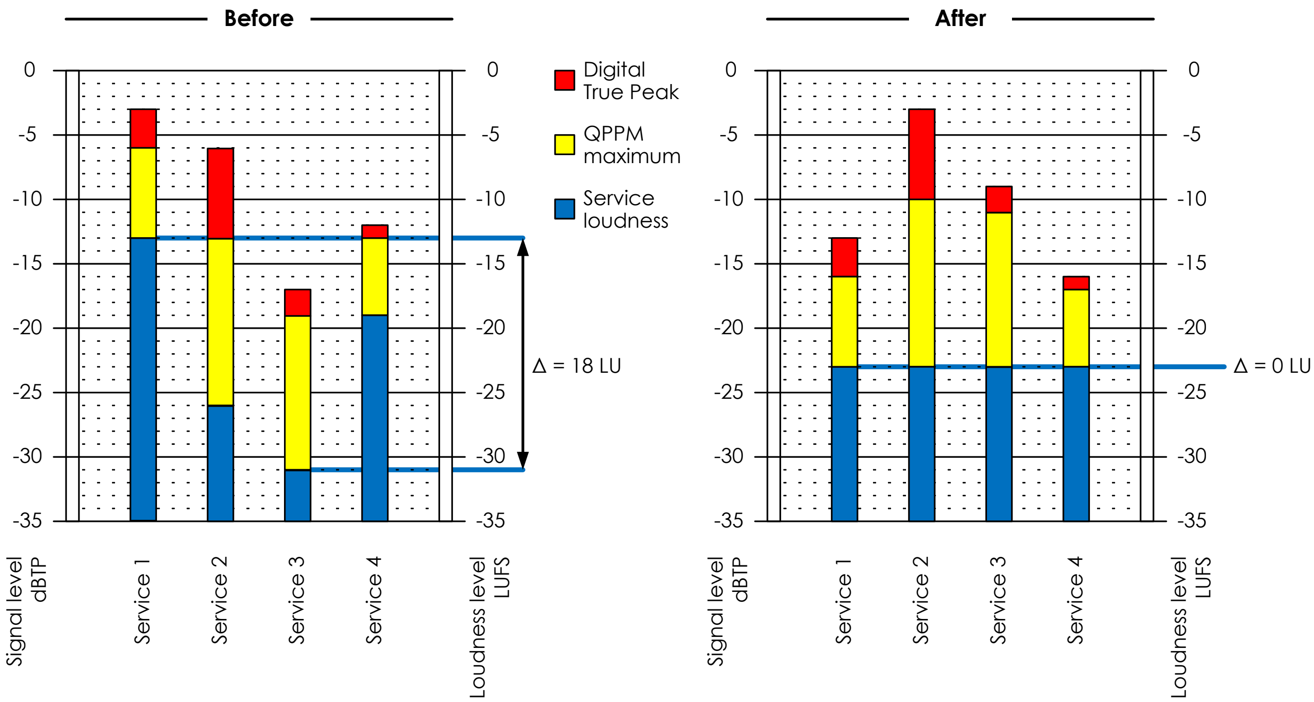


Figure 3.1.1: The effect of loudness normalisation in the distribution stage.

3.2 Active loudness normalisation of digitally-distributed radio and television services

Figure 3.2.1 shows a block diagram of a digital head-end with integrated loudness normalisation; its application to specific platforms, for example IPTV or satellite distribution, can be derived from this figure. The design includes analogue modulation if applicable. More specific details about analogue transmission can be found in § 4.

The method of applying loudness normalisation in radio and television distribution systems requires three different components:

1. A measurement unit.
2. A steering unit.
3. An adaptation unit.

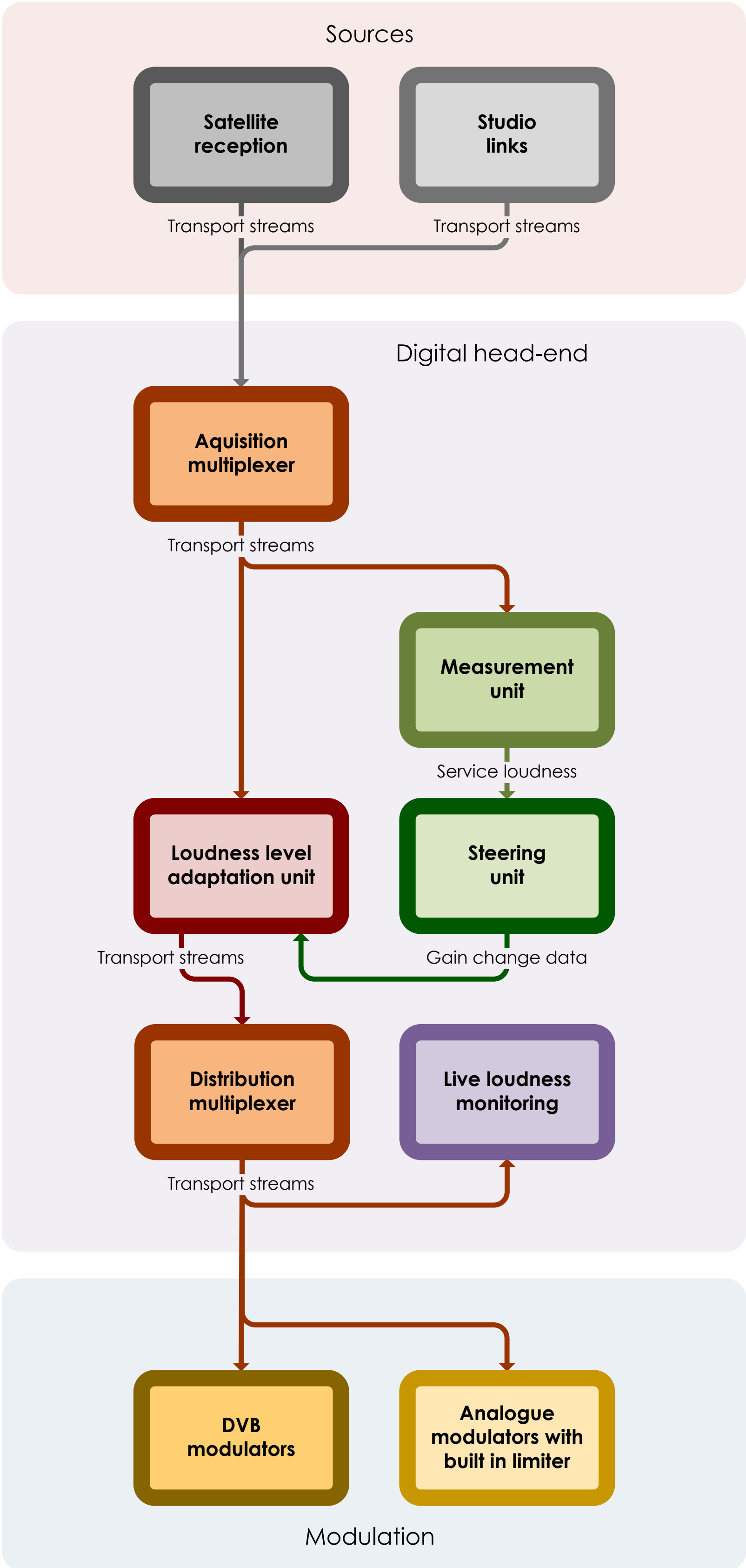


Figure 3.2.1: Block diagram of a digital head-end with integrated loudness normalisation

The adaptation unit could be integrated with a DVB multiplexer or a similar digital signal processing device. The measurement and steering units may also be integrated into a single appliance. Adaptation of loudness levels can also be effected by controlling the input gain of encoders, if applicable. Re-encoding into the same compression format is, however, not considered as a preferred solution due to quality loss and cost inefficiency.

The loudness normalisation system can support one or more codecs. The adaptation process of the loudness level depends on the type of codec used:

- Directly in the audio bitstream for MPEG-1 Layer II.
- Directly in the accompanying metadata for DD/DD+ and HE-AAC.

The following specifies the method of assessing the need for adaptation:

The loudness of decoded signals is continuously measured over a full day (24 hours), split up into 24 blocks of 1 hour each. The start time of block number one is 03:00; the start time of block number 24 is 02:00 the following day. The reason for applying this time of the night is to have minimal influence on daily programming. The integrated (I) measurement is applied on the individual blocks, as specified in EBU Tech 3341 [10]. This means that the same parameters for the measurement as described in EBU R 128 shall be used, including gating below the derived long-term ungated loudness level and an additional silence gate on a fixed level of -70 LUFS. The latter ensures correct operation of the gate when short items are preceded or followed by long periods containing no modulation at all. For optimal stability, only the incoming services are measured (feed-forward principle).

Where DD/DD+ and/or HE-AAC are used, metadata indicating the loudness level is included in the measurement at all times in order to retrieve the reproduction loudness level. The measurement system shall apply a loudness reference level of -31 LUFS for DD/DD+ codecs and -23 LUFS for MPEG-1 Layer II and HE-AAC codecs. For DD/DD+ systems this shall be achieved by applying the '*Dialnorm*' descriptor of the DD/DD+ bitstream and by applying the regular Sound Reproduction Level of -31 LUFS. For HE-AAC this shall be achieved by applying the Programme Reference Level descriptor (*prog_ref_level*, specified in ISO/IEC 14496-3) of the HE-AAC bitstream and by applying the Decoder Target Level descriptor (*target_level*, specified in ISO/IEC 14496-3) at a level of -23 LUFS. If the HE-AAC stream does not contain loudness metadata, insertion of new data or re-encoding of the service is required if the Services Loudness does not comply with -23 LUFS ±1 LU.

The 24 blocks of the day are examined and the block values that are within 2 LU of the highest value are integrated in the power domain. This corresponds with a range of ±1 LU, which is in accordance with EBU R 128. The outcome represents the averaged maximum loudness of the broadcast station operating in its prime time window and is further identified as Service Loudness. This value can deviate slightly from individual programmes that are measured exactly at Target Level. Therefore, the loudness level and permitted deviation for programmes as described in EBU R 128 must be measured at a point before the normalisation system. The measurement unit can optionally perform that task. The principle behind EBU R 128, loudness equalisation between programmes of an individual service, is not influenced by the normalisation described in this Tech Doc, as the correction is applied on all programmes based on the integrated outcome of several full day measurements.

The list of services is stored in a database which reflects all radio and television stations that are being measured and normalised, including those featuring audio services for multiple languages and Audio Description. The database also contains the loudness measurement results. The steering unit compares the Service Loudness (including metadata correction factor) with the EBU R 128 Target Level for MPEG-1 Layer II and HE-AAC services and with the Sound Reproduction Level for DD/DD+ services.

After new data of all services has been acquired at 03:00, the steering unit compares the data with the Target Level and applies an offset if the measured value deviates more than ±1 LU from target, as soon as all information has been processed. In that way the long-term maximum output loudness of all services remains on the EBU R 128 Target Level, while potential undesired side effects of the normalisation are avoided. Tables on the following pages show examples of the applied adjustment. The indicated gain ranges of the correction factor in these tables represent a practical example and do not signify a limitation. The loudness offset value will remain fixed for the next 24 hours. The steering unit must be prevented from pushing the loudness metadata of the corrected audio stream outside the legal range. The adaptation device shall store the settings in non-volatile memory and is only updated by the steering unit. In this way, the measurement and steering process is not part of the path which can critically affect the service.

To prevent inconsistencies, the steering unit shall use a resolution of 0.5 LU per 24 hours. Based on the codec-specific system properties, the change will be applied if the system step size has been reached.

The following system step sizes shall be applied:

- 2 LU for MPEG-1 Layer II bitstream adaptation systems.
- 1 LU for MPEG-1 Layer II encoders and DD/DD+ and HE-AAC metadata adaptation systems.

Optionally, maximum True Peak values of decoded signals are measured. Measurement of maximum True Peak values of the stereo down-mix of a multi-channel service can also be an optional feature, which is particularly useful if the loudness level of a service needs to be increased. The measurement data is made available to the steering unit. Measurement of corrected services for monitoring purposes is optional. Special care has to be taken with switched and time-shared services. The steering unit shall feature functionality to handle these situations.

In order to support dynamically-changing loudness indicators, such as Dialnorm and Programme Reference Level (PRL) in codecs that carry audio metadata, the compensation shall always be applied as an offset to the received value. The offset applied to DD/DD+ and HE-AAC systems is negative towards the received value, which means that to decrease the loudness of the service, the offset has to increase. The offset applied to MPEG-1 Layer II systems is positive towards the received value, which means that to decrease the loudness of the service, the offset has to decrease as well.

The following tables show examples of the relationship between measured loudness and the values of the change to be applied for the various codec systems. The adaptation device processes the offset. It must prevent the loudness metadata of the corrected audio stream from falling outside the legal range. The adaptation device shall be capable, on request, of reporting the actual offset values to the steering unit, which can be displayed in a user interface next to the loudness measurement values. The steering unit shall store the last applied loudness settings in non-volatile memory and shall compare these values with the actual applied offset values stored in the adaptation unit to avoid undesired loudness jumps after a system start-up.

Table 1: Gain adaptation for MPEG-1 Layer II bitstream adaptation systems

Input loudness [1] LUFS	Loudness error LU	Level change to apply [2], LU	Corrected loudness LUFS	
-5.0	+18.0	-18	-23.0	
-6.0	+17.0	-18	-24.0	
-7.0	+16.0	-16	-23.0	
-8.0	+15.0	-16	-24.0	
-9.0	+14.0	-14	-23.0	
-10.0	+13.0	-14	-24.0	
-11.0	+12.0	-12	-23.0	
-12.0	+11.0	-12	-24.0	
-13.0	+10.0	-10	-23.0	
-14.0	+9.0	-10	-24.0	
-15.0	+8.0	-8	-23.0	
-16.0	+7.0	-8	-24.0	
-17.0	+6.0	-6	-23.0	
-18.0	+5.0	-6	-24.0	
-19.0	+4.0	-4	-23.0	
-20.0	+3.0	-4	-24.0	
-21.0	+2.0	-2	-23.0	
-21.9	+1.1	-2	-23.9	
-22.0	+1.0	0	-22.0	↑
-23.0	0.0	0	-23.0	Unity gain range
-24.0	-1.0	0	-24.0	↓
-24.1	-1.1	+2	-22.1	
-25.0	-2.0	+2	-23.0	
-26.0	-3.0	+4	-22.0	
-27.0	-4.0	+4	-23.0	
-28.0	-5.0	+6	-22.0	
-29.0	-6.0	+6	-23.0	
-30.0	-7.0	+8	-22.0	
-31.0	-8.0	+8	-23.0	
-32.0	-9.0	+10	-22.0	

Note 1: The correction shall be applied in integers. For the unity gain range of ±1 LU, no offset is required.

Note 2: The indicated range of the level correction (-18 to +10 LU) is a practical example and does not signify a limitation.

Table 2: Gain adaptation for DD/DD+ metadata adaptation systems

Input loudness [1, 2] LUFS	Loudness error LU	Dialnorm change to apply [3]	Corrected loudness LUFS	
-13.0	+18.0	+18	-31.0	
-14.0	+17.0	+17	-31.0	
-15.0	+16.0	+16	-31.0	
-16.0	+15.0	+15	-31.0	
-17.0	+14.0	+14	-31.0	
-18.0	+13.0	+13	-31.0	
-19.0	+12.0	+12	-31.0	
-20.0	+11.0	+11	-31.0	
-21.0	+10.0	+10	-31.0	
-22.0	+9.0	+9	-31.0	
-23.0	+8.0	+8	-31.0	
-24.0	+7.0	+7	-31.0	
-25.0	+6.0	+6	-31.0	
-26.0	+5.0	+5	-31.0	
-27.0	+4.0	+4	-31.0	
-28.0	+3.0	+3	-31.0	
-29.0	+2.0	+2	-31.0	
-29.9	+1.1	+1	-30.9	
-30.0	+1.0	0	-30.0	↑
-31.0	0.0	0	-31.0	Unity gain range
-32.0	-1.0	0	-32.0	↓
-32.1	-1.1	-1	-31.1	
-33.0	-2.0	-2	-31.0	
-34.0	-3.0	-3	-31.0	
-35.0	-4.0	-4	-31.0	
-36.0	-5.0	-5	-31.0	
-37.0	-6.0	-6	-31.0	
-38.0	-7.0	-7	-31.0	
-39.0	-8.0	-8	-31.0	
-40.0	-9.0	-9	-31.0	

Note 1: The measured loudness is the value including metadata correction factor.

Note 2: The correction shall be applied in integers. For the unity gain range of ±1 LU, no offset is required.

Note 3: The indicated range of the Dialnorm correction (+18 to -9) is a practical example and does not signify a limitation.

Table 3: Gain adaptation for HE-AAC metadata adaptation system

Input loudness [1, 2] LUFS	Loudness error LU	PRL change to apply [3]	Corrected loudness LUFS	
-5.0	+18.0	+18	-23.0	
-6.0	+17.0	+17	-23.0	
-7.0	+16.0	+16	-23.0	
-8.0	+15.0	+15	-23.0	
-9.0	+14.0	+14	-23.0	
-10.0	+13.0	+13	-23.0	
-11.0	+12.0	+12	-23.0	
-12.0	+11.0	+11	-23.0	
-13.0	+10.0	+10	-23.0	
-14.0	+9.0	+9	-23.0	
-15.0	+8.0	+8	-23.0	
-16.0	+7.0	+7	-23.0	
-17.0	+6.0	+6	-23.0	
-18.0	+5.0	+5	-23.0	
-19.0	+4.0	+4	-23.0	
-20.0	+3.0	+3	-23.0	
-21.0	+2.0	+2	-23.0	
-21.9	+1.1	+1	-22.9	
-22.0	+1.0	0	-22.0	↑
-23.0	0.0	0	-23.0	Unity gain range
-24.0	-1.0	0	-24.0	↓
-24.1	-1.1	-1	-23.1	
-25.0	-2.0	-2	-23.0	
-26.0	-3.0	-3	-23.0	
-27.0	-4.0	-4	-23.0	
-28.0	-5.0	-5	-23.0	
-29.0	-6.0	-6	-23.0	
-30.0	-7.0	-7	-23.0	
-31.0	-8.0	-8	-23.0	
-32.0	-9.0	-9	-23.0	

Note 1: The measured loudness is the value including metadata correction factor.

Note 2: The correction shall be applied in integers. For the unity gain range of ±1 LU, no offset is required.

Note 3: The indicated range of the Programme Reference Level correction (+18 to -9) is a practical example and does not signify a limitation.

Figure 3.2.2 shows the input loudness measurement and normalisation principle for MPEG-1 Layer II services; Figure 3.2.3 does the same for DD/DD+ and HE-AAC services.

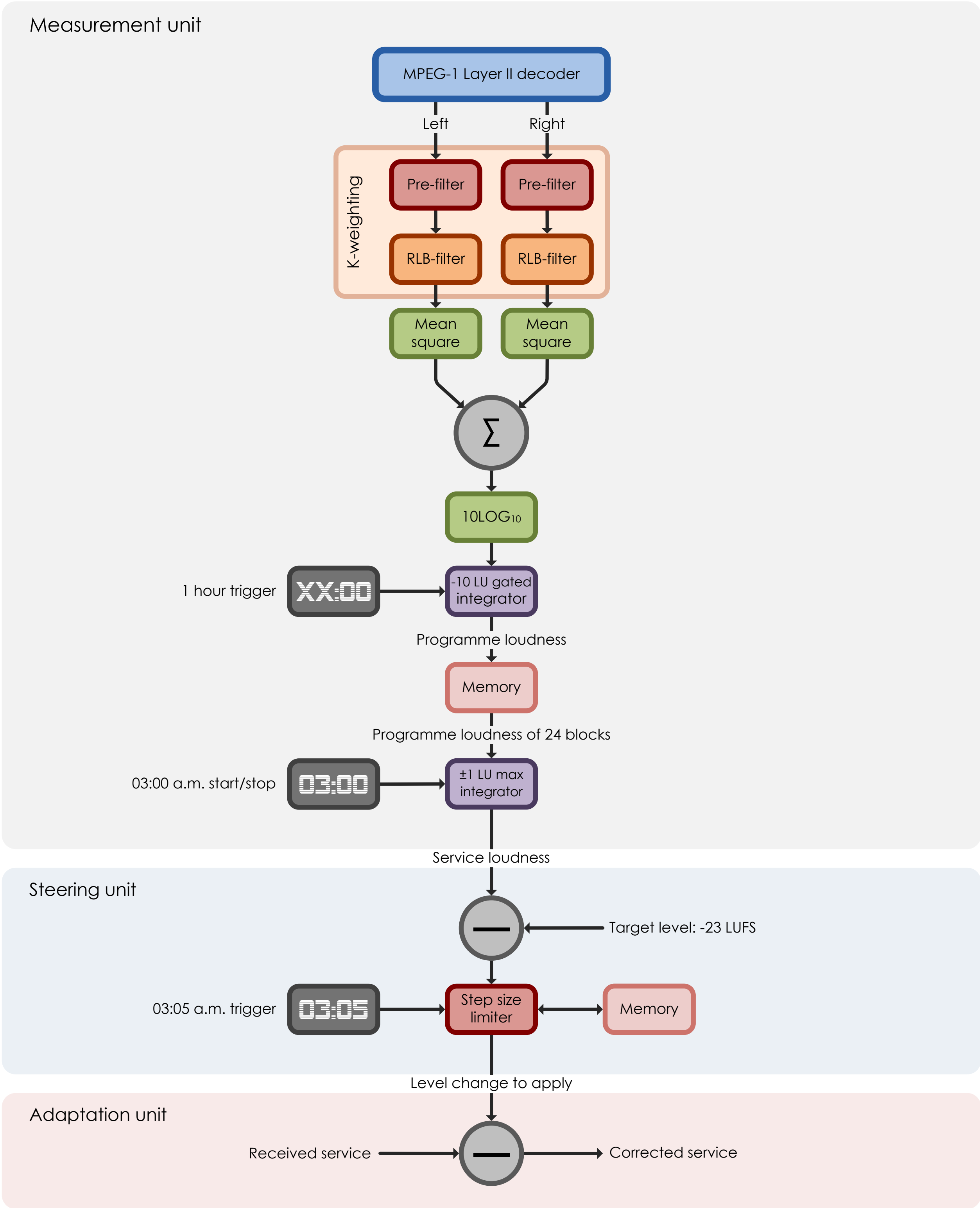


Figure 3.2.2: Block diagram of the continuous measurement process for MPEG-1 Layer II services. The trigger is applied at 03:05 in this case because this system has processed all the blocks by this time.

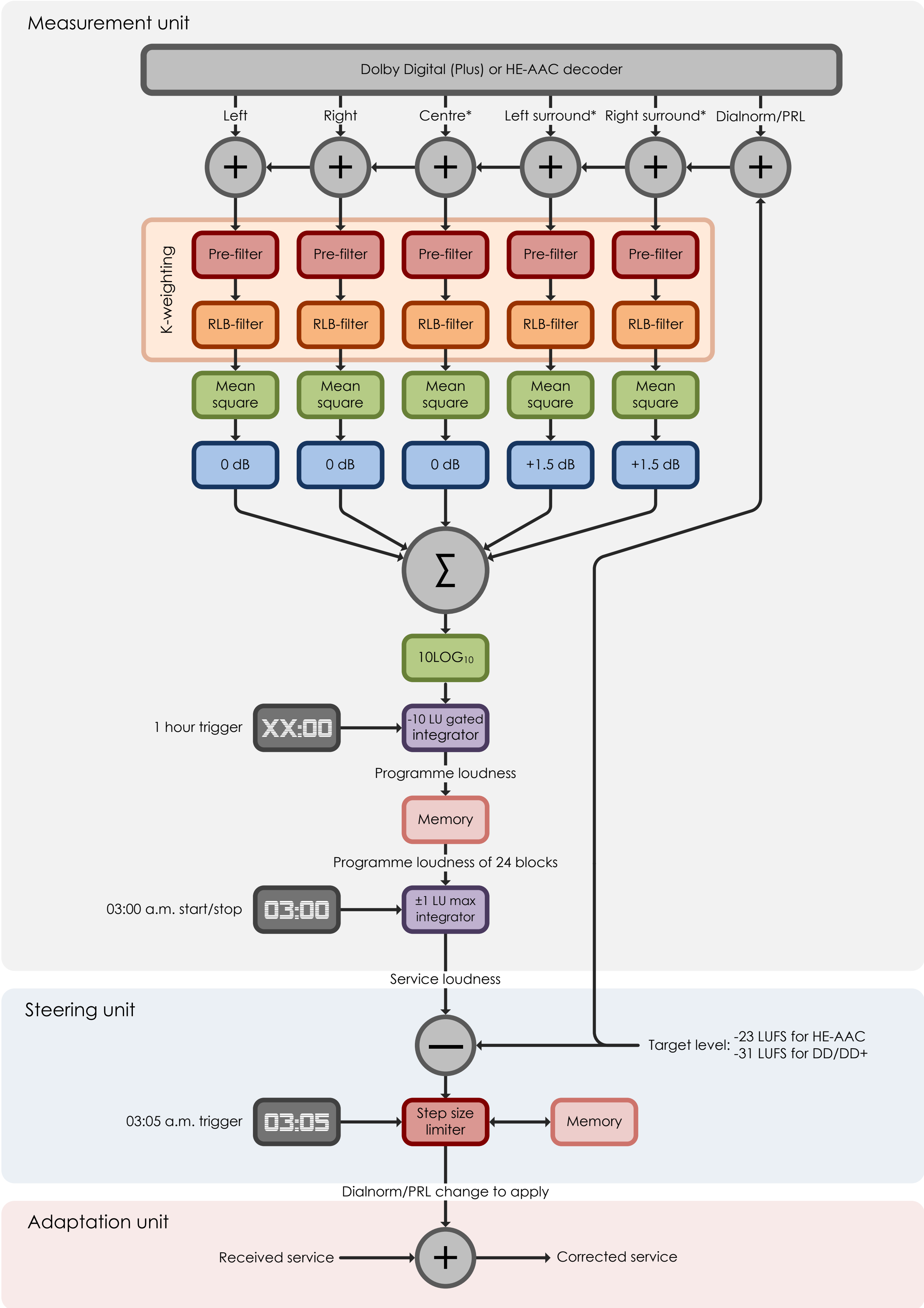


Figure 3.2.3: Block diagram of the continuous measurement process for DD/DD+ and/or HE-AAC services. The trigger is applied at 03:05 in this case because this system has processed all the blocks by this time. * = only present for multi-channel services

3.3 Newly-added services

For newly-added services or in case of a disturbance, the step size and time schedule may be manually and temporarily overruled to allow a quick configuration, for example by directly applying the loudness measurement over the last day or over the last few hours. For newly-added services this should preferably be done before such a service is audible to the listener. The steering unit shall feature functionality to handle this situation.

3.4 Logging and alarm traps

It is recommended that a loudness logging and reporting feature be implemented in the steering unit in order to monitor the automatic normalisation process. For monitoring purposes, it is suggested also that the offset value applied and the actual received and corrected loudness metadata values be reported. It is recommended that at least on the following conditions alarm messages be generated:

- If the True Peak level of a decoded signal reaches or exceeds -1 dBTP (see *Note 1*) after normalisation (see *Note 2*).
- If the True Peak level of the stereo down-mix of a decoded signal reaches or exceeds -1 dBTP (see *Notes 1 & 3*) after normalisation.
- If the difference between two sequential full day Service Loudness measurements is greater than a user-selectable threshold, for example 3 LU.
- If the offset value applied to MPEG-1 Layer II systems is below or above user-selectable thresholds, for example +6 and -14 LU (see *Note 4*).
- If the offset value attempts to push the loudness metadata of the corrected audio streams of DD/DD+ and HE-AAC systems below -31 or above a user-selectable threshold, for example -10 (see *Note 4*).
- If loudness metadata in DD/DD+ or HE-AAC services is invalid or missing.

Note 1: Only if the True Peak measurement option is present. Alternatively, sample peak measurement may be used with an alarm triggering level at -3 dBFS if True Peak measurement requires too much computational load. Sample peak measurement at that level may however not register occasional True Peak level overloads.

Note 2: Values after normalisation can be calculated by adding/subtracting (depending on the codec system) the offset value to the measured input loudness.

Note 3: Only if the stereo down-mix True Peak level measurement option is present. Measurement of down-mix levels is particularly useful if the loudness level of a service needs to be increased. There are two stereo down-mix levels to be checked: for -31 LUFS and -23 LUFS reproduction levels. Metadata incorporating overload protection must be applied, if available. For DD/DD+ RF mode decoding, the DRC metadata shall also be applied.

Note 4: It is recommended that measurements outside the average range be monitored and/or investigated. Some (radio) stations can be found that transmit audio with extremely high loudness levels.

Loudness normalisation shall be a continuous, automated process. It is however recommended that the alarm reports of the measurement system be monitored on a daily basis and that loudness metering in the master control room or network operations centre be implemented to permit random checks.

3.5 Local digital head-ends

Local digital head-ends are preferably fed by means of a contribution link from a (redundant) central head-end where loudness normalisation is applied. Figure 3.5.1 shows an example; its application to specific platforms, for example IPTV distribution, can be derived from this figure.

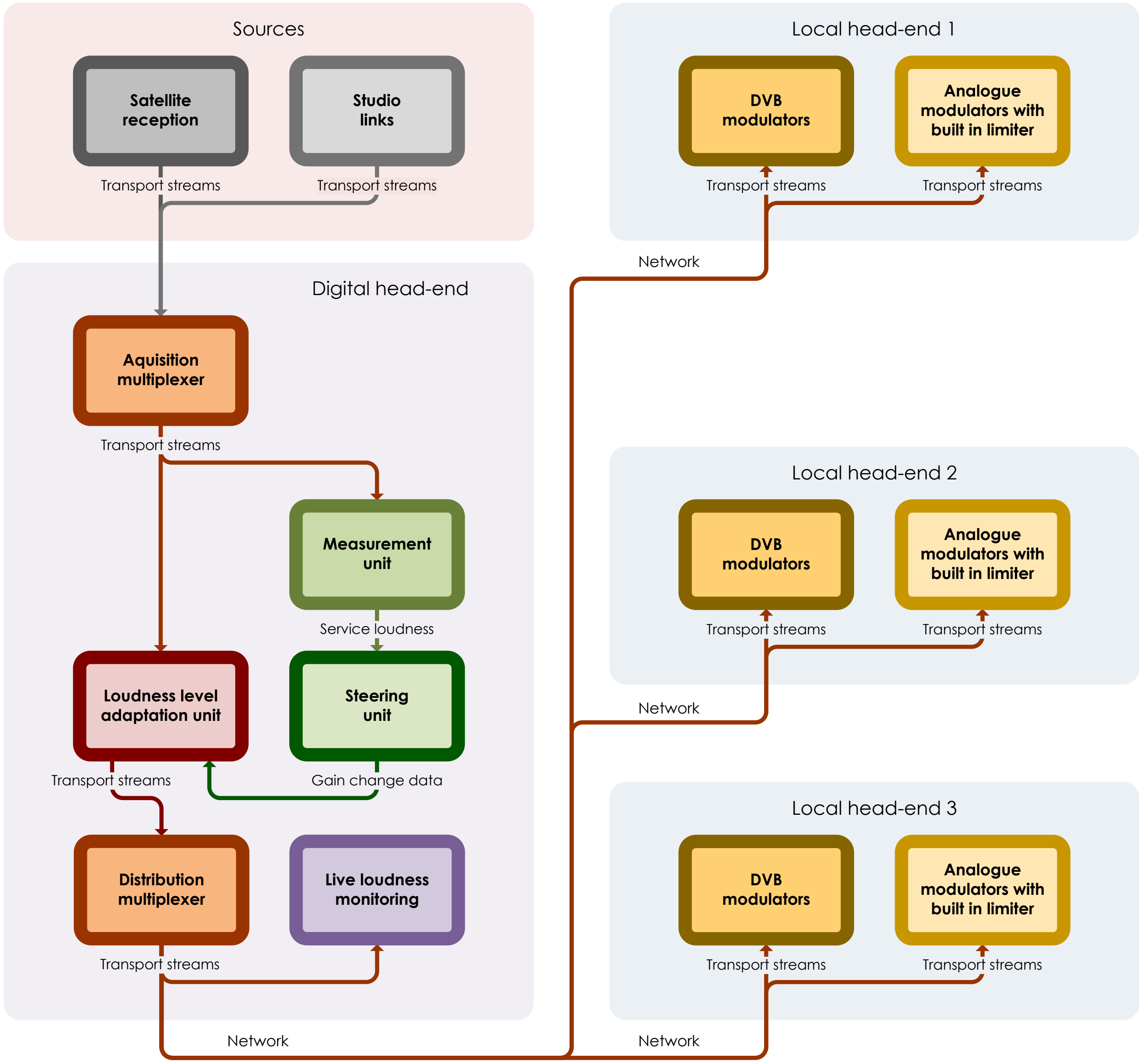


Figure 3.5.1: Contribution to several local head-ends from a central head-end with loudness normalisation.

If this centralised network supply is not possible and the local head-end has to use its own acquisition feeds, loudness normalisation shall be applied locally, based on the same principles as described in § 3.2. Optionally, centrally-gathered loudness data can be used to control local systems remotely via a data connection. However, it should be ensured that there is no loudness difference between the locally received sources and, for example, a studio link source in the central head-end that is used to feed the measurement system. If the local source is, for example, terrestrial reception and the loudness level appears to be different, the same signal can be fed into the measurement system at the central head-end, assuming that loudness levels do not differ over the same terrestrial network, even if this signal is not used in the central head-end for its primary distribution. Optionally, the audio part of services that are available on a local basis only could be fed back to the measurement system of the central head-end. To save bandwidth on the connection in such a situation, the video part of a television service could be split off. Figure 3.5.2 shows an example of shared use of loudness data; its application to specific platforms, for example IPTV distribution, can be derived from this figure.

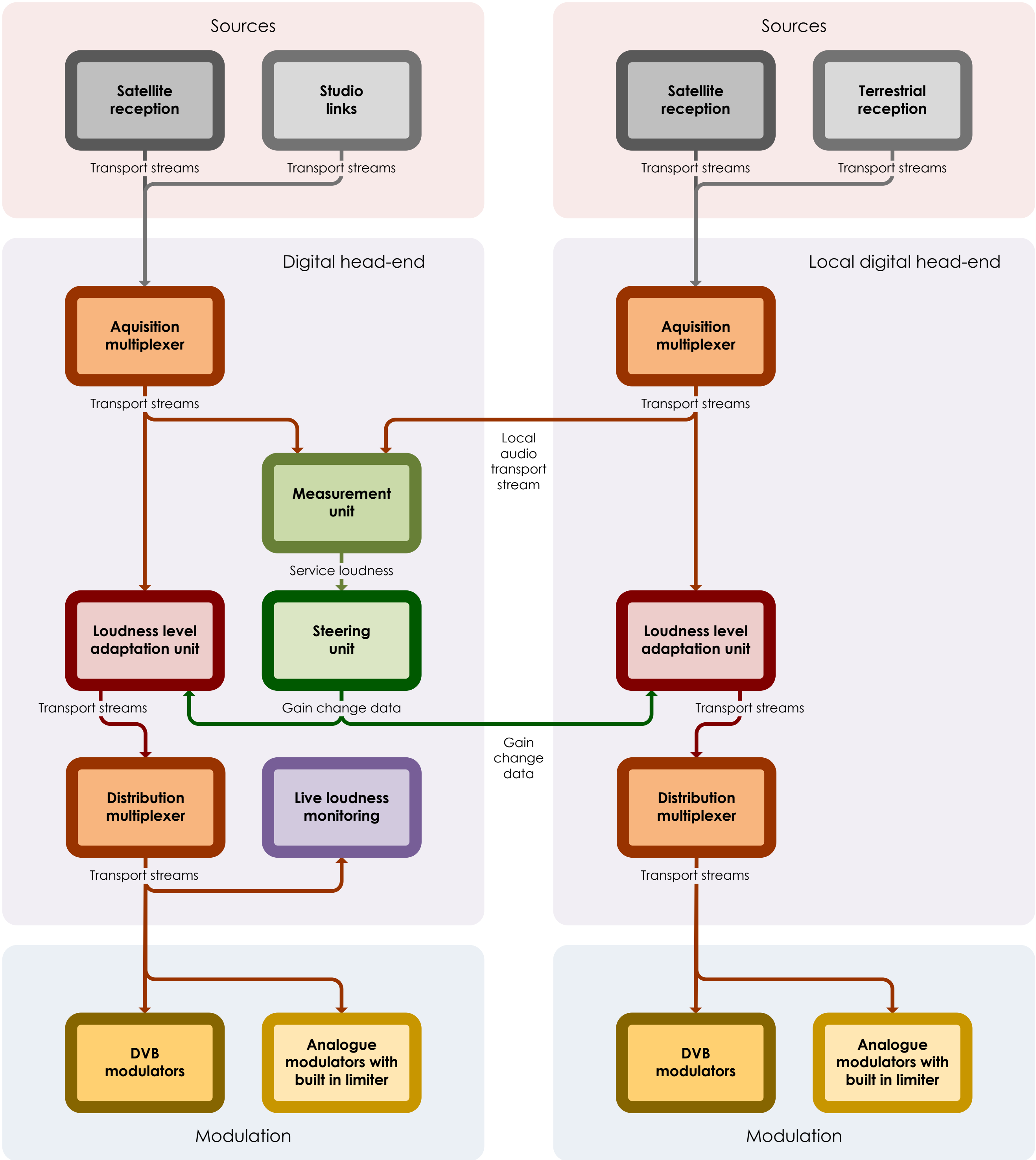


Figure 3.5.2: Supply of loudness data to one or more local head-ends from a central head-end.

3.6 Inconsistent audio sources

The broadcast station appears to have inconsistent audio if the alarm conditions described in § 3.4 show up regularly. Note that, as with Service 3 in Figure 3.1.1, some services may need to be increased in loudness. While being too soft will not cause the problems of listener fatigue and clipping which are associated with excessively-loud signals, being too soft still presents listeners with the problem of loudness-level change when switching to or away from the service. However, in some cases, the sound on the service may repeatedly be background music which supports a sequence of news and weather slides, say. In such cases, there may need to be consultation between the broadcaster and the distributor to ensure that the levels are satisfactory and it may be necessary to enable static normalisation for services with such irregular content. Static normalisation means that the loudness level is adjusted manually and checked repeatedly by use of the logs of the measurement unit. The steering unit shall feature functionality to handle static normalisation.

3.7 Interactive applications

Interactive applications on an IRD, IDTV or media player that make use of accompanying sound can be made consistent with EBU R 128 by normalising the audio in advance by, for example, an algorithm implemented in software. Care shall be taken within the design of the IRD so that the signal alignment corresponds to that of the broadcasted audio via all audio interfaces with the aim of achieving an equal integrated loudness level.

3.8 Advertisement insertion

In systems where locally-inserted advertisements are applied, the switching shall be located after the loudness normalisation system. The commercials shall be normalised in advance to the EBU R 128 Target Level by, for example, an algorithm implemented in software. As a result of that, the average loudness level of the content being broadcasted is made equal to the Target Level at the position of the ad-insertion switch. In DD/DD+ or HE-AAC sound systems, loudness metadata of the locally-inserted advertisements shall always correctly indicate the actual loudness. If a service is passed through a system that plays out the advertisements, no gain or attenuation shall be applied in that device. Algorithms inside a play-out device that are designed to follow the average loudness of the main programme shall be switched off if the content has been pre-processed. Loudness measurement for monitoring purposes after the ad-insertion switch is optional.

3.9 Video on demand and other play-out systems

Distribution companies that make use of their own play-out systems for video on demand (VOD) and similar services are considered to act as a broadcast station. This means that the EBU R 128, this document and the other accompanying loudness guideline documents are applicable:

EBU Tech 3341 Loudness Metering: 'EBU Mode' metering to supplement loudness normalisation in accordance with EBU R 128.

EBU Tech 3342 Loudness Range: A descriptor to supplement loudness normalisation in accordance with EBU R 128.

EBU Tech 3343 Practical Guidelines for Production and Implementation in accordance with EBU R 128.

The content stored on play-out systems can be checked and normalised in advance by an algorithm implemented in software, for example. By doing this, the average loudness level of the content being broadcast is made equal to the EBU R 128 Target Level. In DD/DD+ or HE-AAC audio systems, loudness metadata of VOD programmes shall always correctly indicate the actual loudness. Continuous loudness measurement of VOD content for monitoring purposes is recommended.

Figure 3.9.1 indicates loudness normalisation in video on demand and ad-insertion systems; its application to specific platforms, for example IPTV distribution, can be derived from this figure.

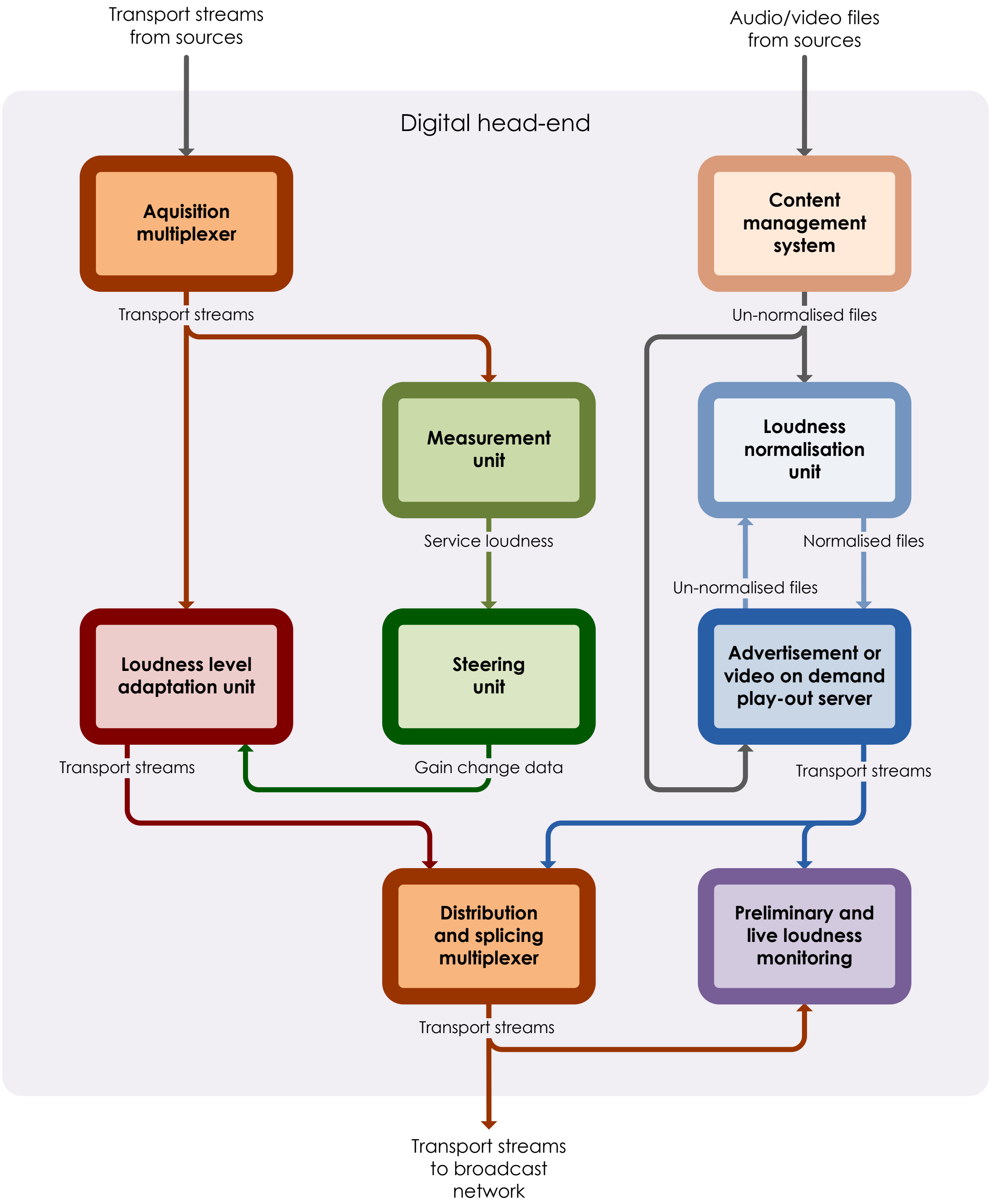


Figure 3.9.1: Block diagram of a digital head-end with integrated loudness normalisation including video on demand and ad-insertion.

Note 1: While a file may be pre-processed by the loudness-normalisation device before it is stored on the play-out server, it is also possible that the loudness-normalisation device may access a file which already exists on the server. This is the reason why two routes to the normalisation device are drawn.

3.10 *Regionally switched services*

A main service that is regularly interrupted by a regional service and is switched as a DVB stream can suffer from loudness differences between the sources. If the regional service is played out from a file-based system, the same approach as for video on demand described in § 3.9 can be applied. If the regional service is live transmission, the studio must take care that the transmitted loudness is according to EBU R 128. The regional service can be corrected if necessary by separately applying long-term measurement on this signal. This can be achieved by only measuring during the time slots of regional transmission. The steering unit shall feature functionality to handle these situations. The following diagram indicates loudness normalisation in the case of a regionally switched service; its application to specific platforms, for example IPTV distribution, can be derived from this figure.

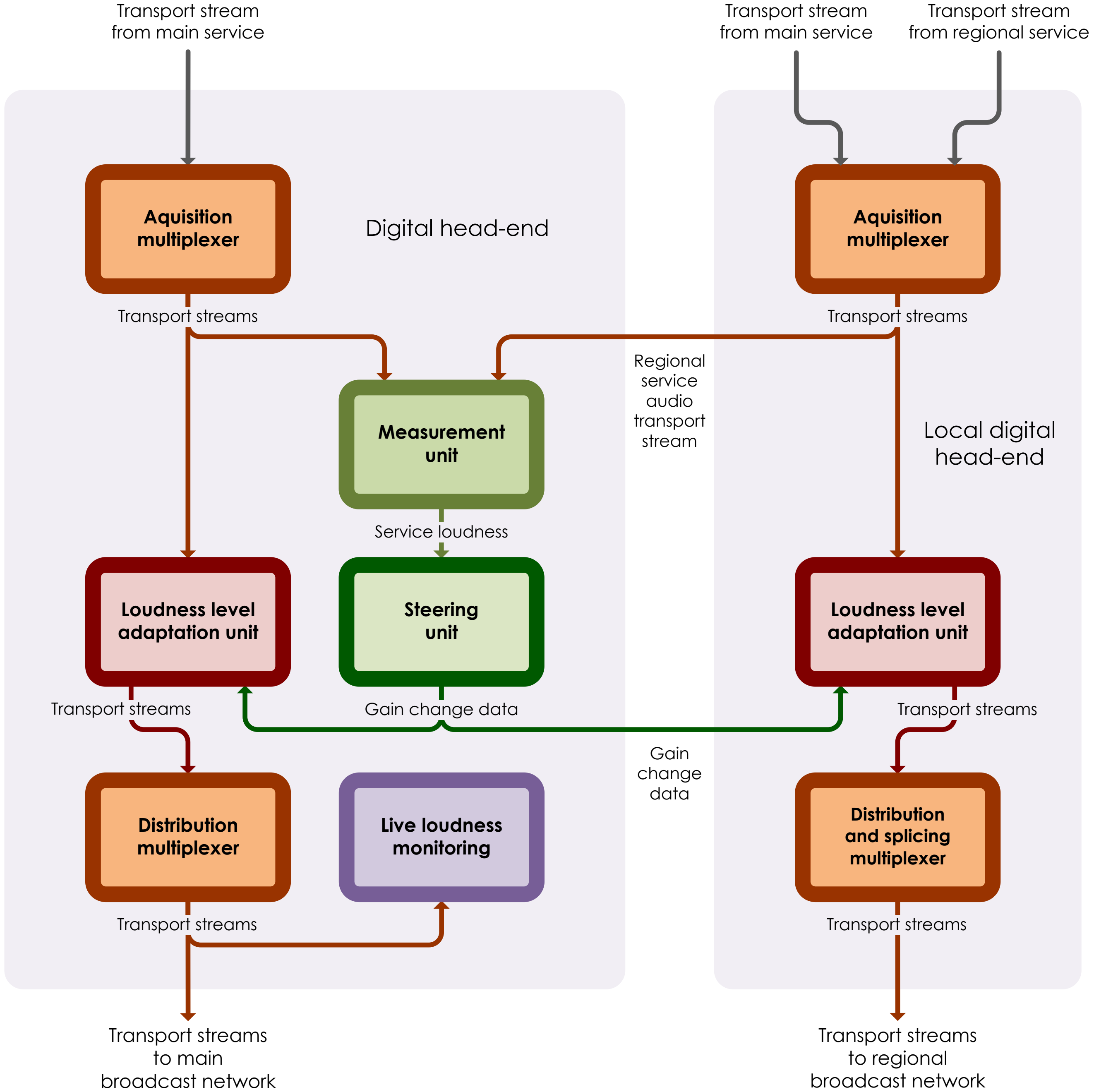


Figure 3.10.1: Block diagram of a digital head-end with integrated loudness normalisation and a local head-end where the main service is interrupted by a regional service.

4. Loudness normalisation in analogue distribution systems

4.1 Loudness level differences in distribution

This section is similar to § 3.1, with the difference that the effect of pre-emphasis is added. Figure 4.1.1 indicates the effect of loudness normalisation in the distribution stage. Four services are shown with different characteristics. The left-hand scale in each image shows the studio signal level in the digital domain. The right-hand scale shows the loudness level. The tops of the red bars represent the maximum peak hold digital True Peak levels. True Peak level is the maximum level of an audio signal measured with an oversampling True Peak meter. The tops of the yellow bars correspond with the maximum peak hold level measured with a Quasi Peak Programme Meter (QPPM) according to IEC 60268-10. The tops of the blue bars show the integrated loudness levels measured according to EBU R 128. The tops of the orange bars represent the signal after applying pre-emphasis gain inside the analogue modulator. All levels are measured over one full day (24 hours). The range of average loudness levels is shown by 'Δ'.

After the normalisation process, the long-term loudness levels are equal. Maximum True Peak levels, QPPM levels and pre-emphasis levels of several services may end up being quite different, which is a known and harmless aspect of loudness normalisation as long as the average target is such that digital clipping does not occur and as long as appropriate limiting is applied at the modulator stage. The Target Level described in EBU R 128 and the concept of EBU Tech 3344 complies with both requirements.

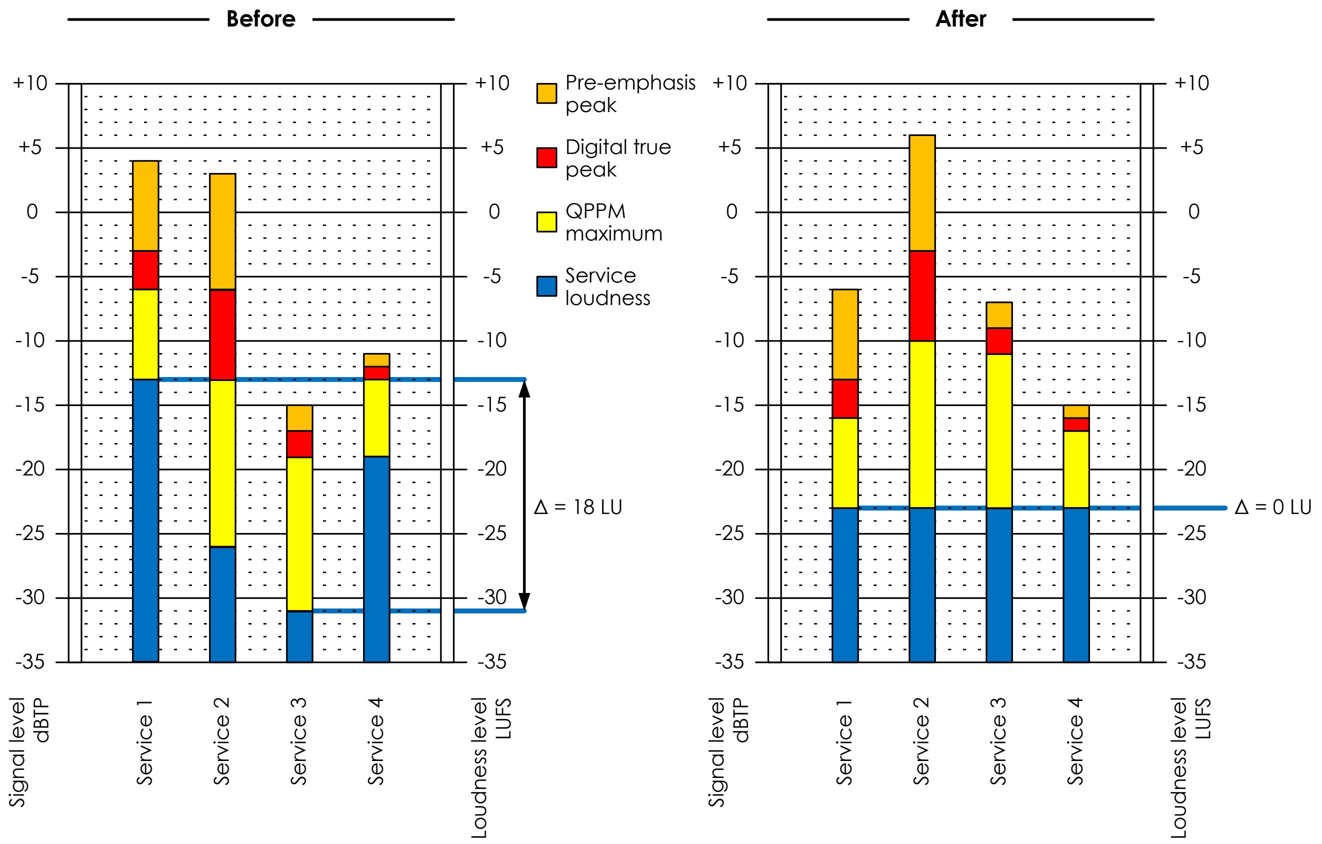


Fig 4.1.1: The effect of loudness normalisation in the distribution stage

4.2 Limiting

After loudness levels have been normalised according to EBU R 128, peak and pre-emphasis levels could end up too high for the analogue transmission system. Therefore True Peak and pre-emphasis limiting shall be applied (no pre-emphasis is used for AM system L, which means that True Peak limiting would be sufficient). This processing could be done with dedicated equipment or could be built into the analogue modulator itself.

For FM modulated radio and television transmission systems, it is recommended that pre-emphasis limiting be applied according to, or compatible with, ITU-R BS.642. Because of reasons described in § 5.2, it could be necessary to decrease the limiter level which reduces headroom. Consequently, the best position for the pre-emphasis limiter is directly connected to or built into the modulator.

Figure 4.2.1 indicates the effect of limiting at the modulator stage. The same services are shown as in the previous section. The recommended end stage limiter cuts off the level including pre-emphasis gain. As the loudness Target Level is set sufficiently low, the limiter will need to attack only occasionally, on (pre-emphasis) peaks. The Target Level described in EBU R 128 complies with that requirement. Therefore, the limiter will have minimal influence on the average loudness. As an example, the figure reflects the levels for television systems B, B1, D, D1, G, H, K, K1, I and I1 and for the FM radio system according to ITU-R BS.450-3 (modulation levels for all European television systems can be found in § 5.2).

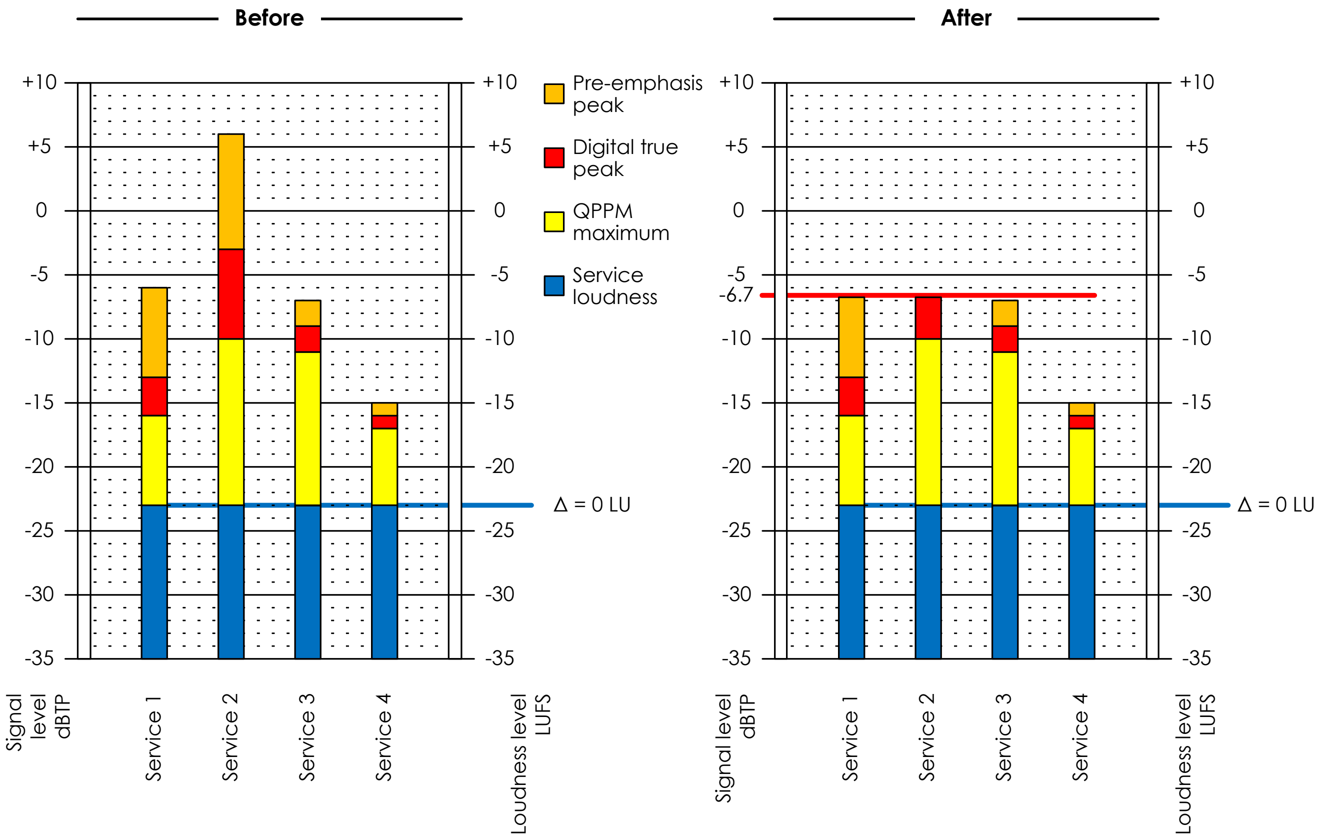


Fig 4.2.1: The effect of pre-emphasis limiting in the modulator stage

4.3 Active loudness normalisation of analogue distributed radio and television services

Analogue head-ends are preferably fed by means of a distribution link from a (redundant) central digital head-end where loudness normalisation is applied (see § 3.5). As a result, the process can be implemented in a very efficient way wherein both transmission systems are served. Analogue modulators can be set to a fixed default adjustment and require no further maintenance to control the audio levels with which they are concerned. This simplifies operational procedures and optimises cost efficiency and consistency.

4.4 Local analogue head-ends

Where the analogue head-end cannot be fed from a digital platform and has to use its own acquisition feeds, loudness normalisation shall be applied locally, based on the same principles as discussed in § 3. Optionally, centrally-gathered loudness data can be used to control local systems remotely via a data connection; more detail is given in § 3.5.

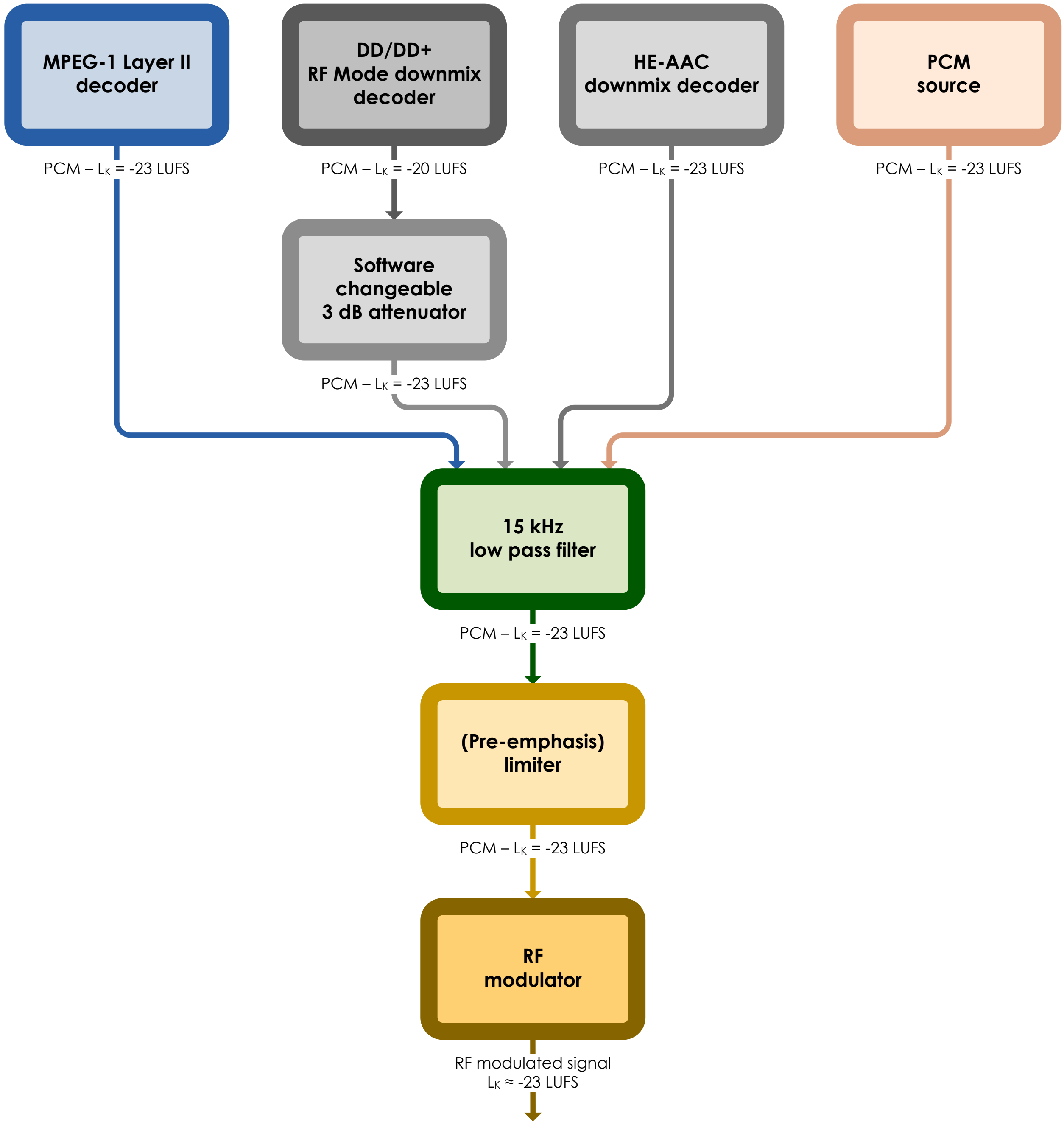


Fig 4.5.1: The design of an analogue head-end for television systems and FM radio

Instead of PCM audio, an analogue input signal can be used, based on the mapping in § 5.

4.5 Head-end design for television systems and FM radio

Figure 4.5.1, above, shows the generic design of the analogue head-end for television systems and FM radio, where loudness normalisation has been performed in the digital head-end or a similar previous stage. The source could, for example, be an MPEG-1 Layer II audio stream, a DD/DD+ down-mix, an HE-AAC down-mix, or a direct line from the studio.

A 15 kHz low-pass filter is included, compliant to transmission standards. It is positioned in front of the pre-emphasis limiter to prevent the limiter from reacting to content in the frequency range above 15 kHz. Please refer to the modulation levels and limiter thresholds in § 5.2.

5. Level alignment in analogue and digital distribution systems

5.1 Level alignment between systems and interfaces

This section gives an overview of the level alignment for European television and radio transmission systems. Based on what is pointed out in EBU R 128, the maximum programme levels in operational guidelines that were based on ITU-R BS.645 [13] will become obsolete and need to be replaced by new values, guaranteeing equal loudness between systems and interfaces and assuring efficient use of the available headroom. To avoid loudness inconsistencies, the level alignment schemes between the transmission systems and output interfaces described in this section shall be applied, as determined for the relevant network. The alignment schemes are compliant to CENELEC EN50049 [14]; the European standard that specifies the SCART interface. In this section, a ‘set-top box’ is further referred to as an ‘Integrated Receiver Decoder’ (IRD).

5.2 Modulation levels for analogue television and radio systems

If loudness normalisation based on a Target Level of -23 LUFS is applied prior to the analogue modulator, the equipment can be aligned to a default setting and it will require no further adjustment of audio levels. Several figures in this section show the relationship between the levels at several stages and systems (interfaces, codecs and analogue as well as digital modulation).

A 1 kHz sine wave is used as a reference, according to CENELEC EN50049. The limiter thresholds include pre-emphasis gain and are based on True Peak values.

The following settings shall be used for television systems according to ITU-R BT.2043 and for FM Radio:

Television systems	B, B1, D, D1, G, H, K, K1, I and I1
Modulation	FM
Level alignment (1, 5)	-6.7 dBTP using a 1 kHz sine wave in phase on left and right channel results in 50 kHz FM deviation.
	-12 dBTP using a 1 kHz sine wave in phase on left and right channel results in 27.0 kHz FM deviation.
Limiter threshold (1, 2)	-6.7 dBTP referenced to 1 kHz.
Pre-emphasis limiting	50 μs
Low-pass filter	15 kHz

Television system	**L**
Modulation	AM
Level alignment (1, 5)	-7 dBTP using a 1 kHz sine wave in phase on left and right channel results in 96% AM modulation depth. -12 dBTP using a 1 kHz sine wave in phase on left and right channel results in 54.0% AM modulation depth.
Limiter threshold (1, 2)	-7 dBTP
Pre-emphasis limiting	None
Low-pass filter	15 kHz
Television systems	**B, B1, D1, G, H, K1 and L**
Modulation	NICAM
Level alignment (1, 5)	-12 dBTP using a 1 kHz sine wave results in -11.2 dBTP digital coding level inside the NICAM modulator.
Limiter threshold (1, 2)	-2 dBTP referenced to 1 kHz.
Pre-emphasis limiting	ITU-T J.17 [15]
Low-pass filter	15 kHz
Television systems	**I and I1**
Modulation	NICAM
Level alignment (1, 5)	-12 dBTP using a 1 kHz sine wave results in -15.8 dBTP digital coding level inside the NICAM modulator.
Limiter threshold (2)	Optional, 0 dBTP referenced to 1 kHz.
Pre-emphasis limiting	Optional, ITU-T J.17
Low-pass filter	15 kHz
Radio system	**ITU-R BS.450-3**
Modulation	FM stereo
Level alignment (1, 3, 5)	-6.6 dBTP using a 1 kHz sine wave in phase on left and right channel results in 65 (75) kHz FM deviation. -12 dBTP using a 1 kHz sine wave in phase on left and right channel results in 35.0 (45.0) kHz FM deviation.
Limiter threshold (1, 2)	-6.6 dBTP referenced to 1 kHz
Pre-emphasis limiting	50 μs
Low-pass filter	15 kHz
Radio system	**ITU-R BS.450-3**
Modulation	FM mono
Level alignment (1, 4, 5)	-5.4 dBTP using a 1 kHz sine wave results in 75 kHz FM deviation. -12 dBTP using a 1 kHz sine wave results in 35.0 kHz FM deviation.
Limiter threshold (1, 2)	-5.4 dBTP referenced to 1 kHz
Pre-emphasis limiting	50 μs
Low-pass filter	15 kHz

Note 1: True Peak Level is the maximum peak level of an audio signal measured with an oversampling True Peak Meter. If a True Peak meter is not available, a sine wave at 997 Hz, encoded at the specified level in dBFS, may be used for reference.

*Note 2: Due the following causes it could be necessary to decrease the pre-emphasis limiter level with **a few tenths to several dB** to avoid overstepping the maximum allowed modulation:*

- *Overshoot can be present between the True Peak measurement value based on the use of a four times oversampling interpolation filter (see ITU-R BS.1770 [16] for details) and the analogue level after digital to analogue conversion. This maximum difference is less than 1 dB (typically a few tenths of a dB).*
- *Most transmission standards do not specify the group delay of the pre-emphasis filter. Therefore, overshoots occur if the limiter is based on digital calculation and the modulator is using an analogue RC-network. Modern modulators that generate the analogue composite signal and pre-emphasis in the digital domain are usually more accurate.*
- *Overshoots occur if there is a codec between the limiter and the modulator which is not lossless (for example MPEG-1 Layer II) and/or if there are circuits between those devices that introduce frequency-response errors or non-constant group delay.*
- *If analogue interfaces are used, some variations in level accuracy could be present.*

Because of these reasons, the best position for the pre-emphasis limiter is directly connected to or built into the modulator itself.

Note 3: Pilot tone, RDS and other additional signals within the FM stereo multiplex signal are assumed to represent in total 10 kHz FM deviation. The listed values represent the FM deviation caused by the 1 kHz tone, including the effect of pre-emphasis gain. The value in brackets represents the total FM deviation including pilot tone, RDS and other additional signals. The quantity of additional signals only influences the maximum headroom available for audio and the corresponding limiter level. In all cases, the loudness reference for the measurement and the alignment is unambiguous and remains the same.

Note 4: This alignment is valid for mono transmission without any additional signals such as RDS. The listed values represent the FM deviation caused by the 1 kHz tone, including the effect of pre-emphasis gain. Any added signal in the spectrum only decreases headroom and the corresponding limiter level. To simplify operational procedures, the same limiter level as that used for FM stereo radio may also be used for mono signals. In all cases, the loudness reference for the alignment is unambiguous and remains the same.

Note 5: -12 dBTP corresponds to an analogue relative level of +6 dBu0s, as specified in ITU-R BS.645. In countries which apply a normalizing factor of 0 dBrs, -12 dBTP corresponds to an absolute analogue level of +6 dBu. In countries which apply a normalizing factor of -3 dBrs, -12 dBTP corresponds to an absolute analogue level of +3 dBu.

5.3 Note about IEC EN60728-5

The alignment levels described in this document are not in accordance with the modulator input signal level specifications described in section 6.5.3 of IEC EN60728-5 [17]. As no information is included in that standard about loudness of radio and television broadcasts, it is hoped that EBU R 128 and EBU Tech 3344 can be taken into account for future revisions. For the time being it is strongly recommended to ignore the audio input level specification in that standard.

5.4 Clarification for the level alignment figures

The figures in § 5.5 - § 5.10 show a graphical representation of the level alignment between systems and interfaces. This signal level relationship is only valid for all systems using both left and right channel simultaneously.

The red lines represent the maximum peak level of the transmission. The purple lines indicate the limiter level for transmission systems and the corresponding alignment to the input and output interfaces.

The left side of the red bars represent the maximum peak level allowed on the interface or codec system. In case of a 1 kHz sine wave, the red lines point to levels that can be measured.

The level of programme material may peak higher than the red line at the output interfaces of systems due to overshoots in the codec system. These overshoots should however not come into the range of the red bars, as this could result in clipping. Based on measurements of live transmission, it might therefore be necessary to decrease the limiter level in front of the codec system if relatively low bitrates are used which in general cause a rise of overshoots. Grey lines emphasize the alignment relationship at levels that have a particular meaning:

- -12 dBTP corresponds with the reference level as specified in CENELEC EN50049. EBU Tech 3344 is fully compatible with that standard.
- -18 dBTP is the alignment signal as specified in ITU-R BS.645. The signal level that should be present at the input and output of several codec systems, RF modulation systems and interfaces can be read out from the graph.
- -23 dBTP is the signal level using a 1 kHz sine wave on both left and right channel that is equivalent with a loudness level of -23 LUFS. This level is particularly useful if an EBU R 128 compliant loudness meter is used to check the alignment.

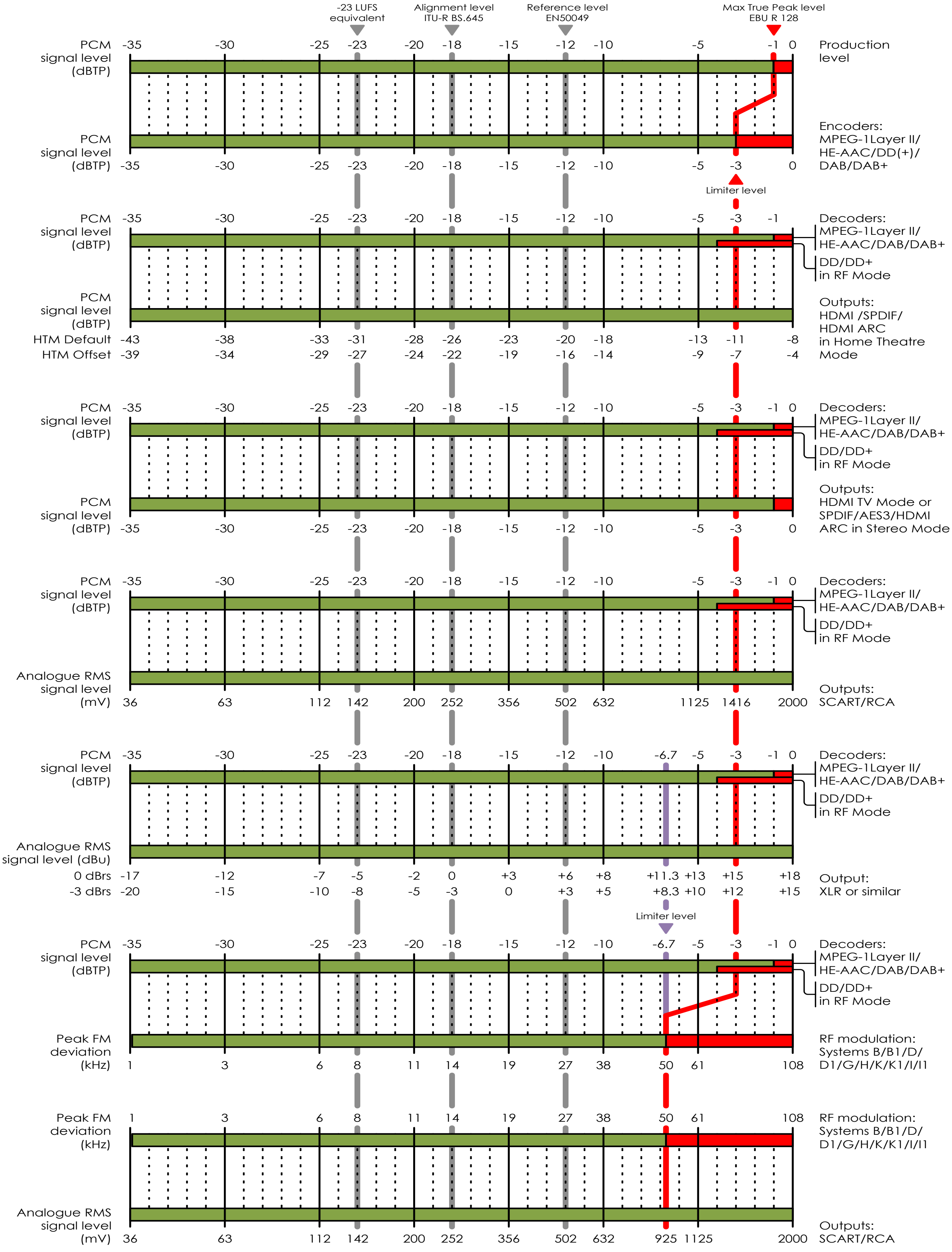
The following table describes the signal level bars from top to bottom:

Subject	**Description**
Production level	This is the digital audio signal level measured in dBTP of audio source material ingested and played out in the broadcast studio. According to EBU R 128, the limit for the production level is specified at -1 dBTP. This level covers the maximum under-read of a True Peak meter that is using a four times oversampling interpolation filter (see ITU-R BS.1770 for details).
Encoders: MPEG-1 Layer II/ HE-AAC/DD(+)/ DAB/DAB+	This is the digital signal level measured in dBTP of the encoded audio using one of the codecs and systems listed. The red line represents the maximum peak level. In practice, this means that the signal shall be passed through an end-stage peak limiter set to a level of -3 dBTP before applying it to the encoder. It might be necessary to decrease the limiter level in front of the codec system if relatively low bitrates are used.
Decoders: MPEG-1 Layer II/ HE-AAC/DAB/ DAB+ and DD/DD+ in RF Mode	This is the digital signal level measured in dBTP of the decoded audio, either using an HE-AAC, MPEG-1 Layer II or an EBU Tech 3344 compliant DD/DD+ decoder in RF Mode. The signal level relationship will be different if a Dialnorm value or a Programme Reference Level other than that corresponding with the EBU R 128 Target Level of -23 LUFS is used. In case of programme material, the output level of the decoder may peak higher than the input level due to overshoots. These overshoots should however remain below the levels indicated by the red bars. Especially if relatively low bitrates are being applied, the graph can be used to compare the measured levels with the specified maximum levels to check if it is

Subject	Description
	necessary to decrease the limiter level. For HE-AAC, MPEG-1 Layer II and for the DD/DD+ decoder in RF Mode, different maximum peak levels are shown. Due to the internal loudness reference level of -20 LUFS of the DD/DD+ decoder being decreased to -23 LUFS by a software-changeable attenuator of 3 dB (see § 6 for details), the actual clipping level using a 1 kHz sine wave is -3 dBTP. The maximum recommended output level of the DD/DD+ RF Mode decoder for programme material is therefore indicated as -4 dBTP (1 dB lower). The recommended studio limiter level of -3 dBTP exceeds that maximum threshold. Nevertheless this is not an error. As the DD/DD+ system contains its own internal overload protection, the maximum peak level for encoding does not have to be decreased.
Outputs: HDMI/SPDIF/ HDMI ARC in Home Theatre Mode	This is the digital PCM signal level of the decoded audio, measured in dBTP at the HDMI, the SPDIF or the HDMI ARC if the IRD, IDTV or media player operates in Home Theatre Mode (HTM, § 6.4.1 and § 7.4.1 can be checked for details). In IRD/IDTV output modes that support codec bitstreams, the internal DD/DD+ or HE-AAC decoder shall not be used to feed the SPDIF and HDMI (ARC), unless a specific application such as Audio Description (which requires it) is being used, or where the E-EDID query identifies the sink as supporting basic audio only. There are two rows displaying the output levels depending on the Home Theatre Mode setting in the installation menu of the IRD, IDTV or media player (*default or offset*).
Outputs: HDMI in TV Mode/ SPDIF/HDMI ARC/ AES3 in Stereo Mode	This is the digital PCM signal level of the decoded audio, measured in dBTP at the HDMI if the IRD operates in TV Mode, or at the SPDIF, HDMI ARC or AES3 interface if the IRD or IDTV operates in Stereo Mode (see § 6.4.1 and § 7.4.1 for details).
Outputs: SCART or RCA interface	This is the analogue RMS signal level, measured in millivolts, of the decoded audio on the SCART or RCA outputs of the IRD, IDTV or media player. It also represents the analogue RMS signal level, measured in millivolts, on the SCART or RCA outputs of a television set using the HDMI or SPDIF interface as an input.
Output: XLR or similar interface	This is the decoded, analogue RMS audio signal measured on the (balanced) XLR outputs (or similar alternative) of the professional IRD. As an example, the relative levels in dBu0s are shown as absolute levels in dBu, if a normalisation factor of respectively 0 dBrs or -3 dBrs is applied, as specified in ITU-R BS.645.
RF Modulation: Systems B, B1, D, D1, G, H, K, K1, I and I1	This is the peak FM deviation, measured in kHz, as a result of the decoded audio, including the effect of pre-emphasis gain, being connected to the input of the RF modulator. The red line represents the maximum peak level. In practice, this means that the signal shall be passed through an end-stage pre-emphasis peak limiter set to a level of -6.7 dBTP before being applied to the modulator. The purple line indicates the alignment for the limiter level. The limiter could also be built into the modulator itself. Because of reasons described in § 5.2 it could be necessary to decrease the limiter level.

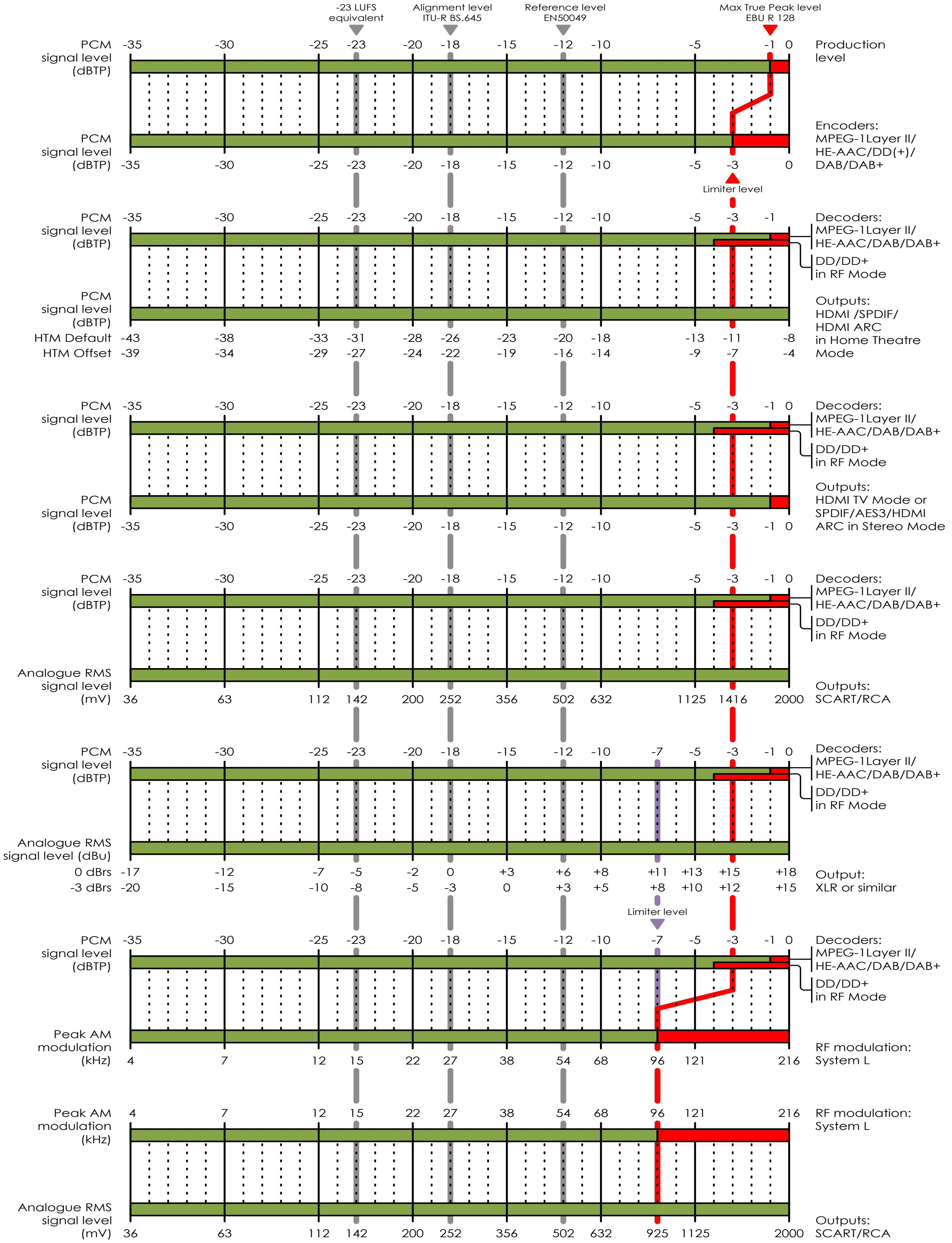
Subject	Description
RF Modulation: System L	This is the peak AM modulation depth measured in percent as a result of the decoded audio being fed to the input of the RF modulator. The red line represents the maximum peak level. In practice this means that the signal shall be passed through an end-stage limiter set to a level of -7 dBTP before being applied to the modulator. The limiter could also be built into the modulator itself. Because of reasons described in § 5.2 it could be necessary to decrease the limiter level.
RF Modulation: NICAM 728 systems B, B1, D1, G, H, K1 and L	This is the Digital Coding Level inside the NICAM modulator measured in dBTP as a result of the decoded audio, including the effect of pre-emphasis gain, being connected to the input of the NICAM RF modulator. The red line represents the maximum peak level. In practice, this means that the signal shall be passed through an end-stage pre-emphasis peak limiter set to a level of -2 dBTP before being applied to the modulator. The limiter could also be built into the modulator itself. Because of reasons described in § 5.2 it could be necessary to decrease the limiter level.
RF Modulation: NICAM 728 systems I and I1	This is the Digital Coding Level inside the NICAM modulator measured in dBTP as a result of the decoded audio, including the effect of pre-emphasis gain being connected to the input of the NICAM RF modulator. As there is more headroom in NICAM system I compared to other systems, an end-stage pre-emphasis peak limiter is optional.
RF Modulation: FM stereo radio	This is the peak FM deviation measured in kHz as a result of the decoded audio, including the effect of pre-emphasis gain, being connected to the input of the FM modulator. The red line represents the maximum peak level after allowing for a reservation of 10 kHz for additional signals such as RDS and pilot tone. Any other value only influences the maximum headroom available for audio and the corresponding limiter level. In this case it means that the signal shall be passed through an end-stage pre-emphasis peak limiter set to a level of -6.6 dBTP before being applied to the modulator. The limiter could also be built into the modulator itself. Because of reasons described in § 5.2 it could be necessary to decrease the limiter level. There are two rows displaying the audio-related multiplex (MPX) levels and the total MPX level. The latter represents the FM deviation including audio, pilot tone, RDS and other additional signals.
RF Modulation: FM mono radio	This is the peak FM deviation measured in kHz as a result of the decoded audio, including the effect of pre-emphasis gain, being connected to the input of the FM modulator. The red line represents the maximum peak level assuming that there is no bandwidth reserved for additional signals such as RDS. This means that the signal shall be passed through an end-stage pre-emphasis peak limiter set to a level of -5.4 dBTP before being applied to the modulator. The limiter could also be built into the modulator itself. Because of reasons described in § 5.2 it could be necessary to decrease the limiter level. Any added signal in the spectrum only decreases headroom and the corresponding limiter level. To simplify operational procedures, the same limiter level as for FM stereo radio may also be used for mono signals.
IRD outputs: SCART *or* RCA interface	This is the analogue RMS signal level, measured in millivolts, of the demodulated audio on the SCART or RCA outputs using the built in RF tuner of a television set or a recording device, or the RCA outputs of an FM radio receiver.

5.5 Level alignment in systems B, B1, D, D1, G, H, K, K1, I and I1



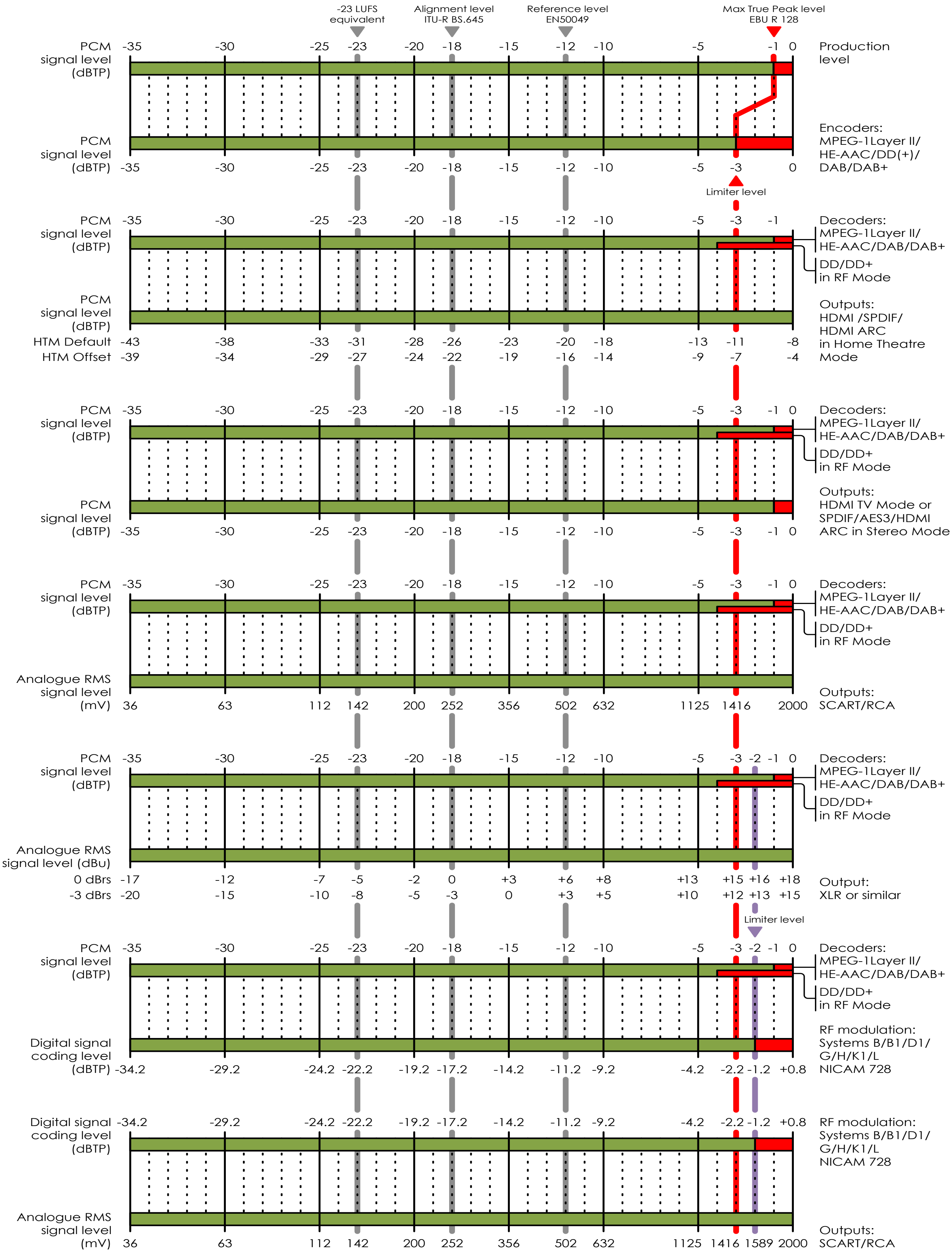
Conditions: 1 kHz sine wave in phase on Left and Right channel only, Dialnorm = -23, Mode = RF, DRC = None, PRL = -23, TL = -23

5.6 Level alignment in television system L



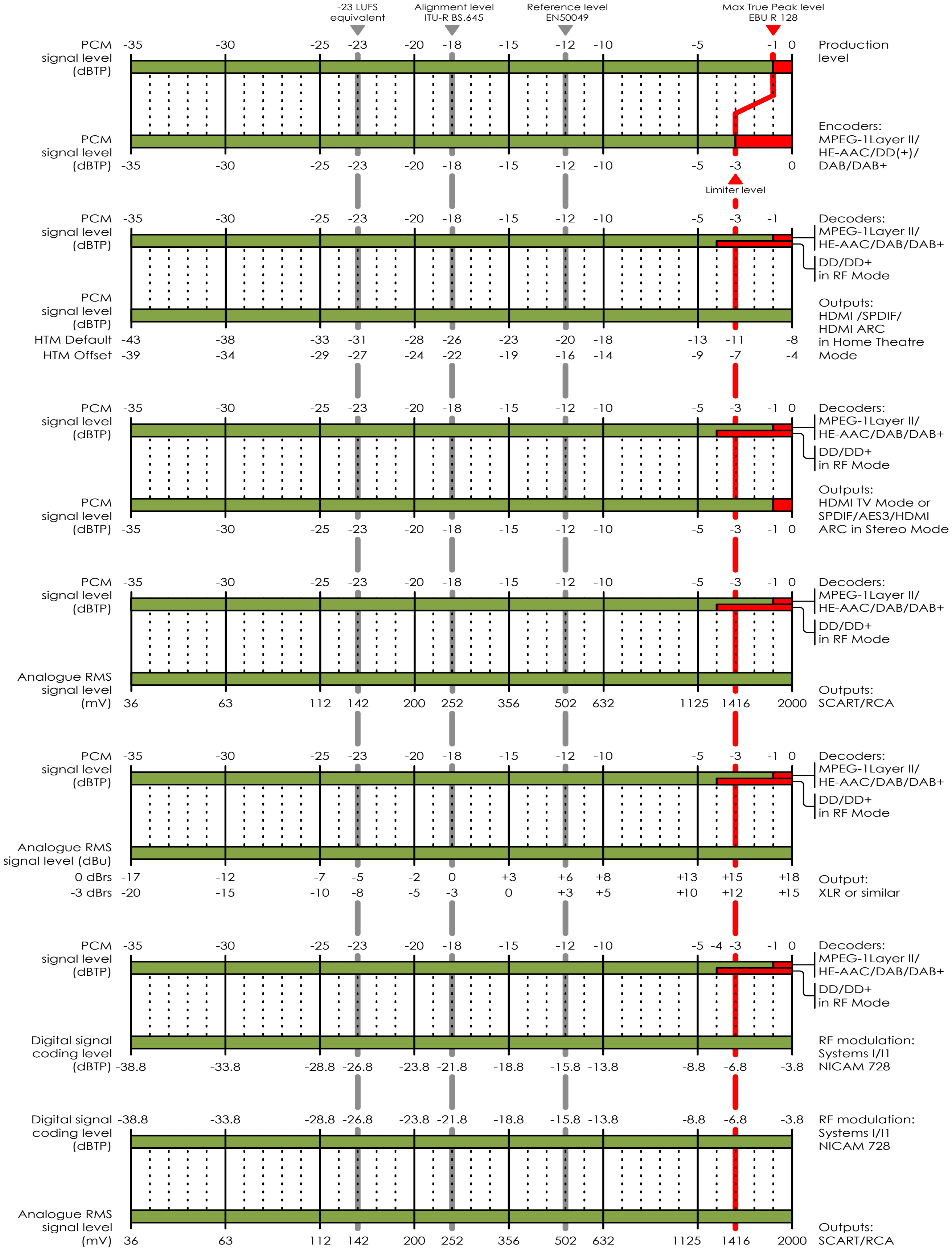
Conditions: 1 kHz sine wave in phase on Left and Right channel only, Dialnorm = -23, Mode = RF, DRC = None, PRL = -23, TL = -23

5.7 Level alignment in NICAM television systems B, B1, D1, G, H, K1 and L



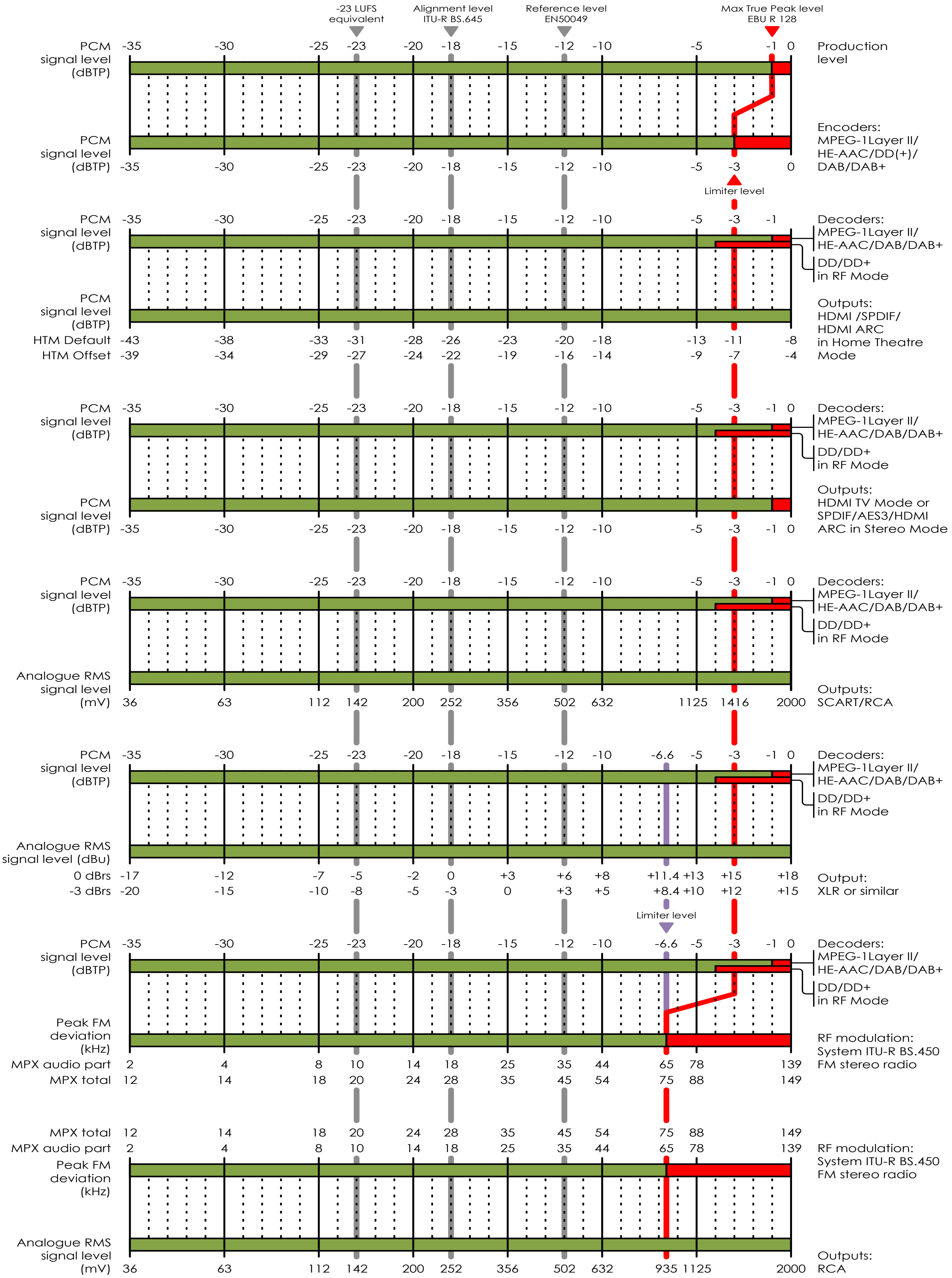
Conditions: 1 kHz sine wave in phase on Left and Right channel only, Dialnorm = -23, Mode = RF, DRC = None, PRL = -23, TL = -23

5.8 Level alignment in NICAM television system I and I1



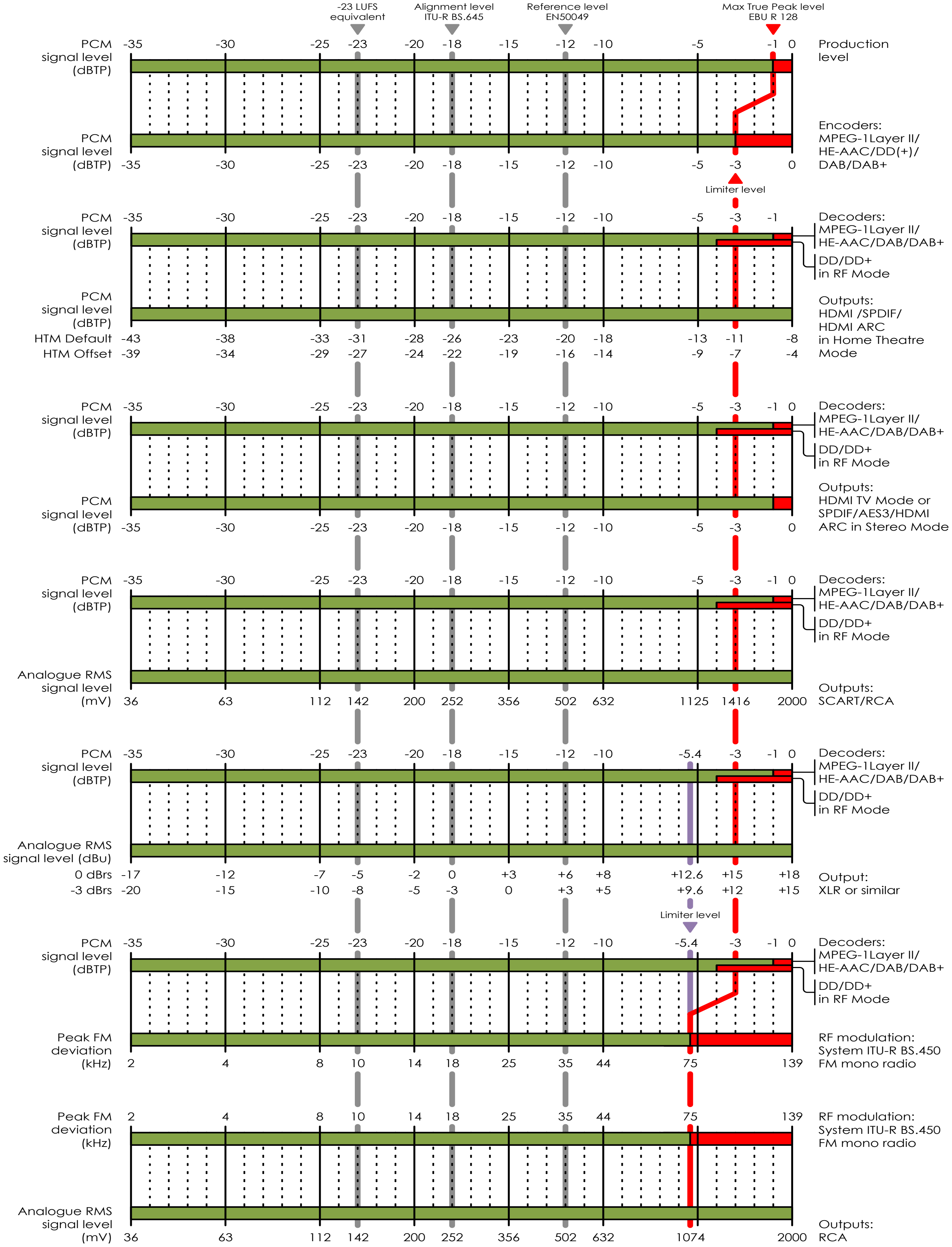
Conditions: 1 kHz sine wave in phase on Left and Right channel only, Dialnorm = -23, Mode = RF, DRC = None, PRL = -23, TL = -23

5.9 Level alignment in FM stereo radio



Conditions: 1 kHz sine wave in phase on Left and Right channel only, Dialnorm = -23, Mode = RF, DRC = None, PRL = -23, TL = -23

5.10 Level alignment in FM mono radio



Conditions: 1 kHz sine wave in phase on Left and Right channel only, Dialnorm = -23, Mode = RF, DRC = None, PRL = -23, TL = -23

6. Set-top boxes and professional integrated receiver decoders

6.1 Application

The guidelines described in this section are applicable to set-top boxes and to professional integrated receiver decoders, where the design allows the opportunity to change audio levels by means of a software update. A set-top box is referred to as an Integrated Receiver Decoder (IRD). For Integrated Digital Televisions (IDTVs) see § 7. The guidelines affecting loudness normalisation shall also be included in EBU Tech 3333, revision 1 [18], the EBU HDTV Receiver Requirements.

6.2 Audio systems

In this section, two transmission variants are distinguished. The audio signal could be carried by System A or B, as determined for the relevant network. An IRD can feature either one of the following variants, or both:

- System A featuring MPEG-1 Layer II and Dolby Digital (DD) or Dolby Digital Plus (DD+)
- System B featuring MPEG-1 Layer II and HE-AAC, optionally transcoded to Dolby Digital (DD) or DTS

6.3 Line Mode and RF Mode

The terms ‘RF Mode’ and ‘Line Mode’ are described in Dolby Technical Bulletin 11 and in other Dolby guidelines. Line Mode uses an internal loudness level equivalent to the Sound Reproduction Level of -31 LUFS. In DD/DD+ decoders operating in RF Mode this level is raised to -20 LUFS with a compressed dynamic range, which is meant to be more compatible with signal levels used in analogue transmission. To be compliant with the EBU R 128 Target Level, this document specifies that the loudness level of the decoder operating in RF Mode be decreased to -23 LUFS by use of a software-adjustable attenuator of 3 dB.

6.4 Level adaptation

An output-dependent configurable setting shall be implemented in the user menu that switches the PCM loudness level by an amount depending on the connected equipment. Basically, the user chooses during installation what kind of equipment is connected to the SPDIF and the HDMI. Subsequently, the IRD applies the correct level adjustment. It is suggested that a ‘wizard’ procedure be included to assist the user by displaying generic images of the connected equipment. For devices featuring an output for headphones, the audio alignment shall be the same as for the analogue line outputs. For remarks about the IRD volume control, see § 6.4.7.

Note 1: The menu structure for HDMI outputs could be replaced in the future by adaptation of the HDMI specification so that identification of connected equipment and the control of the corresponding loudness levels can be handled automatically.

6.4.1 Level adaptation settings to be implemented in the installation menu

The following base menu settings are recommended for IRDs featuring an HDMI output:

Item	Choice	Description
HDMI DEVICE	TELEVISION	This setting is intended if a television set is directly connected to the IRD using HDMI. It is backwards compatible with television sets which are not compliant with EBU Tech 3344 and it is the recommended setting for the installed base as well as new devices not feeding a home theatre device. The PCM loudness reference level is -23 LUFS. The IRD shall output only PCM signals to the HDMI, even where the E-EDID query identifies the sink as supporting compressed audio.
HDMI DEVICE	TELEVISION → HOME THEATRE	This setting is intended for a connected television set that is compliant with EBU Tech 3344. Subsequently, a home theatre device can be connected to the television set if it features an SPDIF output or HDMI ARC. The PCM loudness reference level is -23 LUFS. Both PCM signals as well as codec bitstreams are supported, except where the E-EDID query identifies the sink as supporting basic audio only. The connected television set needs to decrease the PCM levels on its SPDIF and HDMI ARC output to correctly pass-through audio to home theatre equipment (see § 7 for details).
HDMI DEVICE	HOME THEATRE → TELEVISION	This setting is intended if a home theatre device is directly connected to the IRD using HDMI. Subsequently, the television set can be connected to the home theatre device if both feature HDMI. To avoid loudness differences, the PCM loudness reference level is brought in line with the Sound Reproduction Level of -31 LUFS, the internal loudness level used in DD/DD+ codecs. Both PCM signals as well as codec bitstreams are supported, except where the E-EDID query identifies the sink as supporting basic audio only.
HDMI DEVICE	HOME THEATRE (OFFSET MODE) → TELEVISION	This is a similar application to the previous one, but now intended for a home theatre device that uses a PCM loudness reference level of -27 LUFS (see *Note 1*).
HDMI DEVICE	NONE	This setting can be used if no device is connected to HDMI. It is included to support the user with a complete set of choices and is a logical setting if, for example, the television set is connected to the SCART interface. The audio output signal on the HDMI shall be muted in this mode to support the effect of this choice.

The following additional menu setting is recommended for IRDs featuring an HDMI output and specific functionality:

Item	Choice	Description
HDMI DEVICE	HOME THEATRE (PCM MCA MODE) → TELEVISION	This setting is only relevant if the IRD features an additional multi-channel decoder (see § 6.4.2 for details). This setting is intended if a home theatre device is directly connected to the IRD using HDMI. Subsequently, the television set can be connected to the home theatre device if both feature HDMI. To avoid loudness differences, the PCM loudness reference level is brought in line with the Sound Reproduction Level of -31 LUFS, the internal loudness level used in DD/DD+ codecs. The IRD shall output only PCM signals to the HDMI, even if the E-EDID query identifies the sink as supporting compressed audio.

The following base menu settings are recommended for IRDs featuring an SPDIF output:

Item	Choice	Description
SPDIF DEVICE	HOME THEATRE	This setting is intended if a home theatre device is directly connected to the IRD using SPDIF. To avoid loudness differences, the PCM Loudness reference level is brought in line with the Sound Reproduction Level of -31 LUFS, the internal loudness level used in DD/DD+ codecs. Both PCM signals as well as codec bitstreams are supported.
SPDIF DEVICE	HOME THEATRE (OFFSET MODE)	This is a similar application to the previous one, but now intended for a home theatre device that applies a PCM loudness reference level of -27 LUFS (see *Note 1*).
SPDIF DEVICE	STEREO EQUIPMENT (PCM)	This setting is intended if a PCM stereo device such as an amplifier or a recording device is directly connected to the IRD using SPDIF. The PCM loudness reference level is -23 LUFS. The IRD shall only output PCM signals to the SPDIF in this mode.
SPDIF DEVICE	NONE	This setting can be used if no device is connected to the SPDIF. This setting is included to support the user with a complete set of choices. The output signal on the SPDIF shall be muted in this mode to support the effect of this choice.

The following additional menu setting is recommended for IRDs featuring an SPDIF output and specific functionality:

SPDIF DEVICE	HOME THEATRE (HE-AAC MODE)	Only relevant for System B IRDs. This setting enables that HE-AAC codec bitstreams are output instead of transcoded DD bitstreams for home theatre equipment that supports HE-AAC decoding.

Note 1: It appears that a substantial number of home theatre devices process PCM input signals with a fixed 4 dB offset compared to the output of the DD/DD+ decoder. Among, but not limited to, them is equipment certified to THX specifications. This state of affairs applies to old and current designs. This offset for PCM input signals is considered to be undesirable. It is hoped that this document can be taken into account for future specifications for home theatre equipment, which means that this fixed gain offset for

PCM signals shall not be present. Nevertheless, to achieve loudness consistency with the major part of home theatre devices, an alternative offset mode is included in this document based on the use of this paradigm.

Note 2: If it cannot be avoided, menu entries may be made depending on each other to reduce internal complexity. SPDIF modes may be limited to NONE if the user has chosen one of the following settings:

HDMI DEVICE = TELEVISION → HOME THEATRE

HDMI DEVICE = TELEVISION → HOME THEATRE (OFFSET MODE)

HDMI DEVICE = HOME THEATRE → TELEVISION

HDMI DEVICE = HOME THEATRE (OFFSET MODE) → TELEVISION

HDMI DEVICE = HOME THEATRE → TELEVISION (PCM MCA MODE)

It is recommended to restore the last applied choice for SPDIF DEVICE once the user chooses HDMI DEVICE = TELEVISION or HDMI DEVICE = NONE.

Note 3: In modes that support codec bitstreams, the internal DD/DD+ or HE-AAC decoder shall not be used to feed the SPDIF and HDMI, unless a specific application is being used like Audio Description which requires it, or where the E-EDID query identifies the sink as supporting basic audio only.

Note 4: In case the HDMI E-EDID query identifies the sink as supporting basic audio only, the IRD shall block codec bitstreams, but shall continue to use the same PCM attenuation for that setting. This shall be done to avoid wrong audio levels after a failure reading the E-EDID.

Note 5: If the E-EDID query identifies the sink as supporting HE-AAC audio, HE-AAC codec bitstreams are output instead of transcoded DD bitstreams (only relevant for the HDMI output on System B IRDs).

Note 6: It is recommended <u>not</u> to force the user choice for the HDMI setting based on the E-EDID query, as in practice errors can occur during the HDMI handshake procedure, which could result in a wrong choice and consequently result in loudness jumps.

Note 7: It is recommended to program HDMI DEVICE = TELEVISION and SPDIF DEVICE = HOME THEATRE as a factory default.

Note 8: The traditionally applied user setting to control the preference for the use of the internal DD/DD+ or HE-AAC decoder or outputting codec bitstreams on the HDMI and SPDIF is obsolete using the new paradigm described in this document. This choice is fully integrated in the settings described in this section.

6.4.2 Additional information to implement the adaption in IRDs

The IRD shall adjust the output level of all built-in audio decoders according to the figures in § 6.4.4 and § 6.4.5 so that the perceived programme loudness is consistent for all audio-coding schemes. The following information clarifies how level adaptation shall be implemented:

- **MPEG-1 Layer II processing (System A and B)**

  The IRD shall include a PCM level attenuator to reduce the level of decoded MPEG-1 Layer II audio at the SPDIF and HDMI for modes indicated in § 6.4.1 intended for the use of home theatre equipment. The gain reduction steps (0, -4 and -8 dB) shall be programmable to allow for a potential future change. This facility may be provided by a software update. Gain reduction shall not be applied to the stereo analogue outputs. It shall not be applied either to the HDMI or SPDIF for modes where PCM attenuation is indicated as 0 dB in § 6.4.6.

- **DD/DD+ processing (System A)**

  The IRD featuring System A shall include a PCM level attenuator to reduce the level of decoded DD/DD+ audio in RF Mode operation to accommodate reproduction in line with the Target Level of -23 LUFS, which means an attenuation of 3 dB. This gain reduction shall be programmable to allow for a potential future change. This facility may be provided by a software update. From that point, for modes and applications where use of the internal decoder is required to supply the SPDIF and/or HDMI and for the analogue outputs, the IRD shall apply the same procedure for decoded DD/DD+ signals as for MPEG-1 Layer II.

  On the SCART and stereo analogue RCA outputs, a PCM loudness level of -23 LUFS shall be applied by using the main decoder. On IRDs featuring multi-channel PCM or multi-channel analogue outputs, the device shall include an additional DD/DD+ decoder in Line Mode operation applying a PCM equivalent loudness level of -31 LUFS. This multi-channel decoder shall have a user switchable down-mix mode to enable a default stereo reproduction on the left and right loudspeaker of the multi-channel sound system (see § 6.11 about applying the correct loudness level when down-mixing).

  If DD/DD+ codec bitstreams are passed through to the SPDIF or HDMI, the IRD shall neither change the audio content nor the accompanying metadata.

- **HE-AAC processing (System B)**

  The IRD featuring System B shall apply a PCM loudness reference level equivalent to the EBU R 128 Target Level. This shall be achieved by applying the Programme Reference Level descriptor (*prog_ref_level*, specified in ISO/IEC 14496-3) of the HE-AAC bitstream and the Decoder Target Level descriptor (*target_level*, specified in ISO/IEC 144496-3) at a level of -23 LUFS. From that point, for the analogue outputs and for modes and application where use of the internal decoder is required, the IRD shall apply the same procedure for decoded HE-AAC signals as for MPEG-1 Layer II.

  On the SCART and stereo analogue RCA outputs, a PCM equivalent loudness level of -23 LUFS shall be applied by using the main decoder. On IRDs featuring multi-channel PCM or multi-channel analogue outputs, the IRD shall use an additional HE-AAC decoder. This decoder shall use a PCM loudness level equivalent to the Sound Reproduction Level of -31 LUFS by applying the Programme Reference Level descriptor (*prog_ref_level*, specified in ISO/IEC 14496-3) of the HE-AAC bitstream and the Decoder Target Level descriptor (*target_level*, specified in ISO/IEC 144496-3) at a level of -31 LUFS. This multi-channel decoder shall have a user switchable down-mix mode to enable a default stereo reproduction on the left and right loudspeaker of the multi-channel sound system (see § 6.11 about applying the correct loudness level when down-mixing).

  If an HE-AAC bitstream does not contain loudness metadata, the IRD shall follow the MPEG-4 standard by assuming that the audio is already at the EBU R 128 Target Level.

  If HE-AAC codec bitstreams are passed through to the SPDIF or HDMI, the IRD shall neither change the audio content nor the accompanying metadata.

  In case of transcoding from HE-AAC to DD, the IRD shall preserve the level of the audio and transcode the accompanying metadata to ensure correct reproduction level in a downstream decoder. If the HE-AAC stream does not contain loudness metadata, the IRD shall signal a Dialnorm of -23 in the DD bitstream assuming a Programme Reference Level of the incoming audio of -23 LUFS.

Note 1: Manufacturers wishing to use the "Dolby Pulse" implementation of HE-AAC should consult Dolby (in particular, Technical Bulletin 11) for information about the additional steps needed to meet the requirements of this Tech Doc.

6.4.3 Notes about the graphical representation of the audio processing within devices

The figures in the following sections show a graphical representation of the audio processing within the device. The following notes are relevant for these figures:

Note 1: The term 'L_K' refers to loudness. On analogue outputs, the term '$L_K \approx$' refers to the loudness of the decoded PCM signal based on the mapping specified in this document between the levels in the analogue and digital domain.

Note 2: The device can have more or fewer input and output interfaces and more or fewer features, depending on model and application.

6.4.4 Graphical representation of the level adaptation in the System A IRD

The following figure shows a graphical representation of the audio processing for a System A IRD featuring DD/DD+. Its application to IRDs which operate in a different transmission system, which offer fewer or more options or which offer Audio Description by using two MPEG-1 Layer II and/or two DD/DD+ decoders, can be derived from this figure.

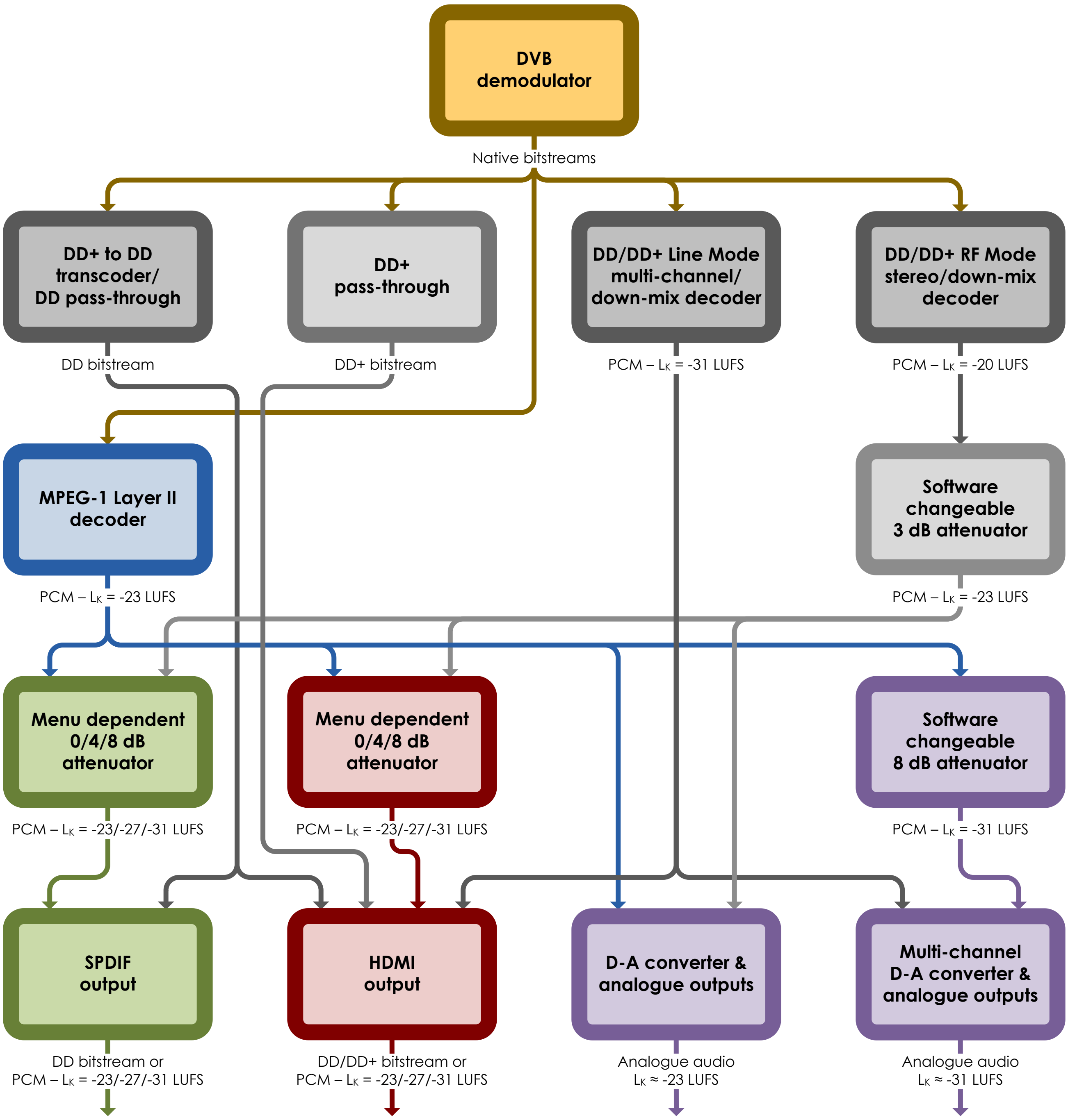


Figure 6.4.4.1: Audio processing within the System A IRD

6.4.5 Graphical representation of the level adaptation in the System B IRD

The following figure shows a graphical representation of the audio processing for a System B IRD featuring HE-AAC. Its application to IRDs which operate in a different transmission system, which offer fewer or more options or which offer Audio Description by using two MPEG-1 Layer II and/or two HE-AAC decoders, can be derived from this figure.

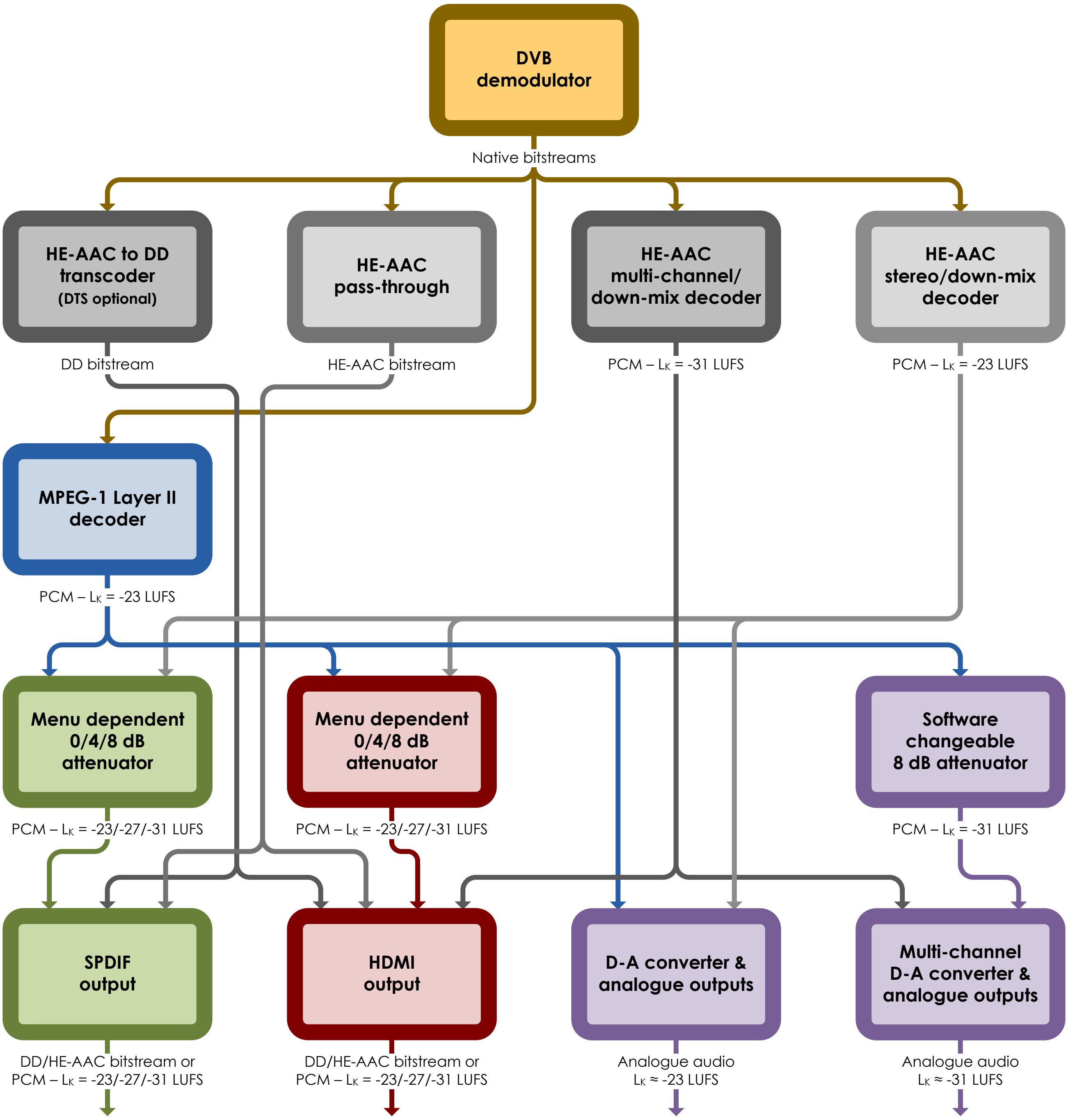


Figure 6.4.5.1: Audio processing within the System B IRD

6.4.6 Overview of the level adaptation required for IRD installation menu settings

Interface	Setting	PCM loudness level (LUFS)	PCM attenuation MPEG-1 Layer II decoder (dB)	PCM attenuation DD/DD+ decoder (dB)	PCM attenuation HE-AAC decoder (dB)	Codec bitstream support
HDMI	TELEVISION	-23	0	3	0	No
HDMI	TELEVISION → HOME THEATRE	-23	0	3	0	Yes
HDMI	HOME THEATRE → TELEVISION	-31	8	11	8	Yes
HDMI	HOME THEATRE (OFFSET) → TELEVISION	-27	4	7	4	Yes
HDMI	HOME THEATRE (PCM MCA) → TELEVISION (1)	-31	8	11	8	No
SPDIF	HOME THEATRE	-31	8	11	8	Yes
SPDIF	HOME THEATRE (OFFSET)	-27	4	7	4	Yes
SPDIF	HOME THEATRE (HE-AAC) (2)	-31	8	11	8	Yes
SPDIF	STEREO EQUIPMENT (PCM)	-23	0	3	0	No

Note 1: Only relevant for IRDs featuring a multi-channel decoder.

Note 2: Only relevant for System B IRDs.

For specific details, please see the notes in § 6.4.1. The relationship between discrete input and output levels can be found in § 5.

6.4.7 IRD volume control

It is strongly recommended that the audio levels inside the IRD be unaffected by use of its volume control. Instead, the volume control of the IRD should preferably use the remote control code of other equipment (e.g. television set and/or home theatre equipment) or use the HDMI Consumer Electronics Control (CEC) feature. It must be emphasised that this concept optimises comfort, as the user is able to control the volume for all signal formats (e.g. PCM and passed-through codec bitstreams). It also avoids conflicts between the volume setting of the IRD and the volume settings of both the television set and home theatre equipment as well as avoiding conflict with the alignment of other sources to that equipment. To use this functionality, the IRD remote control must have an option to choose between the IRD and a connected home theatre device. The CEC feature can handle this automatically. Audio mute functionality within the IRD can be preserved. For an IRD offering Audio Description, the volume control can still be used to adjust its headphone output, once the device detects that a headphone has been connected.

Note 1: For (older) IRD models that do not have the ability to use the remote control code or HDMI CEC, it is recommended only to apply the volume control if the setting TELEVISION is chosen for the HDMI output, in order to be able to offer comfort to the user by another means. This reduces the negative impact of changing the levels inside the IRD. The PCM levels on the SPDIF and on the HDMI using other settings than TELEVISION shall be left unaffected.

6.4.8 Analogue output level relationship

The output level on the SCART and RCA analogue interface shall be 2.0 V RMS using a sine wave at 1 kHz encoded at 0 dBTP (see *Note 1*). Accordingly, 0 dBTP corresponds to a signal peak level of 2.83 V. This specific output level is required in order to align with the analogue modulation levels specified in this document. This level alignment is compatible with CENELEC EN50049. A graphical representation of level relationships for several television systems and for FM radio is shown in § 5.

The following level alignment shall be applied to SCART/RCA interfaces:

Level alignment for the analogue and SCART input & output (1, 2, 3)	-12 dBTP using a 1 kHz sine wave results in an RMS signal level of 502 mV (±1 dB).

The following level alignment shall be applied to (balanced) XLR interfaces (or similar alternative) on a professional IRD:

Level alignment for the XLR analogue input & output (1, 3, 4)	-12 dBTP using a 1 kHz sine wave results in an RMS signal level of +6 dBu (±1 dB), if a normalisation factor of 0 dBrs is applied.
	-12 dBTP using a 1 kHz sine wave results in an RMS signal level of +3 dBu (±1 dB), if a normalisation factor of -3 dBrs is applied. The term dBrs is specified in ITU-R BS.645.

Note 1: True Peak Level is the maximum peak level of an audio signal measured with an oversampling True Peak Meter. If a True Peak meter is not available, a sine wave at 997 Hz, encoded at the specified level in dBFS, may be used for reference.

Note 2: For digital processing: To prevent clipping on the analogue inputs, an input attenuator of, for example, 6 dB can be applied followed by a digital gain shift (of 6 dB) to arrive at the same value.

Note 3: To reduce attenuation of the output level, it is recommended that the source impedance of an output interface is as low as possible, so long as the output remains unconditionally stable. CENELEC EN50049 specifies output impedance between 300 Ω and 1000 Ω for the SCART audio output interface. It is recommended that 300 Ω be applied to reduce variations in loudness level.

Note 4: If the XLR outputs (or similar alternative) of the professional IRD feed the input of a head-end RF modulator, it is strongly recommended to compensate the level uncertainty due to source and load impedances and other variations.

6.4.9 Additional setting to be implemented in the user menu of the professional IRD

The following setting applies a normalisation factor of respectively 0 dBrs or -3 dBrs as specified in ITU-R BS.645:

Item	Choice
ANALOGUE OUTPUT LEVEL	0 dBrs or -3 dBrs

Note 1: Another way to specify this is by setting the clipping level to +18 dBu or +15 dBu respectively.

6.5 Audio preference settings

A service can supply more than one audio stream. In general, the user may or may not have a preference for DD/DD+ or HE-AAC encoded streams (if they are supplied with the service) instead of MPEG-1 Layer II. A general setting in the user preference menu assists the user in automatically choosing the preferred setting if a new service is added. It is strongly recommended to implement a service dependent setting to override the general setting. The following sections describe this in more detail.

6.5.1 Audio preference settings to be implemented in the user preference menu

Item	Choice	Description
AUDIO STREAM	MPEG-1 LAYER II *or* DOLBY DIGITAL *or* HE-AAC o*r* AUTO	Defines the general preference of the user for the MPEG-1 Layer II, the DD/DD+ or the HE-AAC stream if it is supplied with the service. If set on AUTO, the IRD follows the PSI/SI information.

6.5.2 Audio preference settings to be implemented in the service dependent user menu

Item	Choice	Description
AUDIO STREAM	MPEG-1 LAYER II *or* DOLBY DIGITAL *or* HE-AAC o*r* AUTO	Defines the preference of the user for the MPEG-1 Layer II, the DD/DD+ or the HE-AAC stream if it is supplied with the service. If set on AUTO, the IRD follows the setting in the user preference menu.

Note 1: This setting overrides the general preference as set in the installation menu and shall be stored in non-volatile memory so that the IRD returns to the same setting after switching services and after power up. IRDs featuring recording facilities shall store metadata together with the (transport stream) file to indicate and apply the preference.

6.6 Audio resolution

Processing with the IRD should retain at least 24 bit resolution. Dithering shall be used when resolution is reduced.

6.7 Audio processing in the professional IRD

A professional IRD which is to be used in studios and distribution centres shall behave as a set-top box where audio processing is concerned. For the specific application where the professional IRD is integrated in an RF modulator, see § 4.5.

6.8 DD/DD+ Dynamic Range Control (DRC)

For System A IRDs, the decoder shall follow the Dolby metadata. If, for example, the DD/DD+ encoder used the DRC=NONE profile, the decoder shall not apply any compression other than overload protection.

- **RF Mode operation**

  The main DD/DD+ decoder in the IRD used for stereo reproduction shall apply the RF Mode dynamic range metadata.

- **Line Mode operation**

  For IRDs offering an additional multi-channel/stereo down-mix DD/DD+ decoder that applies a PCM loudness level of -31 LUFS, the IRD shall apply the Line Mode dynamic range metadata by default. DRC settings that apply scaling of the gain reduction are optional. It shall always be possible for the user to switch off DRC and the IRD shall store this setting in non-volatile memory, so that it keeps the same setting after power up.

6.9 HE-AAC Dynamic Range Control

For System B IRDs, the main decoder used for stereo reproduction shall follow the metadata in the *dynamic_range_info()* field of the ISO/IEC 14496-3 stream. For IRDs offering an additional multi-channel/stereo down-mix HE-AAC decoder that applies a PCM loudness level of -31 LUFS, it shall always be possible for the user to switch off DRC and the IRD shall store this setting in non-volatile memory, so that it keeps the same setting after power up. For devices offering a monophonic, RF modulated analogue output, the signal on this output should have the DRC metadata '*compression_value*' described in ETSI 101 154 Annex C.5.2.5 applied. When this metadata is not present, the IRD shall revert to the DRC metadata in the *dynamic_range_info()* field of ISO/IEC 14496-3 for this output.

6.10 Additional Dynamic Range Control or audio-enrichment features

Additional proprietary dynamic range control and audio-enrichment functionality shall be optional. These features shall always be switched off by default. For specific listening conditions, for example late at night or in a bedroom, it is useful to implement an additional DRC application. This so-called 'Night Mode' shall not just be based on progressively scaled DD/DD+ DRC metadata, as this might not be active (for example if the DD/DD+ encoder applies the profile DRC=NONE) or not applicable (for example using MPEG-1 Layer II audio).

6.11 Down-mixing of multi-channel audio

Multi-channel broadcasts are often presented in the home on two loudspeakers. To achieve this, the (typically) five channels are combined into two by adding a certain amount of the surround channels' signal to the front channels and some of the centre channel's signal to left and right. The amounts may be controlled by down-mix coefficients transmitted with the audio signal. In some broadcast recommendations there is ambiguity regarding the need to scale the down-mix coefficients in order to avoid signal overload, should all channels contain high-level signals. To maintain consistent signal level between down-mixed multi-channel programmes and native stereo programmes, this scaling should not be applied. The content provider shall ensure that sufficient headroom and/or dynamic range control values are included in the transmission to prevent any overload when down-mixing.

System B IRDs shall apply the down-mix parameters according to ETSI 101 154 Annex C 5.2.4, *down-mixing_levels_MPEG4* (the parameter with increased resolution over that in ISO/IEC 14496-3).

6.12 Interactive applications

Interactive applications on an IRD that make use of accompanying sound can be made consistent with EBU R 128 by normalising the audio in advance by, for example, an algorithm implemented in software. Care shall be taken within the design of the IRD so that the signal alignment corresponds to that of the broadcasted audio via all audio interfaces with the aim of achieving an equal integrated loudness level.

6.13 Internet applications

For IRDs featuring Internet and/or network access, loudness jumps can spoil the quality of experience as audio and video streams can be very loud. Although applications such as Internet access fall outside the scope of the current revision of this document, it is thought that it might be advantageous to include an attenuator for Internet and network stream decoder outputs that can be set by the user.

7. Television sets and IDTVs

7.1 Application

The guidelines described in this section are applicable to television sets and to Integrated Digital Televisions (IDTVs), where the design allows the opportunity to change audio levels by means of a software update. A television set featuring an HDMI input and built in DD/DD+ or HE-AAC decoders (or both) is also identified as an IDTV.

7.2 Audio systems

In this section, two transmission variants are distinguished. The audio signal could be carried by System A or B, as determined for the relevant network. An IDTV can feature either one of the following variants, or both:

- System A featuring MPEG-1 Layer II and Dolby Digital (DD) or Dolby Digital Plus (DD+)
- System B featuring MPEG-1 Layer II and HE-AAC, optionally transcoded to Dolby Digital (DD) or DTS

This section describes systems that can also have all or a part of the following audio system playback functionality:

- Analogue reception (AM or FM demodulation and NICAM decoding)
- PCM

In order to avoid complexity regarding loudness consistency, it is strongly recommended to consider television sets and IDTVs as devices restricted to stereo reproduction on (internal) loudspeakers and headphones. To enable multi-channel reproduction, the IDTV shall be connected to a home theatre device.

Note 1: A 'soundbar' connected to a television set shall be considered as a home theatre device.

7.3 Line Mode and RF Mode

The terms 'RF Mode' and 'Line Mode' are described in Dolby Technical Bulletin 11 and in other Dolby guidelines. In DD/DD+ decoders operating in RF Mode, which is the default mode for IDTVs, this level is raised to -20 LUFS with a compressed dynamic range, which is meant to be more compatible with signal levels used in analogue transmission. To be compliant with the EBU R 128 Target Level, this document specifies that the loudness level of the decoder operating in RF Mode be decreased to -23 LUFS by use of a software-adjustable attenuator of 3 dB.

7.4 Level adaptation

If the television set or IDTV features an SPDIF or an HDMI ARC output, a configurable setting shall be implemented in the user menu of the IDTV that switches the PCM loudness level by an amount depending on the connected equipment. Basically, the user chooses during installation what kind of equipment is connected to the SPDIF and HDMI ARC output. Subsequently, the television set or IDTV applies the correct level adjustment. It is suggested that a ‘wizard’ procedure be included to assist the user by displaying generic images of the connected equipment. For devices featuring an output for headphones, the audio alignment shall be identical as for the loudspeaker outputs.

Note 1: *The menu structure for HDMI inputs and outputs could be replaced in the future by adaptation of the HDMI specification so that identification of connected equipment and the control of the corresponding loudness levels can be handled automatically.*

7.4.1 Level adaptation settings to be implemented in the installation menu

The following menu settings are recommended for IDTVs featuring an SPDIF output:

Item	Choice	Remark
SPDIF DEVICE	HOME THEATRE	This setting is intended if a home theatre device is directly connected to the IDTV using the SPDIF. To avoid loudness differences, the PCM Loudness reference level is brought in line with the Sound Reproduction Level of -31 LUFS, the internal loudness level used in DD/DD+ codecs. Both PCM signals as well as codec bitstreams are supported.
SPDIF DEVICE	HOME THEATRE (OFFSET MODE)	This is a similar application to the previous one, but now intended for a home theatre device that applies a PCM loudness reference level of -27 LUFS (see *Note 1*).
SPDIF DEVICE	STEREO EQUIPMENT (PCM)	This setting is intended if a PCM stereo device like an amplifier or a recording device is directly connected to the IDTV using the SPDIF. The PCM loudness reference level is -23 LUFS. The IDTV shall output only PCM signals to the SPDIF in this mode.
SPDIF DEVICE	NONE	This setting can be used if no device is connected to the SPDIF. This setting is included to support the user with a complete set of choices. The output signal on the SPDIF shall be muted in this mode to support the effect of this choice.

The following additional menu setting is recommended for IDTVs featuring an SPDIF output and specific functionality:

SPDIF DEVICE	HOME THEATRE (HE-AAC MODE)	Only relevant for System B IDTVs. This setting enables that HE-AAC codec bitstreams are output instead of transcoded DD bitstreams for home theatre equipment that supports HE-AAC decoding.

Note 1: *It appears that a substantial number of home theatre devices process PCM input signals with a fixed 4 dB offset compared to the output of the DD/DD+ decoder. Among, but not limited to, them is equipment certified to THX specifications. This state of affairs applies to old and current designs. This offset for PCM input signals is considered to be undesirable. It is hoped that EBU R 128 and this document can be taken into account for future specifications of home theatre equipment, which means that this fixed gain offset for PCM signals shall not be present. Nevertheless, to achieve loudness consistency with the major part of home theatre devices, an alternative offset mode is included in this document, based on the use of this paradigm.*

Note 2: In modes that support codec bitstreams, the internal DD/DD+ or HE-AAC decoder shall not be used to feed the SPDIF, unless a specific application is being used, such as Audio Description.

Note 3: It is recommended to program SPDIF DEVICE = HOME THEATRE as a factory default.

A similar approach shall be applied for IDTVs featuring an Audio Return Channel via the HDMI:

Item	Choice	Remark
HDMI ARC DEVICE	HOME THEATRE	This setting is intended if a home theatre device is directly connected to the IDTV using the HDMI ARC. To avoid loudness differences, the PCM Loudness reference level is brought in line with the Sound Reproduction Level of -31 LUFS, the internal loudness level used in DD/DD+ codecs. Both PCM signals as well as codec bitstreams are supported.
HDMI ARC DEVICE	HOME THEATRE (OFFSET MODE)	This is a similar application to the previous one, but now intended for a home theatre device that applies a PCM loudness reference level of -27 LUFS (see *Note 1*).
HDMI ARC DEVICE	STEREO EQUIPMENT (PCM)	This setting is intended if a PCM stereo device is directly connected to the IDTV using the HDMI ARC. The PCM loudness reference level is -23 LUFS. The IDTV shall output only PCM signals to the HDMI ARC in this mode.
HDMI ARC DEVICE	NONE	This setting can be used if no device is connected to the HDMI ARC. This setting is included to support the user with a complete set of choices. The output signal on the HDMI ARC shall be muted and automatic switching of the loudspeakers of the IDTV shall be disabled in this mode to support the effect of this choice.

The following additional menu setting is recommended for IDTVs featuring an SPDIF output and specific functionality:

HDMI ARC DEVICE	HOME THEATRE (HE-AAC MODE)	Only relevant for System B IDTVs. This setting enables that HE-AAC codec bitstreams are output instead of transcoded DD bitstreams for home theatre equipment that supports HE-AAC decoding.

Note 1: It is recommended to programme HDMI ARC DEVICE = HOME THEATRE as a factory default.

Note 2: If it cannot be avoided, menu entries for SPDIF DEVICE and HDMI ARC DEVICE may be combined to reduce internal complexity. Consequently, the same level adaptation will be applied on both interfaces simultaneously.

7.4.2 Additional information to implement the adaptation in IDTVs

The IDTV shall adjust the output level of all built-in audio decoders according to the figures in § 7.4.4 and § 7.4.5 so that the perceived programme loudness is consistent for all audio-coding schemes. The following information clarifies how level adaptation shall be implemented:

- **MPEG-1 Layer II processing**

  The IDTV shall include a PCM level attenuator to reduce the level of decoded MPEG-1 Layer II audio at the SPDIF and HDMI ARC output for modes indicated in § 7.4.1 intended for the use of home theatre equipment. The gain reduction steps (0, -4 and -8 dB) shall be programmable to allow for a potential future change. This facility may be provided by a software update. Gain reduction shall not be applied to the stereo analogue outputs. It shall not be applied either to the SPDIF and HDMI ARC for modes where PCM attenuation is indicated as 0 dB in § 7.4.6.

- **DD/DD+ processing (System A)**

  The IDTV featuring System A shall include a PCM level attenuator to reduce the level of decoded DD/DD+ audio in RF Mode operation to accommodate reproduction in line with the Target Level of -23 LUFS, which means an attenuation of 3 dB. This gain reduction shall be programmable to allow for a potential future change. This facility may be provided by a software update. From that point, for modes and applications where use of the internal decoder is required to supply the SPDIF, HDMI ARC and for the analogue outputs, the IDTV shall apply the same procedure for decoded DD/DD+ signals as for MPEG-1 Layer II.

  If DD/DD+ bitstreams are passed through to the SPDIF or HDMI ARC output, the IDTV shall neither change the audio content nor the accompanying metadata.

- **HE-AAC processing (System B)**

  The IDTV featuring System B shall apply a PCM loudness reference level equivalent to the EBU R 128 Target Level. This shall be achieved by applying the Programme Reference Level descriptor (*prog_ref_level*, specified in ISO/IEC 14496-3) of the HE-AAC bitstream and the Decoder Target Level descriptor (*target_level*, specified in ISO/IEC 144496-3) at a level of -23 LUFS. From that point, for modes and applications where use of the internal decoder is required to supply the SPDIF, HDMI ARC and for the analogue outputs, the IDTV shall apply the same procedure for decoded HE-AAC signals as for MPEG-1 Layer II.

  If an HE-AAC bitstream does not contain loudness metadata, the IDTV shall follow the MPEG-4 standard by assuming that the audio is already at the EBU R 128 Target Level.

  If HE-AAC codec bitstreams are passed through to the SPDIF or HDMI, the IDTV shall neither change the audio content nor the accompanying metadata.

  In case of transcoding from HE-AAC to DD, the IDTV shall preserve the level of the audio and transcode the accompanying metadata to ensure correct reproduction level in a downstream decoder. If the HE-AAC stream does not contain loudness metadata, the IDTV shall signal a Dialnorm of -23 in the DD bitstream assuming a Programme Reference Level of the incoming audio of -23 LUFS.

Note 1: Manufacturers wishing to use the "Dolby Pulse" implementation of HE-AAC should consult Dolby (in particular Technical Bulletin 11) for information about the additional steps needed to meet the requirements of this Tech Doc.

7.4.3 Notes about the graphical representation of the audio processing within devices

The figures in the following sections show a graphical representation of the audio processing within the device. The following notes are relevant for these figures:

Note 1: The term 'L_K' refers to loudness. On analogue outputs, the term '$L_K \approx$' refers to the loudness of the decoded PCM signal based on the mapping specified in this document between the levels in the analogue and digital domain.

Note 2: The device can have more or fewer input and output interfaces and more or fewer features, depending on model and application.

7.4.4 Graphical representation of the level adaptation in the System A IDTV

The following figure shows a graphical representation of the audio processing for a System A IDTV featuring DD/DD+. Its application to IDTVs which operate in a different transmission system, which offer fewer or more options or which offer Audio Description by using two MPEG-1 Layer II and/or two DD/DD+ decoders, can be derived from this figure.

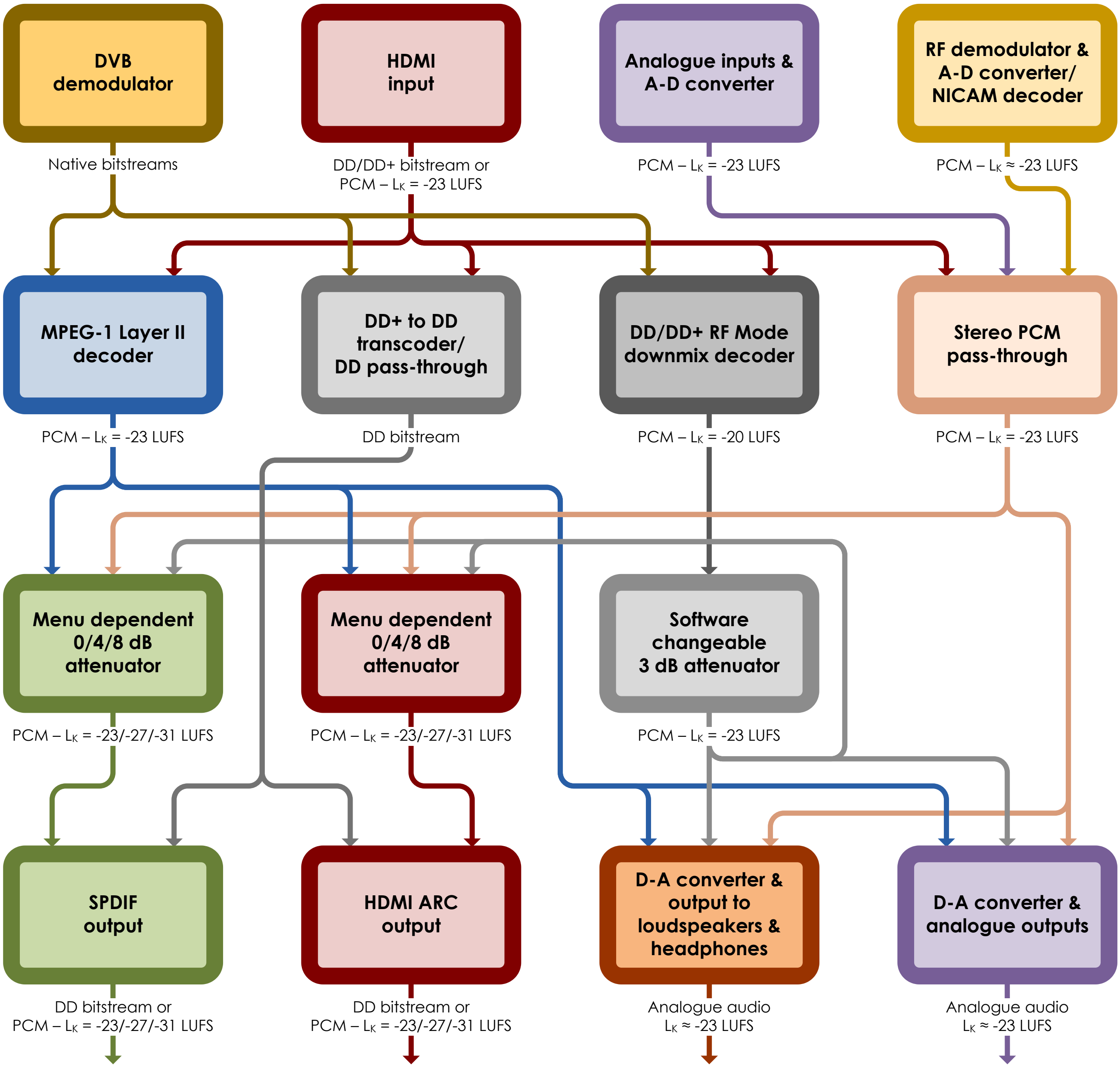


Figure 7.4.4.1: Audio processing within the System A IDTV

7.4.5 Graphical representation of the level adaptation in the System B IDTV

The following figure shows a graphical representation of the audio processing for a System B IDTV featuring HE-AAC. Its application to IRDs which operate in a different transmission system, which offer fewer or more options or which offer Audio Description by using two MPEG-1 Layer II and/or two HE-AAC decoders, can be derived from this figure.

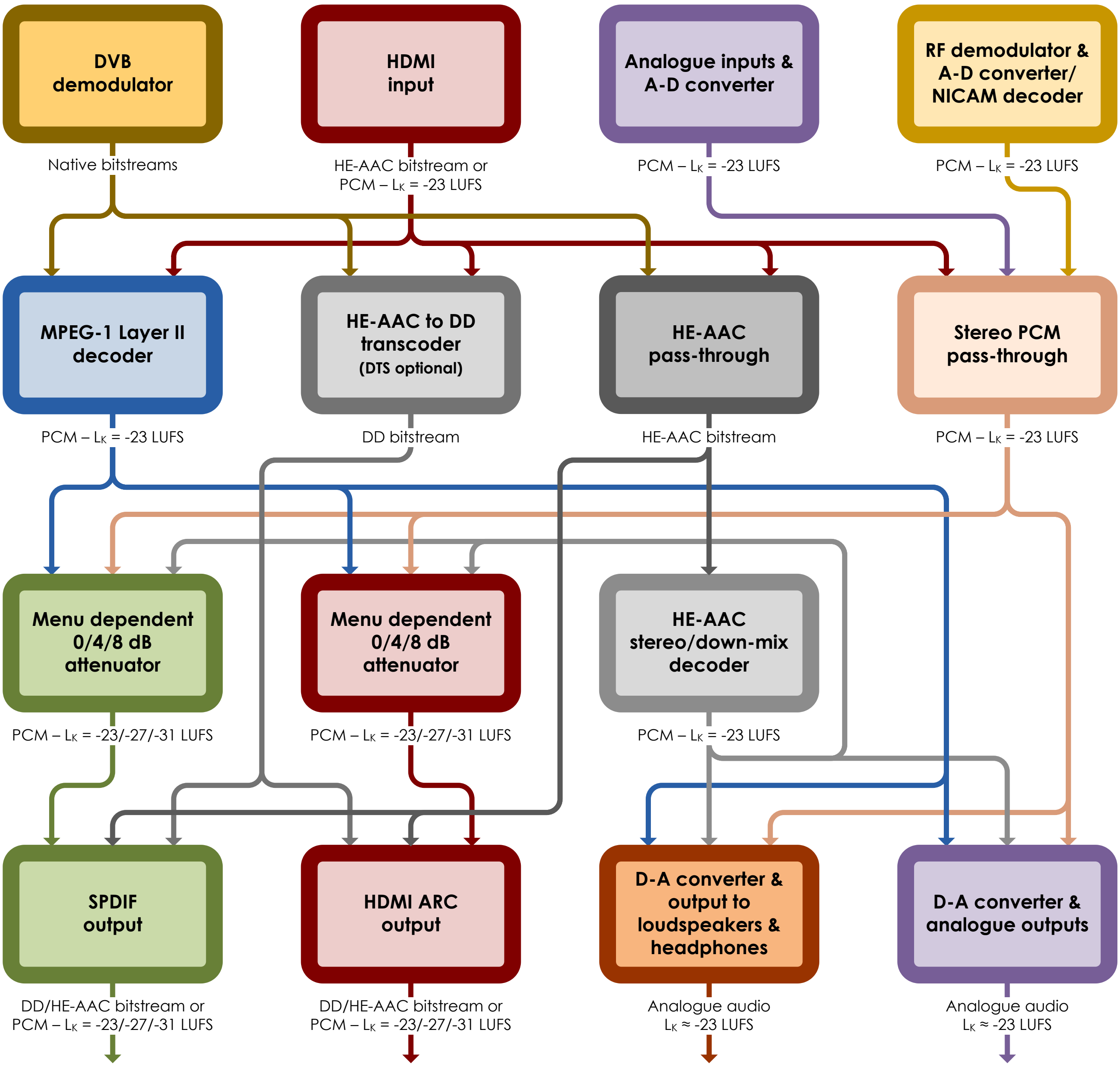


Figure 7.4.5.1: Audio processing within the System B IDTV

7.4.6 Overview of the level adaptation required for IDTV installation menu settings

Interface	Setting	PCM loudness level (LUFS)	PCM attenuation MPEG-1 Layer II decoder (dB)	PCM attenuation DD/DD+ decoder (dB)	PCM attenuation HE-AAC decoder (dB)	Codec bitstream support
SPDIF	HOME THEATRE	-31	8	11	8	Yes
SPDIF	HOME THEATRE (OFFSET)	-27	4	7	4	Yes
SPDIF	HOME THEATRE (HE-AAC) (1)	-31	8	11	8	Yes
SPDIF	STEREO EQUIPMENT	-23	0	3	0	No
HDMI ARC	HOME THEATRE	-31	8	11	8	Yes
HDMI ARC	HOME THEATRE (OFFSET)	-27	4	7	4	Yes
HDMI ARC	HOME THEATRE (HE-AAC) (1)	-31	8	11	8	Yes
HDMI ARC	STEREO EQUIPMENT (PCM)	-23	0	3	0	No

Note 1: Only relevant for System B IRDs.

For specific details, please see the notes in § 7.4.1. The relationship between discrete input and output levels can be found in § 5.

7.4.7 IDTV volume control

It is strongly recommended that the audio levels on the SPDIF and HDMI ARC be unaffected by use of the IDTV volume control. Alternatively the IDTV should use the remote-control code of the connected home theatre equipment or use the HDMI Consumer Electronics Control (CEC) feature. It must be emphasised that this concept optimises comfort, as the user is able to control the volume for all signal formats (e.g. PCM and passed-through codec bitstreams). It also avoids conflicts with the volume setting of the home theatre device as well as avoiding conflict with the alignment of other sources to that equipment. To use this functionality, the IDTV remote control must have an option to choose between the IDTV and a connected home theatre device. The CEC feature can handle this automatically. Audio mute functionality within the IDTV can be preserved. For an IDTV offering Audio Description, the volume control can still be used to adjust its headphone output, once the device detects that a headphone has been connected.

7.4.8 Analogue output level relationship

The output level on the SCART and RCA analogue interface shall be 2.0 V RMS using a sine wave at 1 kHz encoded at 0 dBTP (see *Note 1*). Accordingly, 0 dBTP corresponds to a signal peak level of 2.83 V. This specific output level is required in order to align with the analogue modulation levels specified in this document. This level alignment is compatible with CENELEC EN50049. A graphical representation of level relationships for several television systems is shown in § 5.

The following level alignment shall be applied to SCART/RCA interfaces:

Level alignment for the analogue and SCART input and output (1, 2, 3)	-12 dBTP using an 1 kHz sine wave results in an RMS signal level of 502 mV (±1 dB).

Note 1: True Peak Level is the maximum peak level of an audio signal measured with an oversampling True Peak Meter. If a True Peak meter is not available, a sine wave at 997 Hz, encoded at the specified level in dBFS, may be used for reference.

Note 2: For digital processing: To prevent clipping on the analogue inputs, an input attenuator of, for example, 6 dB can be applied followed by a digital gain shift (of 6 dB) to arrive at the same value.

Note 3: To reduce attenuation of the output level, it is recommended that the source impedance of an output interface is as low as possible, so long as the output remains unconditionally stable. CENELEC EN50049 specifies output impedance between 300 Ω and 1000 Ω for the SCART audio output interface. It is recommended that 300 Ω be applied to reduce variations in loudness level.

7.5 Audio preference settings

A service can supply more than one audio stream. In general, the user may or may not have a preference for DD/DD+ or HE-AAC encoded streams (if they are supplied with the service) instead of MPEG-1 Layer II. A general setting in the user preference menu assists the user in automatically choosing the preferred setting if a new service is added. It is strongly recommended to implement a service dependent setting to override the general setting. The following sections describe this in more detail.

7.5.1 Audio preference settings to be implemented in the user preference menu

Item	Choice	Description
AUDIO STREAM	MPEG-1 LAYER II *or* DOLBY DIGITAL *or* HE-AAC *or* AUTO	Defines the general preference of the user for the MPEG-1 Layer II, the DD/DD+ or the HE-AAC stream if it is supplied with the service. If set on AUTO, the IDTV follows the PSI/SI information.

7.5.2 Audio preference settings to be implemented in the service dependent user menu

Item	Choice	Description
AUDIO STREAM	MPEG-1 LAYER II *or* DOLBY DIGITAL *or* HE-AAC *or* AUTO	Defines the preference of the user for the MPEG-1 Layer II or the DD/DD+ or the HE-AAC stream if it is supplied with the service. If set on AUTO, the IDTV follows the setting in the user preference menu.

Note 1: This setting overrides the general preference as set in the installation menu and shall be stored in non-volatile memory so that the IDTV returns to the same setting after switching services and after power up. IDTVs featuring recording facilities shall store metadata together with the (transport stream) file to indicate and apply the preference.

7.6 Audio resolution

Processing with the IDTV should retain at least 24 bit resolution. Dithering shall be used when resolution is reduced.

7.7 DD/DD+ Dynamic Range Control

For System A IDTVs, the decoder shall follow the Dolby metadata. If, for example, the DD/DD+ encoder used the DRC=NONE profile, the decoder shall not apply any compression other than overload protection. The DD/DD+ decoder in the IDTV used for stereo reproduction shall apply the RF Mode dynamic range metadata.

7.8 HE-AAC Dynamic Range Control

For System B IDTVs, the decoder used for stereo reproduction shall follow the metadata in the *dynamic_range_info()* field of the ISO/IEC 14496-3 stream.

7.9 Additional Dynamic Range Control or audio-enrichment features

Additional proprietary dynamic range control and audio-enrichment functionality shall be optional. These features shall always be switched off by default. For specific listening conditions, for example late at night or in a bedroom, it is useful to implement an additional DRC application. This so-called 'Night Mode' shall not just be based on progressively scaled DD/DD+ DRC metadata, as this might not be active (for example if the DD/DD+ encoder applies the profile DRC=NONE) or not applicable (for example using MPEG-1 Layer II audio).

7.10 Down-mixing of multi-channel audio

Multi-channel broadcasts are often presented in the home on two loudspeakers. To achieve this, the (typically) five channels are combined into two by adding a certain amount of the surround channels' signal to the front channels and some of the centre channel's signal to left and right. The amounts may be controlled by down-mix coefficients transmitted with the audio signal. In some broadcast recommendations there is ambiguity regarding the need to scale the down-mix coefficients in order to avoid signal overload, should all channels contain high-level signals. To maintain consistent signal level between down-mixed multi-channel programmes and native stereo programmes, this scaling should not be applied. The broadcaster shall ensure that sufficient headroom and/or dynamic range control values are included in the transmission to prevent any overload when down-mixing.

System B IDTVs shall apply the down-mix parameters according to ETSI 101 154 Annex C 5.2.4, *down-mixing_levels_MPEG4* (the parameter with increased resolution over that in ISO/IEC 14496-3).

7.11 IDTV interactive applications

Interactive applications on an IDTV that make use of accompanying sound can be made consistent with EBU R 128 by normalising the audio in advance by, for example, an algorithm implemented in software. Care shall be taken within the design of the IDTV so that the signal alignment corresponds to that of the broadcasted audio via all audio interfaces with the aim of achieving an equal integrated loudness level.

7.12 Internet applications

For IDTVs featuring Internet and/or network access, loudness jumps can spoil the quality of experience as audio and video streams can be very loud. Although applications such as Internet access fall outside the scope of the current revision of this document, it is thought that it might be advantageous to include an attenuator for Internet and network stream decoder outputs that can be set by the user.

8. Home theatre devices

8.1 Application

The guidelines described in this section are applicable to home theatre devices, where the design allows the opportunity to change audio levels by means of a software update. An AV-receiver is also considered a home theatre device in this document.

8.2 Audio systems

In this section, home theatre devices are described that can have all or a part of the following audio system playback functionality:

- PCM
- MPEG-1 Layer II
- DD or DD+, including high definition variants
- DTS, including high definition variants
- HE-AAC

Codecs used for Internet, network stream-based and file-based applications are included as a suggestion only, as these applications are considered to be outside the scope of the current revision of this document. They may however be included in a future revision.

8.3 Line Mode and RF Mode

The terms ‘RF Mode’ and ‘Line Mode’ are described in Dolby Technical Bulletin 11 and in other Dolby guidelines. Line Mode, which is the default for home theatre devices, uses an internal loudness level equivalent to the Sound Reproduction Level of -31 LUFS.

8.4 Level adaptation

Home theatre devices shall have an internal digital audio loudness reference level equivalent to the Sound Reproduction Level of -31 LUFS, the internal loudness level used in DD/DD+ codecs. On default, home theatre equipment shall apply no attenuation to the PCM signal. This causes a PCM source normalised to -31 LUFS to be played back at the same loudness level. Other codecs that are referenced to -31 LUFS or that include a Decoder Target Level descriptor shall be set to -31 LUFS, thereby bringing them into alignment with the output of the DD/DD+ decoder. For devices featuring an output for headphones, the audio alignment shall be the same as for the pre-amplifier/ loudspeaker outputs.

An input-dependent configurable setting shall be implemented in the user menu to be able to control loudness differences of several sources.

Note 1: The menu structure for HDMI inputs and outputs could be replaced in the future by adaptation of the HDMI specification so that identification of connected equipment and the control of the corresponding loudness levels can be handled automatically.

8.4.1 Level adaptation settings for inputs to be implemented in the user menu

To reduce loudness jumps when switching between inputs, a source-dependent gain setting between 0 and -30 dB shall be implemented in the user menu. This input attenuator shall have a factory default setting such that no attenuation is applied. This default, standard or reference value corresponds to unity gain. The input attenuator shall be available to all input signals except for in the device integrated codecs with built in loudness normalisation system (e.g. Dialnorm in DD/DD+). For practical reasons, the input attenuator may be implemented as part of the volume control, functioning as an offset depending on the chosen source.

In the input attenuator menu settings, more than one source can be mapped to a given input. By doing that, the attenuation can be different if a particular input is used by, for example, a Blu-ray/DVD player playing PCM audio from a DVD or Blu-ray disk that is also used for CD playback. By selecting the relevant source and signal type, the attenuation is also changed to support the specific use.

The following table shows a general suggestion for the representation of the input attenuator in the user menu of a home theatre device with, in this example, nine sources:

SOURCE (menu choice)	**SOURCE NAME (free text from user)**	**INTERFACE (menu choice)**	**GAIN (user setting in dB)**
1	BLU-RAY/DVD	HDMI 1	0
2	CD	HDMI 1	-21
3	SET-TOP BOX	HDMI 2	0
4	TELEVISION	SPDIF 2	0
5	TELEVISION	HDMI 1 ARC	0
6	MEDIA PLAYER	SPDIF 3	-8
7	DAB RECEIVER	SPDIF 1	-8
8	FM CABLE RECEIVER	Analogue 2	-8
9	FM TERRESTRIAL RECEIVER	Analogue 3	-12

Note 1: For analogue inputs, the range may also include a limited positive gain range (for example up to +6 dB) for compatibility with specific or older consumer equipment.

Note 2: When connecting an EBU Tech 3344 compliant television set or an IRD, the user should leave the input setting on default (0 dB attenuation) for loudness consistency. When connecting a DAB tuner or an FM radio cable tuner receiving signals from a distribution network compliant to EBU R 128, the user is recommended to set the attenuator of the home theatre device of that specific input to -8 dB for loudness consistency.

The suggested user setting of -21 dB recommended for CD playback is based on the use of a media player which is not compliant to EBU Tech 3344 and an average loudness levels of CDs of -10 LUFS. Related to the internal loudness reference level of -31 LUFS of home theatre equipment it avoids the risks of undergoing excessive loudness jumps. Loudness levels of CDs are very different; a range from -20 up to -5 LUFS or even wider can occur [#2]. As a result of that, one value for CD input that fits all disks can unfortunately not be given.

A suggested setting for FM terrestrial radio is -12 dB. Currently, loudness levels of terrestrial FM radio services can be very different, depending on local rules. So here *too, one value for FM terrestrial radio input that fits all services cannot be given.*

8.4.2 Level adaptation settings for applications to be implemented in the user menu

To reduce loudness jumps when switching between applications, a source-dependent gain setting between 0 and -30 dB shall be implemented in the user menu following a similar approach to that in § 8.4.1.

For home theatre devices featuring Internet and/or network access, loudness jumps can spoil the quality of experience as audio and video streams can be very loud. Although applications like Internet access fall outside the scope of the current revision of this document, it is thought that it might be advantageous to include a user-adjustable attenuator for Internet and network stream decoder outputs as well.

The following table shows a general suggestion for the representation of the input attenuator in the user menu of home theatre equipment with, in this example, two applications:

SOURCE (menu choice)	**APPLICATION NAME (menu item)**	**SIGNAL TYPE (menu item)**	**GAIN (user setting in dB)**
1	HOME THEATRE	DTS CORE	-4
2	FM RADIO RECEIVER	FM	-8

Note 1: The suggested user setting of -4 dB for the DTS Core codec is based on the assumption that audio of traditional DTS Digital Surround programme material has been normalised to a loudness level of -27 LUFS. This setting shall not influence other DTS codecs that make use of loudness metadata.

Note 2: The application 'FM radio receiver' is only available if the receiver is built in the device. § 5 and § 10 can be checked for the mapping of the internal signal level of the FM radio receiver. For externally connected FM radio receivers, see § 8.4.1. Use of the same mapping makes it possible to compare the level adaptation of an internal FM Radio receiver with an optional external FM Radio receiver. When receiving signals from a distribution network compliant to EBU R 128, the user is recommended to set the attenuator of the home theatre device of that specific input to -8 dB for loudness consistency. A suggested setting for FM terrestrial radio is -12 dB.

8.4.3 Audio on the HDMI output

To simplify the presentation of consistent loudness levels on its HDMI outputs, a home theatre device must employ level adaptation. More advanced devices shall include a decoder for DD/DD+ and/or HE-AAC which applies a PCM loudness level of -23 LUFS for output to stereo analogue RCA ports or stereo PCM to SPDIF or HDMI ports (see section 8.4.4). Signals from other sources shall have a gain of +8dB applied to the stereo outputs, as shown in figure 8.4.6.2. This gain shall be programmable to allow for potential future changes. Distortion due to numerical overload can be prevented by, for example, including a digital limiter after the stereo down-mixer, before the +8 dB gain, as shown in the figure. To avoid loudness inconsistencies when delivering audio to IDTVs which are not compliant with this Tech Doc, the home theatre device should output only PCM to the HDMI port, even where the E-EDID query identifies the IDTV as supporting compressed audio. The approach described in this section supports the use of HDMI CEC to control playback using either the IDTV or home theatre system.

Note 1: Handling of switchable playback could in the future be implemented by adaptation of the HDMI specification so that identification of connected equipment and the control of the corresponding loudness levels can be handled automatically. By use of this identification, control data could, for example, be sent from the television set through the home theatre device back to the source device (e.g. an IRD or media player) if the user prefers to hear the sound via the (internal) loudspeakers of the television set instead of those of the home theatre device. The source device could subsequently apply the correct loudness level and appropriate dynamic range control, if applicable.

8.4.4 Additional information to implement the adaptation in home theatre devices

The following information clarifies how level adaptation shall be implemented for specific codecs:

- **DD/DD+ processing**

  Home theatre devices shall include a DD/DD+ decoder in Line Mode operation applying a PCM loudness level of -31 LUFS.

  On the multi-channel analogue RCA outputs, a PCM equivalent loudness level of -31 LUFS shall be applied by using the main decoder. If the device supports output signals of the decoded bitstreams to stereo analogue RCA outputs and/or a stereo PCM signal on the SPDIF or HDMI output, an additional DD/DD+ decoder in RF Mode operation, which applies a PCM loudness level of -23 LUFS, shall be included.

- **HE-AAC processing**

  HE-AAC decoding for home theatre purposes shall apply the Programme Reference Level descriptor (*prog_ref_level*, specified in ISO/IEC 14496-3) of the HE-AAC bitstream and shall apply the Decoder Target Level descriptor (*target_level*, specified in ISO/IEC 144496-3) at a level of -31 LUFS.

  On the multi-channel analogue RCA outputs, a PCM equivalent loudness level of -31 LUFS shall be applied by using the main decoder. If the equipment supports output signals of the decoded bitstreams to stereo analogue RCA outputs and/or a stereo PCM signal on the SPDIF or HDMI output, an additional HE-AAC decoder shall be included. This decoder shall use a PCM loudness level equivalent to the Sound Reproduction Level of -23 LUFS by applying the Programme Reference Level descriptor (*prog_ref_level*, specified in ISO/IEC 14496-3) of the HE-AAC bitstream and the Decoder Target Level descriptor (*target_level*, specified in ISO/IEC 144496-3) at a level of -23 LUFS.

  If an HE-AAC bitstream does not contain loudness metadata, the home theatre device shall follow the MPEG-4 standard by assuming that the audio is already at the EBU R 128 Target Level.

Note 1: Manufacturers wishing to use the "Dolby Pulse" implementation of HE-AAC should consult Dolby (in particular Technical Bulletin 11) for information about the additional steps needed to meet the requirements of this Tech Doc.

8.4.5 Notes about the graphical representation of the audio processing within devices

The figures in the following sections show a graphical representation of the audio processing within the device. The following notes are relevant for these figures:

Note 1: The term 'L_K' refers to loudness. On analogue outputs, the term '$L_K \approx$' refers to the loudness of the decoded PCM signal based on the mapping specified in this document *between the levels in the analogue and digital domain.*

Note 2: The device can have more or fewer input and output interfaces and more or fewer features, depending on model and application.

8.4.6 Graphical representation of the level adaptation in home theatre equipment

The following figures show a graphical representation of the audio processing in home theatre equipment. Figure 8.4.6.1 shows the base processing. Figure 8.4.6.2 shows the processing for devices which support which support pass-through to stereo analogue RCA outputs and/or a stereo PCM signal on the SPDIF or HDMI output (see § 8.4.3 for details). Its application to devices which offer fewer or more options can be derived from this figure. Processing to playback from files, a network or from the Internet is included as a suggestion only.

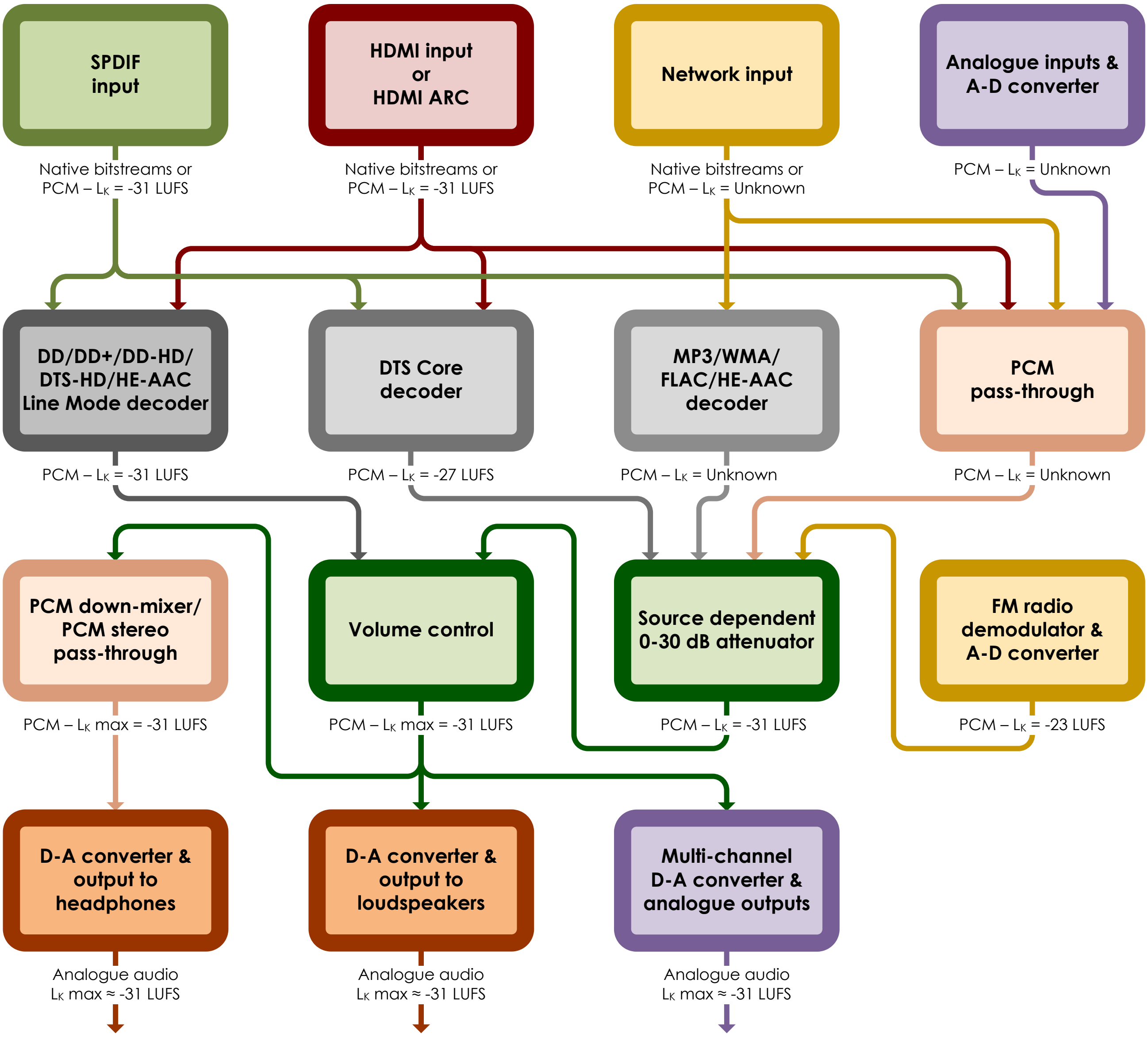


Figure 8.4.6.1: Base audio processing within home theatre equipment

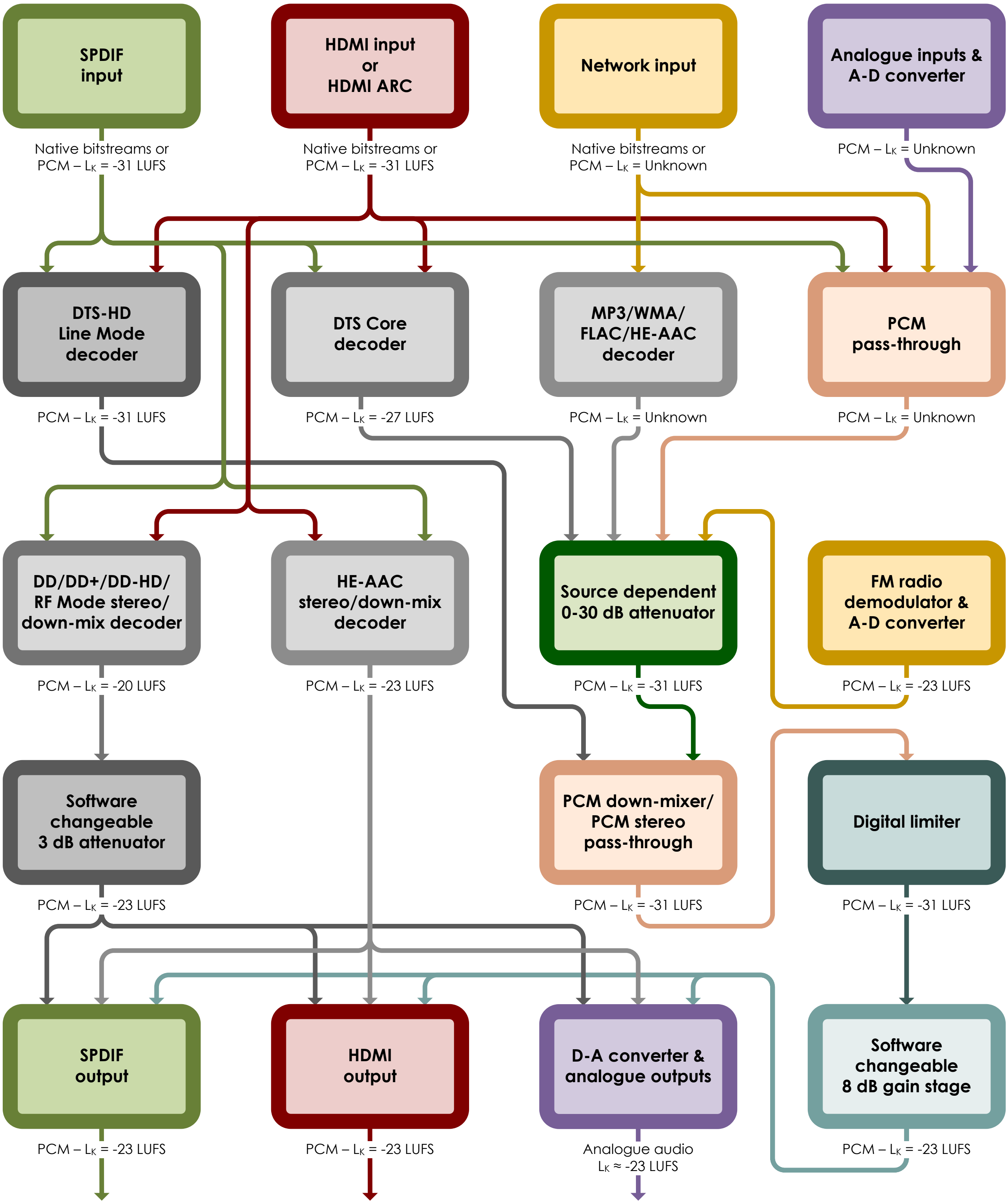


Figure 8.4.6.2: Additional processing to support pass-through to stereo outputs within home theatre equipment (extension to figure 8.4.6.1)

8.5 Audio resolution

Processing with the home theatre device should retain at least 24 bit resolution. Dithering shall be used when resolution is reduced.

8.6 DD/DD+ Dynamic Range Control

The decoder shall follow the Dolby metadata. If, for example, the DD/DD+ encoder used the DRC=NONE profile, the decoder shall not apply any compression other than overload protection.

Home theatre devices shall apply the Line Mode dynamic range metadata by default. DRC settings that apply scaling of the gain reduction are optional. It shall always be possible for the user to switch off DRC and the device shall store this setting in non-volatile memory, so that it returns to the same preference after switching sources and after power up.

8.7 HE-AAC Dynamic Range Control

It shall always be possible for the user to switch off DRC and the device shall store this setting in non-volatile memory, so that it returns to the same preference after switching sources and after power up.

8.8 Additional Dynamic Range Control or audio-enrichment features

Additional proprietary dynamic range control and audio-enrichment functionality shall be optional. These features shall always be switched off by default. For specific listening conditions, for example late at night or in a bedroom, it is useful to implement an additional DRC application. This so-called 'Night Mode' shall not just be based on progressively scaled DD/DD+ DRC metadata, as this might not be active (for example if the DD/DD+ encoder applies the profile DRC=NONE) or not applicable (for example using PCM audio).

8.9 Down-mixing of multi-channel audio

Multi-channel broadcasts are often presented in the home on two loudspeakers. To achieve this, the (typically) five channels are combined into two by adding a certain amount of the surround channels' signal to the front channels and some of the centre channel's signal to left and right. The amounts may be controlled by down-mix coefficients transmitted with the audio signal. In some broadcast recommendations there is ambiguity regarding the need to scale the down-mix coefficients in order to avoid signal overload, should all channels contain high-level signals. To maintain consistent signal level between down-mixed multi-channel programmes and native stereo programmes, this scaling should not be applied. The broadcaster shall ensure that sufficient headroom and/or dynamic range control values are included in the transmission to prevent any overload when down-mixing.

System B home theatre devices shall apply the down-mix parameters according to ETSI 101 154 Annex C 5.2.4, *down-mixing_levels_MPEG4* (the parameter with increased resolution over that in ISO/IEC 14496-3).

9. Media players

9.1 Application

The guidelines described in this section are applicable to DVD, Blu-ray and other media players, where the design allows the opportunity to change audio levels by means of a software update.

9.2 Audio systems

In this section two transmission variants are distinguished. The audio signal could be carried by System A or B, as determined for the relevant network or media. A media player can feature either one of the following variants, or both:

- System A featuring MPEG-1 Layer II and Dolby Digital (DD) or Dolby Digital Plus (DD+)
- System B featuring MPEG-1 Layer II and HE-AAC, optionally transcoded to Dolby Digital (DD) or DTS

This section describes devices that can also have all or a part of the following audio system playback functionality:

- PCM
- High definition variants of DD

9.3 Line Mode and RF Mode

The terms ‘RF Mode’ and ‘Line Mode’ are described in Dolby Technical Bulletin 11 and in other Dolby guidelines. Line Mode uses an internal loudness level equivalent to the Sound Reproduction Level of -31 LUFS. In DD/DD+ decoders operating in RF Mode this level is raised to -20 LUFS with a compressed dynamic range, which is meant to be more compatible with signal levels used in analogue transmission. To be compliant with the EBU R 128 Target Level, this document specifies that the loudness level of the decoder operating in RF Mode be decreased to -23 LUFS by use of a software-adjustable attenuator of 3 dB.

9.4 Level adaptation

An output-dependent configurable setting shall be implemented in the user menu that switches the PCM loudness level by an amount depending on the connected equipment. Basically, the user chooses during installation what kind of equipment is connected to the SPDIF and the HDMI. Subsequently, the media player applies the correct level adjustment. It is suggested that a ‘wizard’ procedure be included to assist the user by displaying generic images of the connected equipment. For devices featuring an output for headphones, the audio alignment shall be the same as for the analogue line outputs. For remarks about the media player volume control, see § 9.4.7.

Note 1: The menu structure for HDMI outputs could be replaced in the future by adaptation of the HDMI specification so that identification of connected equipment and the control of the corresponding loudness levels can be handled automatically.

9.4.1 Level adaptation settings to be implemented in the installation menu

The following base menu settings are recommended for media players featuring an HDMI output:

Item	Choice	Description
HDMI DEVICE	TELEVISION	This setting is intended if a television set is directly connected to the media player using HDMI. It is backwards compatible with television sets which are not compliant with EBU Tech 3344 and it is the recommended setting for the installed base as well as new devices not feeding a home theatre device. The PCM loudness reference level is -23 LUFS. The media player shall output only PCM signals to the HDMI, even where the E-EDID query identifies the sink as supporting compressed audio.
HDMI DEVICE	TELEVISION → HOME THEATRE	This setting is intended for a connected television set that is compliant with EBU Tech 3344. Subsequently, a home theatre device can be connected to the television set if it features an SPDIF output or HDMI ARC. The PCM loudness reference level is -23 LUFS. Both PCM signals as well as codec bitstreams are supported, except where the E-EDID query identifies the sink as supporting basic audio only. The connected television set needs to decrease the PCM levels on its SPDIF and HDMI ARC output to correctly pass-through audio to home theatre equipment (see § 7 for details).
HDMI DEVICE	HOME THEATRE → TELEVISION	This setting is intended if a home theatre device is directly connected to the media player using HDMI. Subsequently, the television set can be connected to the home theatre device if both feature HDMI. To avoid loudness differences, the PCM loudness reference level is brought in line with the Sound Reproduction Level of -31 LUFS, the internal loudness level used in DD/DD+ codecs. Both PCM signals as well as codec bitstreams are supported, except where the E-EDID query identifies the sink as supporting basic audio only.
HDMI DEVICE	HOME THEATRE (OFFSET MODE) → TELEVISION	This is a similar application to the previous one, but now intended for a home theatre device that uses a PCM loudness reference level of -27 LUFS (see *Note 1*).
HDMI DEVICE	NONE	This setting can be used if no device is connected to HDMI. It is included to support the user with a complete set of choices and is a logical setting if, for example, the television set is connected to the SCART interface. The audio output signal on the HDMI shall be muted in this mode to support the effect of this choice.

The following additional menu settings are recommended for media players featuring an HDMI output and specific functionality:

HDMI DEVICE	HOME THEATRE (PCM MCA MODE) → TELEVISION	This setting is only relevant if the media player features an additional multi-channel decoder (see § 9.4.2 for details). This setting is intended if a home theatre device is directly connected to the media player using HDMI. Subsequently, the television set can be connected to the home theatre device if both feature HDMI. To avoid loudness differences, the PCM loudness reference level is brought in line with the Sound Reproduction Level of -31 LUFS; the internal loudness level used in DD/DD+ codecs. The media player shall only output PCM signals to the HDMI, even if the E-EDID query identifies the sink as supporting compressed audio.

The following base menu settings are recommended for media players featuring an SPDIF output:

Item	**Choice**	**Description**
SPDIF DEVICE	HOME THEATRE	This setting is intended if a home theatre device is directly connected to the media player using SPDIF. To avoid loudness differences, the PCM Loudness reference level is brought in line with the Sound Reproduction Level of -31 LUFS, the internal loudness level used in DD/DD+ codecs. Both PCM signals as well as codec bitstreams are supported.
SPDIF DEVICE	HOME THEATRE (OFFSET MODE)	This is a similar application to the previous one, but now intended for a home theatre device that applies a PCM loudness reference level of -27 LUFS (see *Note 1*).
SPDIF DEVICE	STEREO EQUIPMENT (PCM)	This setting is intended if a PCM stereo device such as an amplifier or a recording device is directly connected to the media player using SPDIF. The PCM loudness reference level is -23 LUFS. The media player shall output only PCM signals to the SPDIF in this mode.
SPDIF DEVICE	NONE	This setting can be used if no device is connected to the SPDIF. This setting is included to support the user with a complete set of choices. The output signal on the SPDIF shall be muted in this mode to support the effect of this choice.

The following additional menu setting is recommended for media players featuring an SPDIF output and specific functionality:

SPDIF DEVICE	HOME THEATRE (HE-AAC MODE)	Only relevant for System B media players. This setting enables that HE-AAC codec bitstreams are output instead of transcoded DD bitstreams for home theatre equipment that supports HE-AAC decoding.

Note 1: It appears that a substantial number of home theatre devices process PCM input signals with a fixed 4 dB offset compared to the output of the DD/DD+ decoder. Among, but not limited to, them is equipment certified to THX specifications. This state of affairs applies to old and current designs. This offset for PCM input signals is considered to be undesirable. It is hoped that this document can be taken into account for future

specifications for home theatre equipment, which means that this fixed gain offset for PCM signals shall not be present. Nevertheless, to achieve loudness consistency with the major part of home theatre devices, an alternative offset mode is included in this document based on the use of this paradigm.

Note 2: If it cannot be avoided, menu entries may be made depending on each other to reduce internal complexity. SPDIF modes may be limited to NONE if the user has chosen one of the following settings:

HDMI DEVICE = TELEVISION → HOME THEATRE

HDMI DEVICE = TELEVISION → HOME THEATRE (OFFSET MODE)

HDMI DEVICE = HOME THEATRE → TELEVISION

HDMI DEVICE = HOME THEATRE (OFFSET MODE) → TELEVISION

HDMI DEVICE = HOME THEATRE → TELEVISION (PCM MCA MODE)

It is recommended to restore the last applied choice for SPDIF DEVICE once the user chooses HDMI DEVICE = TELEVISION or HDMI DEVICE = NONE.

Note 3: On modes that support codec bitstreams, the internal DD/DD+ or HE-AAC decoder shall not be used to feed the SPDIF and HDMI, unless a specific application is being used like Audio Description which requires it, or where the E-EDID query identifies the sink as supporting basic audio only.

Note 4: In case the HDMI E-EDID query identifies the sink as supporting basic audio only, the media player shall block codec bitstreams, but shall continue to use the same PCM attenuation for that setting. This shall be done to avoid wrong audio levels after a failure reading the E-EDID.

Note 5: If the E-EDID query identifies the sink as supporting HE-AAC audio, HE-AAC codec bitstreams are output instead of transcoded DD bitstreams (only relevant for the HDMI output on System B media players).

Note 6: It is recommended not to force the user choice for the HDMI setting based on the E-EDID query, as in practice errors can occur during the HDMI handshake procedure, which could result in a wrong choice and correspondingly result in loudness jumps.

Note 7: It is recommended to program HDMI DEVICE = TELEVISION and SPDIF DEVICE = HOME THEATRE as a factory default.

Note 8: The traditionally applied user setting to control the preference for the use of the internal DD/DD+ or HE-AAC decoder or outputting codec bitstreams on the HDMI and SPDIF is obsolete using the new paradigm described in this document. This choice is fully integrated in the settings described in this section.

9.4.2 Additional information to implement the adaption in media players

The media player shall adjust the output level of all built-in audio decoders according to the figures in § 9.4.4 and § 9.4.5 so that the perceived programme loudness is consistent for all audio-coding schemes. The following information clarifies how level adaptation shall be implemented:

- **MPEG-1 Layer II processing (System A and B)**

  The media player shall include a PCM level attenuator to reduce the level of decoded MPEG-1 Layer II audio at the SPDIF and HDMI for modes indicated in § 9.4.1 intended for the use of home theatre equipment. The gain reduction steps (0, -4 and -8 dB) shall be programmable to allow for a potential future change. This facility may be provided by a software update. Gain reduction shall not be applied to the stereo analogue outputs. It shall not be applied either to the HDMI or SPDIF for modes where PCM attenuation is indicated as 0 dB in § 9.4.6.

- **DD/DD+ processing (System A)**

  The media player featuring System A shall include a PCM level attenuator to reduce the level of decoded DD/DD+ audio in RF Mode operation to accommodate reproduction in line with the target loudness level of -23 LUFS, which means an attenuation of 3 dB. This gain reduction shall be programmable to allow for a potential future change. This facility may be provided by a software update. From that point, for modes and applications where use of the internal decoder is required to supply the SPDIF and/or HDMI and for the analogue outputs, the media player shall apply the same procedure for decoded DD/DD+ signals as for MPEG-1 Layer II.

  On the SCART and stereo analogue RCA outputs, a PCM loudness level of -23 LUFS shall be applied by using the main decoder. On media players featuring multi-channel PCM or analogue outputs, the device shall include an additional DD/DD+ decoder in Line Mode operation applying a PCM equivalent loudness level of -31 LUFS. This multi-channel decoder shall have a user switchable down-mix mode to enable a default stereo reproduction on the left and right loudspeaker of the multi-channel sound system (see § 9.10 about applying the correct loudness level when down-mixing).

  If DD/DD+ codec bitstreams are passed through to the SPDIF or HDMI, the media player shall neither change the audio content nor the accompanying metadata.

- **HE-AAC processing (System B)**

  The media player featuring System B shall apply a PCM loudness reference level equivalent to the EBU R 128 Target Level. This shall be achieved by applying the Programme Reference Level descriptor (*prog_ref_level*, specified in ISO/IEC 14496-3) of the HE-AAC bitstream and the Decoder Target Level descriptor (*target_level*, specified in ISO/IEC 144496-3) at a level of -23 LUFS. From that point, for the analogue outputs and for modes and application where use of the internal decoder is required, the media player shall apply the same procedure for decoded HE-AAC signals as for MPEG-1 Layer II.

  On the SCART and stereo analogue RCA outputs, a PCM equivalent loudness level of -23 LUFS shall be applied by using the main decoder. On media players featuring multi-channel PCM or analogue outputs, the media player shall use an additional HE-AAC decoder. This decoder shall use a PCM loudness level equivalent to the Sound Reproduction Level of -31 LUFS by applying the Programme Reference Level descriptor (*prog_ref_level*, specified in ISO/IEC 14496-3) of the HE-AAC bitstream and the Decoder Target Level descriptor (*target_level*, specified in ISO/IEC 144496-3) at a level of -31 LUFS. This multi-channel decoder shall have a user switchable down-mix mode to enable a default stereo reproduction on the left and right loudspeaker of the multi-channel sound system (see § 9.10 about applying the correct loudness level when down-mixing).

  If an HE-AAC bitstream does not contain loudness metadata, the media player shall follow the MPEG-4 standard by assuming that the audio is already at the EBU R 128 Target Level.

  If HE-AAC codec bitstreams are passed through to the SPDIF or HDMI, the media player shall neither change the audio content nor the accompanying metadata.

  In case of transcoding from HE-AAC to DD, the media player shall preserve the level of the audio and transcode the accompanying metadata to ensure correct reproduction level in a downstream decoder. If the HE-AAC stream does not contain loudness metadata, the media player shall signal a Dialnorm of -23 in the DD bitstream assuming a Programme Reference Level of the incoming audio of -23 LUFS.

Note 1: Manufacturers wishing to use the "Dolby Pulse" implementation of HE-AAC should consult Dolby (in particular, Technical Bulletin 11) for information about the additional steps needed to meet the requirements of this Tech Doc.

9.4.3 Notes about the graphical representation of the audio processing within devices

The figures in the following sections show a graphical representation of the audio processing within the device. The following notes are relevant for these figures:

Note 1: The term 'L_K' refers to loudness. On analogue outputs, the term '$L_K \approx$' refers to the loudness of the decoded PCM signal based on the mapping specified in this document between the levels in the analogue and digital domain.

Note 2: The device can have more or fewer input and output interfaces and more or fewer features, depending on model and application.

9.4.4 Graphical representation of the level adaptation in the System A media player

The following figure shows a graphical representation of the audio processing for a System A media player featuring DD/DD+. Its application to media players which offer fewer or more options can be derived from this figure. For specific requirements regarding portable and battery-powered devices, see § 2.7.

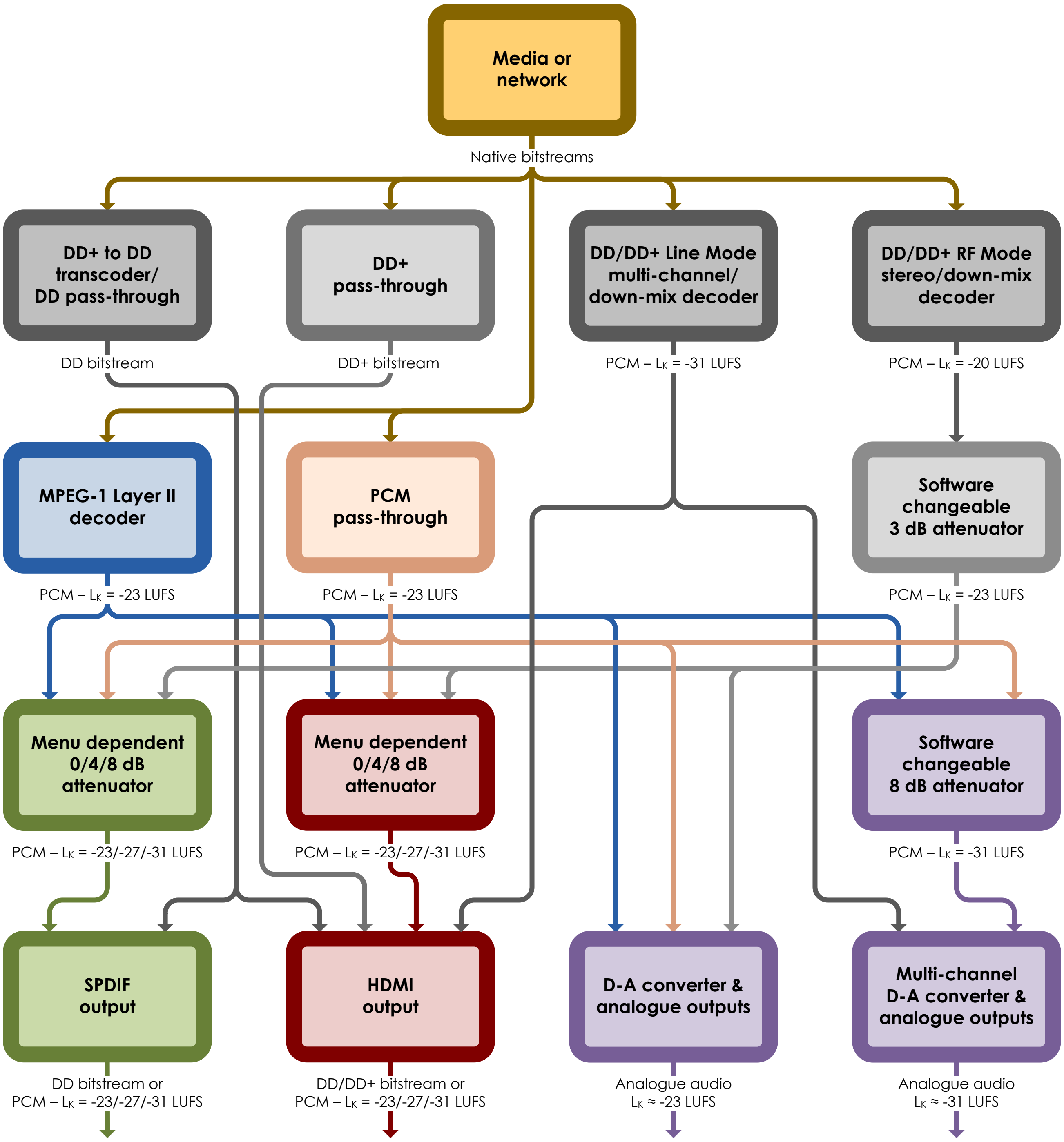


Figure 9.4.4.1: Audio processing within the System A media player

9.4.5 Graphical representation of the level adaptation in the System B media player

The following figure shows a graphical representation of the audio processing for a System B media player featuring HE-AAC. Its application to media players which offer fewer or more options can be derived from this figure. For specific requirements regarding portable and battery-powered devices, see § 2.7.

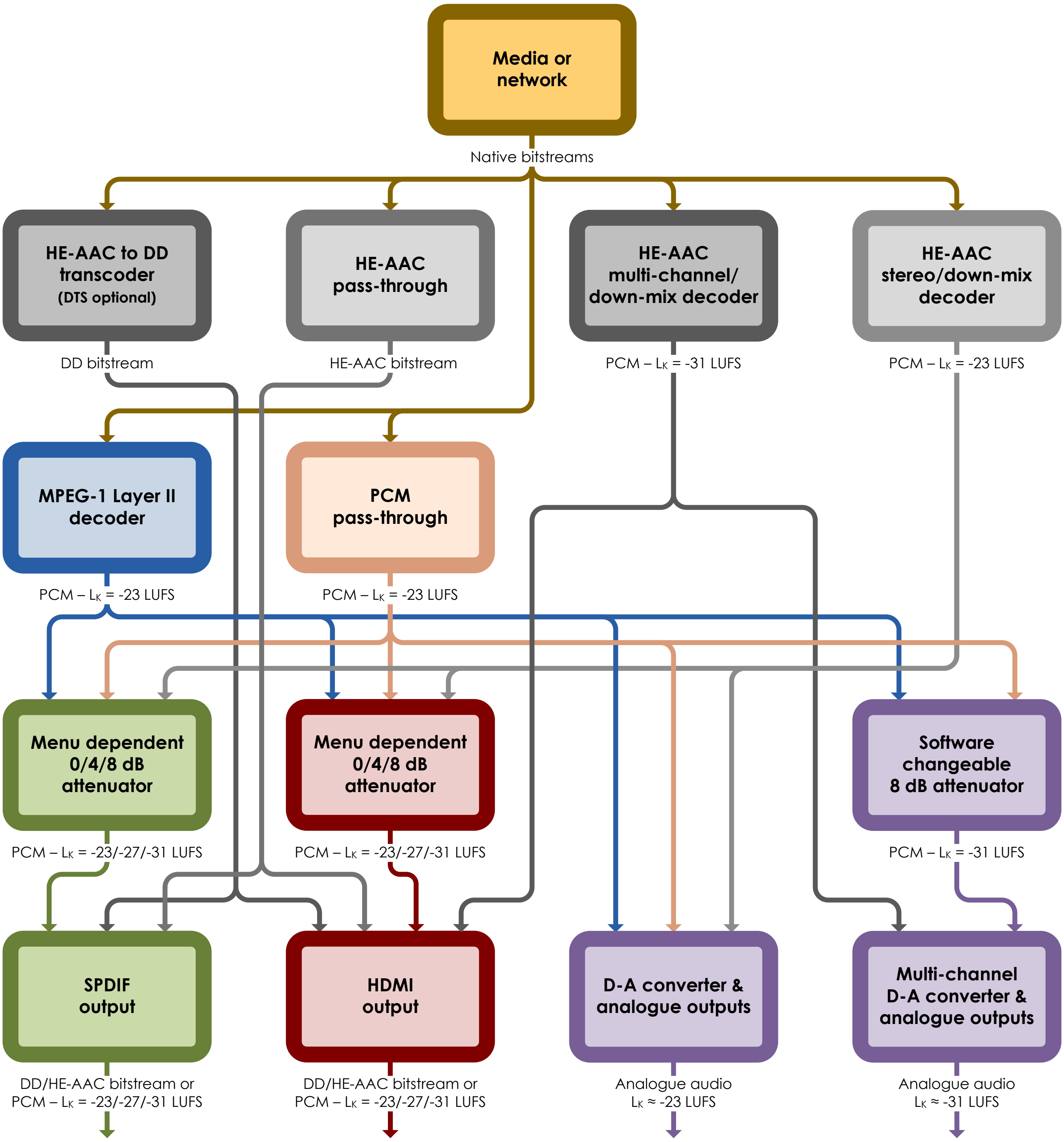


Figure 9.4.5.1: Audio processing within the System B media player

9.4.6 Overview of the level adaptation required for media player installation menu settings

Interface	Setting	PCM loudness level (LUFS)	PCM attenuation MPEG-1 Layer II decoder (dB)	PCM attenuation DD/DD+ decoder (dB)	PCM attenuation HE-AAC decoder (dB)	Codec bitstream support
HDMI	TELEVISION	-23	0	3	0	No
HDMI	TELEVISION → HOME THEATRE	-23	0	3	0	Yes
HDMI	HOME THEATRE → TELEVISION	-31	8	11	8	Yes
HDMI	HOME THEATRE (OFFSET) → TELEVISION	-27	4	7	4	Yes
HDMI	HOME THEATRE (PCM MCA) → TELEVISION (1)	-31	8	11	8	No
SPDIF	HOME THEATRE	-31	8	11	8	Yes
SPDIF	HOME THEATRE (OFFSET)	-27	4	7	4	Yes
SPDIF	HOME THEATRE (HE-AAC) (2)	-31	8	11	8	Yes
SPDIF	STEREO EQUIPMENT (PCM)	-23	0	3	0	No

Note 1: Only relevant for media players featuring a multi-channel decoder.

Note 2: Only relevant for System B media players.

For specific details, please see the notes in § 9.4.1. The relationship between discrete input and output levels can be found in § 5.

9.4.7 Media player volume control

It is strongly recommended that the audio levels inside the media player be unaffected by use of its volume control. Instead, the volume control of the media player should preferably use the remote control code of other equipment (e.g. television set and/or home theatre equipment) or use the HDMI Consumer Electronics Control (CEC) feature. It must be emphasised that this concept optimises comfort, as the user is able to control the volume for all signal formats (e.g. PCM and passed-through codec bitstreams). It also avoids conflicts between the volume setting of the IRD and the volume settings of both the television set and home theatre equipment as well as avoiding conflict with the alignment of other sources to that equipment. To use this functionality, the media player remote control must have an option to choose between the media player and a connected home theatre device. The CEC feature can handle this automatically. Audio mute functionality within the media player can be preserved. For a media player offering Audio Description, the volume control can still be used to adjust its headphone output, once the device detects that a headphone has been connected.

Note 1: For (older) media player models that do not have the ability to use the remote control code or HDMI CEC, it is recommended only to apply the volume control if the setting TELEVISION is chosen for the HDMI output, in order to be able to offer comfort to the user by another means. This reduces the negative impact of changing the levels inside the media player. The PCM levels on the SPDIF and on the HDMI using other settings than TELEVISION shall be left unaffected.

9.4.8 Analogue output level relationship

The output level on the SCART and RCA analogue interface shall be 2.0 V RMS using a sine wave at 1 kHz encoded at 0 dBTP (see *Note 1*). Accordingly, 0 dBTP corresponds to a signal peak level of 2.83 V. This specific output level is required in order to align with the analogue modulation levels specified in this document. This level alignment is compatible with CENELEC EN50049. A graphical representation of level relationships for several television systems and for FM radio is shown in § 5.

The following level alignment shall be applied to SCART/RCA interfaces:

Level alignment for the analogue and SCART input & output (1, 2, 3)	-12 dBTP using a 1 kHz sine wave results in an RMS signal level of 502 mV (±1 dB).

The following level alignment shall be applied to (balanced) XLR interfaces (or similar alternative) on a professional media player:

Level alignment for the XLR analogue input & output (1, 3)	-12 dBTP using a 1 kHz sine wave results in an RMS signal level of +6 dBu (±1 dB), if a normalisation factor of 0 dBrs is applied.
	-12 dBTP using a 1 kHz sine wave results in an RMS signal level of +3 dBu (±1 dB), if a normalisation factor of -3 dBrs is applied. The term dBrs is specified in ITU-R BS.645.

Note 1: True Peak Level is the maximum peak level of an audio signal measured with an oversampling True Peak Meter. If a True Peak meter is not available, a sine wave at 997 Hz, encoded at the specified level in dBFS, may be used for reference.

Note 2: For digital processing: To prevent clipping on the analogue inputs, an input attenuator of, for example, 6 dB can be applied followed by a digital gain shift (of 6 dB) to arrive at the same value.

Note 3: To reduce attenuation of the output level, it is recommended that the source impedance of an output interface is as low as possible, so long as the output remains unconditionally stable. CENELEC EN50049 specifies output impedance between 300 Ω and 1000 Ω for the SCART audio output interface. It is recommended that 300 Ω be applied to reduce variations in loudness level.

9.5 Audio preference settings

A service or media can supply more than one audio stream. In general, the user may or may not have a preference for DD/DD+ or HE-AAC encoded streams if supplied with the service or media instead of PCM or MPEG-1 Layer II. A general setting in the user preference menu assists the user in automatically choosing the preferred setting. It is strongly recommended to implement a service or media dependent setting to override the general setting. The following sections describe this in more detail.

9.5.1 Audio preference settings to be implemented in the user preference menu

Item	**Choice**	**Description**
AUDIO STREAM	PCM *or* MPEG-1 LAYER II *or* DOLBY DIGITAL *or* HE-AAC *or* AUTO	Defines the general preference of the user for the PCM, the MPEG-1 Layer II, the DD/DD+ or the HE-AAC stream if it is supplied with the service. If set on AUTO, the media player follows the PSI/SI or media information.

9.5.2 Audio preference settings to be implemented in the service or media dependent user menu

Item	Choice	Description
AUDIO STREAM	PCM *or* MPEG-1 LAYER II *or* DOLBY DIGITAL *or* HE-AAC *or* AUTO	Defines the preference of the user for the PCM, the MPEG-1 Layer II, the DD/DD+ or the HE-AAC stream if it is supplied with the service or media. If set on AUTO, the media player follows the setting in the user preference menu.

Note 1: The setting overrides the general preference as set in the installation menu and shall be stored in non-volatile memory so that the media player returns to the same preference after switching services. Media players featuring recording facilities shall store metadata together with the (transport stream) file to indicate the preference.

9.6 Audio resolution

Processing with the media player should retain at least 24 bit resolution. Dithering shall be used when resolution is reduced.

9.7 DD/DD+ Dynamic Range Control

For System A media players, the decoder shall follow the Dolby metadata. If, for example, the DD/DD+ encoder used the DRC=NONE profile, the decoder shall not apply any compression other than overload protection.

- **RF Mode operation**

  The main DD/DD+ decoder in the media player used for stereo reproduction shall apply the RF Mode dynamic range metadata.

- **Line Mode operation**

  For media players offering an additional multi-channel/stereo down-mix DD/DD+ decoder which applies a PCM loudness level of -31 LUFS, the media player shall apply the Line Mode dynamic range metadata by default. DRC settings that apply scaling of the gain reduction are optional. It shall always be possible for the user to switch off DRC and the media player shall store this setting in non-volatile memory, so it keeps the same setting after power up.

9.8 HE-AAC Dynamic Range Control

For System B media players, the main decoder used for stereo reproduction shall follow the metadata in the *dynamic_range_info()* field of the ISO/IEC 14496-3 stream. For media players offering an additional multi-channel/stereo down-mix HE-AAC decoder which applies a PCM loudness level of -31 LUFS, it shall always be possible for the user to switch off DRC and the media player shall store this setting in non-volatile memory, so that it keeps the same setting after power up. For devices offering a monophonic, RF modulated analogue output, the signal on this output should have the DRC metadata '*compression_value*' described in ETSI 101 154 Annex C.5.2.5 applied. When this metadata is not present, the media player shall revert to the DRC metadata in the *dynamic_range_info()* field of ISO/IEC 14496-3 for this output.

9.9 Additional Dynamic Range Control or audio-enrichment features

Additional proprietary dynamic range control and audio-enrichment functionality shall be optional. These features shall always be switched off by default. For specific listening conditions, for example late at night or in a bedroom, it is useful to implement an additional DRC application. This so-called 'Night Mode' shall not just be based on progressively scaled DD/DD+ DRC metadata, as this might not be active (for example if the DD/DD+ encoder applies the profile DRC=NONE) or not applicable (for example using MPEG-1 Layer II audio).

9.10 Down-mixing of multi-channel audio

Multi-channel broadcasts are often presented in the home on two loudspeakers. To achieve this, the (typically) five channels are combined into two by adding a certain amount of the surround channels' signal to the front channels and some of the centre channel's signal to left and right. The amounts may be controlled by down-mix coefficients transmitted with the audio signal. In some broadcast recommendations there is ambiguity regarding the need to scale the down-mix coefficients in order to avoid signal overload, should all channels contain high-level signals. To maintain consistent signal level between down-mixed multi-channel programmes and native stereo programmes, this scaling should not be applied. The content provider shall ensure that sufficient headroom and/or dynamic range control values are included in the stream/file to prevent any overload when down-mixing.

System B media players shall apply the down-mix parameters according to ETSI 101 154 Annex C 5.2.4, *down-mixing_levels_MPEG4* (the parameter with increased resolution over that in ISO/IEC 14496-3).

9.11 Interactive applications

Interactive applications on a media player that make use of accompanying sound can be made consistent with EBU R 128 by normalising the audio in advance by, for example, an algorithm implemented in software. Care shall be taken within the design of the media player so that the signal alignment corresponds to that of the broadcasted audio via all audio interfaces with the aim to achieve an equal integrated loudness level.

9.12 Internet applications

For media players featuring Internet and/or network access, loudness jumps can spoil the quality of experience as audio and video streams can be very loud. Although applications such as Internet access fall outside the scope of the current revision of this document, it is thought that it might be advantageous to include an attenuator for Internet and network stream decoder outputs that can be set by the user.

9.13 Portable and battery-powered devices

See § 2.7.

10. FM Radio and DAB receivers

10.1 Application

The guidelines described in this section are applicable to FM radio receivers and DAB receivers, where the design allows the opportunity to change audio levels by means of a software update.

10.2 Output level relationship on FM radio receivers

The following level alignment shall be applied between input and output interfaces on FM radio receivers:

Radio system	ITU-R BS.450
Modulation	FM stereo radio
Level alignment on the RCA analogue output (1)	35.0 (45.0) kHz FM deviation using a 1 kHz sine wave modulated in phase on left and right channel results in an RMS signal level of 502 mV (±1 dB).
Radio system	ITU-R BS.450
Modulation	FM mono radio
Level alignment on the RCA analogue output	35.0 kHz FM deviation using a 1 kHz sine wave results in an RMS signal level of 502 mV (±1 dB).

Note 1: The listed value represents the FM deviation caused by the 1 kHz tone. Pilot tone, RDS and other additional signals within the FM stereo multiplex signal are assumed to represent in total 10 kHz FM deviation but only influence the available headroom, not the loudness alignment.

Note 2: A graphical representation of level relationships for the FM radio system is shown in § 5.

10.3 Portable and battery-powered FM radio receivers

See § 2.7.

10.4 Output level relationship on DAB and DAB+ receivers

The output level on the SCART and RCA analogue interface shall be 2.0 V RMS using a sine wave at 1 kHz encoded at 0 dBTP (see *Note 1*). Accordingly, 0 dBTP corresponds to a signal peak level of 2.83 V. This specific output level is required in order to align with the analogue modulation levels specified in this document. A graphical representation of level relationships for several television systems is shown in § 5.

The following level alignment shall be applied to DAB/DAB+ receivers:

Radio system	DAB and DAB+
Modulation	Orthogonal frequency-division multiplexing
Level alignment for the digital output (1)	-12 dBTP using a 1 kHz sine wave results in an output level of -12 dBTP.
Level alignment for the analogue output (1)	-12 dBTP using a 1 kHz sine wave results in an RMS signal level of 502 mV (±1 dB).

Note 1: True Peak Level is the maximum peak level of an audio signal measured with an oversampling True Peak Meter. If a True Peak meter is not available, a sine wave at 997 Hz, encoded at the specified level in dBFS, may be used for reference.

Note 2: A graphical representation of level relationships for the FM radio system is shown in § 5.

10.5 Portable and battery-powered DAB receivers

See § 2.7.

11. References

11.1 Normative references

The technical guidelines or specifications contained in this document refer to broadcast recommendations and standards developed by standard-settings organisations, in particular:

[1]	EBU R 128 (2011)	Loudness normalisation and permitted maximum level of audio signals.
[2]	ISO/IEC 14496-3 (2009)	Information technology — Coding of audio-visual objects — Part 3: Audio.
[3]	ETSI TS 102 366 v1.2.1 (2008)	Digital Audio Compression (AC-3, Enhanced AC-3) Standard.
[4]	ITU-R BS.450-3 (2001)	Transmission standards for FM sound broadcasting at VHF.
[5]	ITU-R BS.707-5 (2005)	Transmission of multi-sound in terrestrial television systems.
[6]	ITU-R BT.2043 (2004)	Analogue television systems currently in use throughout the world.
[7]	ITU-R BS.642-1 (1990)	Limiters for high quality sound programme signals.
[8]	ITU-R BS.412-9 (1998)	Planning standards for FM sound broadcasting at VHF.
[9]	IEC 60268-10 (1991)	Sound system equipment — Peak programme level meters.
[10]	EBU Tech 3341 (2011)	Loudness Metering — 'EBU Mode' metering to supplement loudness normalisation in accordance with EBU R 128.
[11]	EBU Tech 3342 (2011)	Loudness Range: A descriptor to supplement loudness normalisation in accordance with EBU R 128.
[12]	EBU Tech 3343 (2011)	Practical Guidelines for Production and Implementation in accordance with EBU R 128.
[13]	ITU-R BS.645-2 (1992)	Test signals and metering to be used on international sound programme connections.
[14]	CENELEC EN50049 (2000)	Domestic and similar electronic equipment interconnection requirements: Peritelevision connector.
[15]	ITU-T J.17 (1988)	Pre-emphasis used on sound-programme circuits
[16]	ITU-R BS.1770-2 (2011)	Algorithms to measure audio programme loudness and true-peak audio level.
[17]	IEC EN60728-5 (2007)	Cable network equipment for television signals, sound signals and interactive services — Head-end equipment.
[18]	EBU Tech 3333 (2009)	EBU HDTV Receiver Requirements.
[19]	ETSI TS 101 154 v1.10.1 (2009)	Specification for the use of Video and Audio Coding in Broadcasting Applications based on the MPEG-2 TS

11.2 Informative references

[#1]	Dolby Technical Bulletin 11 (2010)	Requirement Updates for Dolby Decoders in DVB Consumer Broadcast Receivers.
[#2]	Arne von Ruschkowski — Hamburg University (2007)	Loudness war: a psychoacoustic investigation of popular music CDs.

12. List of abbreviations

AC3	Audio Coding 3 (also known as Dolby Digital)
AES	Audio Engineering Society
AM	Amplitude Modulation
ARC	Audio Return Channel
AV	Audio/Video
CEC	Consumer Electronics Control
CENELEC	Comité Européen de Normalisation Electrotechnique (European Committee for Electrotechnical Standardisation)
Codec	Encoder and decoder system (coder/decoder)
DAB	Digital Audio Broadcasting
DAB+	DAB using the AAC codec
DAC	Digital to Analogue Converter
dB	decibel
dBTP	decibel True Peak
DD	Dolby Digital (also known as AC3, Audio Coding 3)
DD+	Dolby Digital Plus (also known as E-AC3, Enhanced Audio Coding 3)
DRC	Dynamic Range Control
DTS	Digital Theatre Systems
DVB	Digital Video Broadcasting
E-AC3	Enhanced Audio Coding 3 (also known as Dolby Digital Plus)
EBU	European Broadcasting Union
E-EDID	Extended Display Identification Data
ETSI	European Telecommunications Standards Institute
FM	Frequency Modulation
HDMI	High-Definition Multimedia Interface
HE-AAC	High Efficiency Advanced Audio Coding
HTM	Home Theatre Mode
IDTV	Integrated Digital (or Decoder) TeleVision
IEC	International Electro-technical Commission
IRD	Integrated Receiver Decoder (also known as STB, Set-Top Box)
ITU-R	International Telecommunication Union – Radio communications sector
ITU-T	International Telecommunication Union - Telecommunications sector

LU	Loudness Unit
LUFS	Loudness Unit relative to Full Scale
MCA	Multi-Channel Audio
MPEG	Moving Pictures Experts Group
PAL	Phase Alternation by Line
MPX	Multiplex
PCM	Pulse Code Modulation
PRL	Programme Reference Level
QPPM	Quasi-Peak Programme Meter
PSI/SI	Program Specific Information/Service Information
RCA	Radio Corporation of America
RF	Radio Frequency
RMS	Root Mean Square
SCART	Syndicat des Constructeurs d'Appareils Radiorécepteurs et Téléviseurs (radio and television receiver manufacturers' association)
SPDIF	Sony Philips Digital Interface
STB	Set-Top Box (also known as IRD, Integrated Receiver Decoder)
THX	Tomlinson Holman's eXperiment
TL	Target Level
VOD	Video On Demand