EBU

OPERATING EUROVISION AND EURORADIO

TECH 3344

GUIDELINES FOR DISTRIBUTION AND REPRODUCTION IN ACCORDANCE WITH EBU R 128

SUPPLEMENTARY INFORMATION FOR R 128

VERSION 2.1

Geneva
July 2016

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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.

Informative text is potentially helpful to the user, but it is not indispensable and it can be removed, changed or added editorially without affecting the normative text. Informative text does not contain any conformance keywords.

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.

Acknowledgements

Although this document is the result of much collaborative work within the EBU's PLOUD group, it would not have seen the light of day without the skill, dedication and effort of Richard van Everdingen, supported by Ian Rudd, who honed the text into its publication form. Very many thanks are due.

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Guidelines for distribution and reproduction in accordance with EBU R 128

EBU Committee	First Issued	Revised	Re-issued
TC	2011	2016*	

Keywords: Loudness, normalisation, transmission, distribution, head-end, consumer equipment, Integrated Receiver Decoder (IRD), Integrated Digital Television (IDTV), Media Player, FM Radio, DAB, Set-top-box, STB.

1. Scope

This document represents guidelines for broadcast distribution and reproduction. The guidelines are meant to specify relevant settings and processing in the signal chain from the studio up to and including consumer equipment.

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 [1] 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 television sets and set-top boxes, Home Theatre Equipment (such as AV-receivers), Media Players and radio receivers.

2. Guidelines for distribution and reproduction of television and radio services

2.1 Objectives and basic principles

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 consumer as well as professional equipment and specifies levels for encoding, decoding and modulation.

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.

* July 2016 version corrects some typography and pagination. No material content change occurred.

- 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 or be delivered via a separate path, which thereby avoids needless limitation of the digital transmission systems. It has to be pointed out that pre-emphasis limiting, although absolutely necessary if analogue modulation is expected, must not be used on signals with digital-only distribution, in order to offer the highest transparent quality for the digital listener.

- 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 in this document and implement them in their own specifications and operational workflow. Distributors are encouraged to actively support the aim to achieve loudness normalisation in radio and television broadcasting.

2.2 Use of the EBU loudness logo

Any company that develops, manufactures, distributes or otherwise markets, advertises or promotes products, equipment (for example, a 'family of products') or services which support signal level management according to EBU Tech Doc 3344, is entitled to use the EBU loudness logo as shown below free-of-charge in order to demonstrate the product's, equipment's or service's technical compliance with the EBU R 128 specification. Please check https://tech.ebu.ch/loudness for details.

Figure 1: EBU loudness compliance logo

TECH 3344 PART 1 – DISTRIBUTION

3. Distribution

§§ 3 & 4 describe the guidelines for audio distribution of radio and television signals.

3.1 Tech 3344 compliance list for Distribution

The following items are specified to achieve compliance with this Tech Doc. Involved parties: broadcasters, content distributors and broadcast electronics equipment industry.

Subject	Section	Requirement
Loudness level adaptation in distribution systems	3.4	Optional
Level alignment between systems and interfaces	4.1	Mandatory
Modulation levels for analogue television and radio systems	4.2	Mandatory
(Pre-emphasis) limiting on analogue transmission systems	4.2	Mandatory

3.2 Analogue television and FM radio transmission via cable networks

Traditionally, preparation for radio and television audio broadcast has been done by the content provider taking into account 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. It also avoids needless attenuation of loudness levels on digital networks, due to the effect of pre-emphasis processing. This approach applies to the FM radio system specified in ITU-R BS.450-3 [2] and to the television systems using FM, AM and NICAM audio carriers specified in ITU-R BS.707 [3] with the Note that for the AM system L according to ITU-R BT.2043 [4] no pre-emphasis is used. For FM radio, EBU Tech 3344 defines a loudness reference, independent from stereo or mono modulation and irrespective of the usual amount of bandwidth used for additional signals in the FM multiplex.

3.3 Service Loudness

In Master Control Rooms as well as in the distribution stage it is very useful to measure the loudness of the radio and television services over a full day. A simple integrated measurement over 24 hours would however be vulnerable for errors, for example caused by programmes with background sound only. Therefore the measurement 'Service Loudness' is defined in this Tech Doc. The measurement works as follows: The loudness of decoded signals is continuously measured over a full day, split up into 24 blocks of one hour each. The start time of block number one is 03:00 a.m.; the start time of block number 24 is 02:00 a.m. the following day. The reason for applying this time of the night is to have minimal influence on daily programming when applying an overall gain correction. The integrated (I) measurement as specified in EBU R 128 is applied on the individual blocks. For codecs that use loudness indicating metadata, this correction is included in the measurement at all times in order to retrieve the reproduction level. The measurement system applies a loudness reference level of −23 LUFS for MPEG-1 Layer II and -31 LUFS for codecs which carry loudness level metadata. 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. Figure 2 shows a graphical representation of the measurement in case of MPEG-1 Layer II. Figure 3 shows the same for codecs which carry loudness indicating metadata. Next to Service Loudness, it is also beneficial to measure True Peak levels in this same system, in that case however without loudness metadata correction. If these levels repeatedly reach higher than -1 dBTP before calculation of the loudness indicating metadata, the broadcaster is advised to check the True Peak limiter. However, bit errors in the reception can also generate signal level clipping.

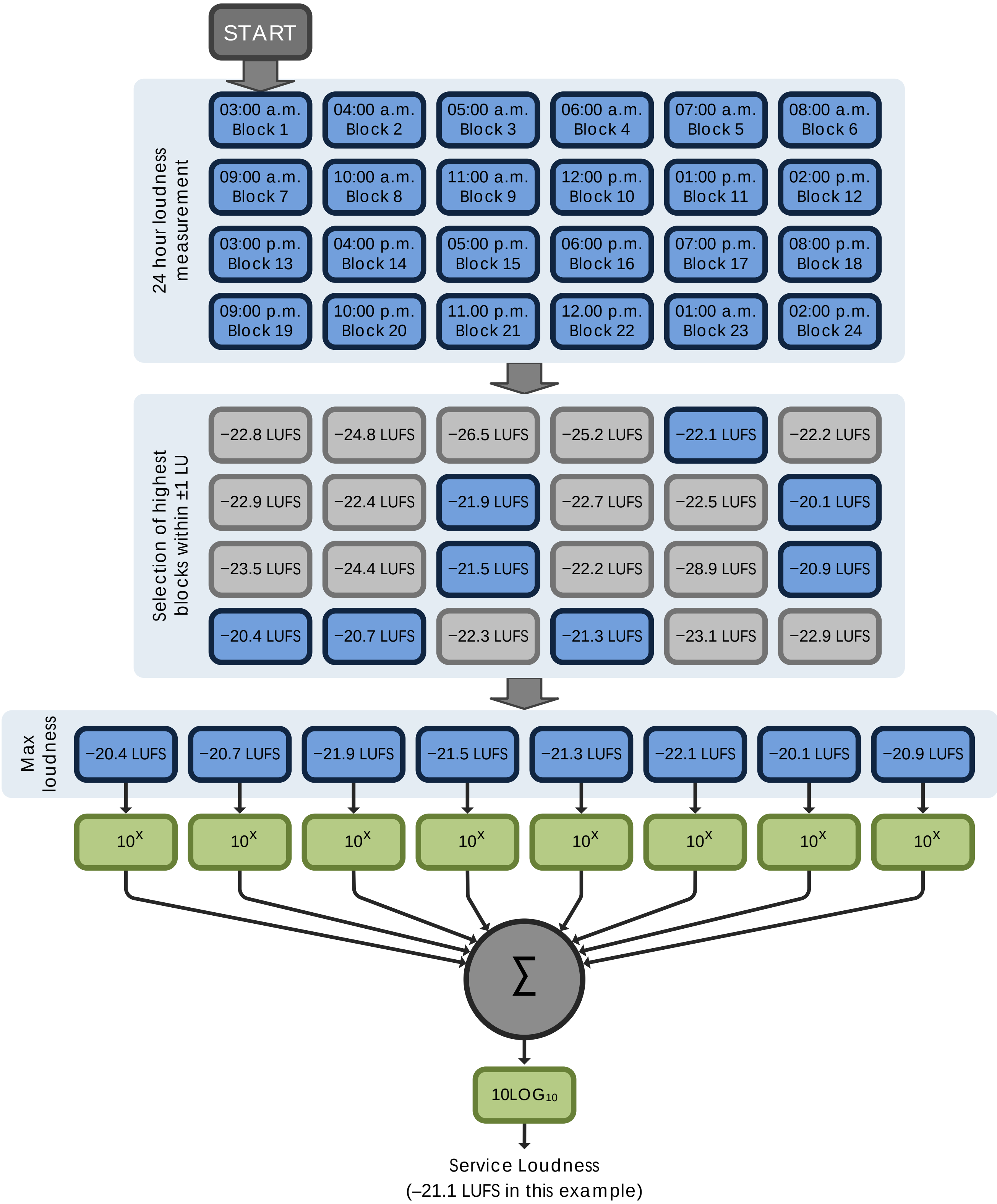

Figure 2: Service Loudness measurement for MPEG-1 Layer II

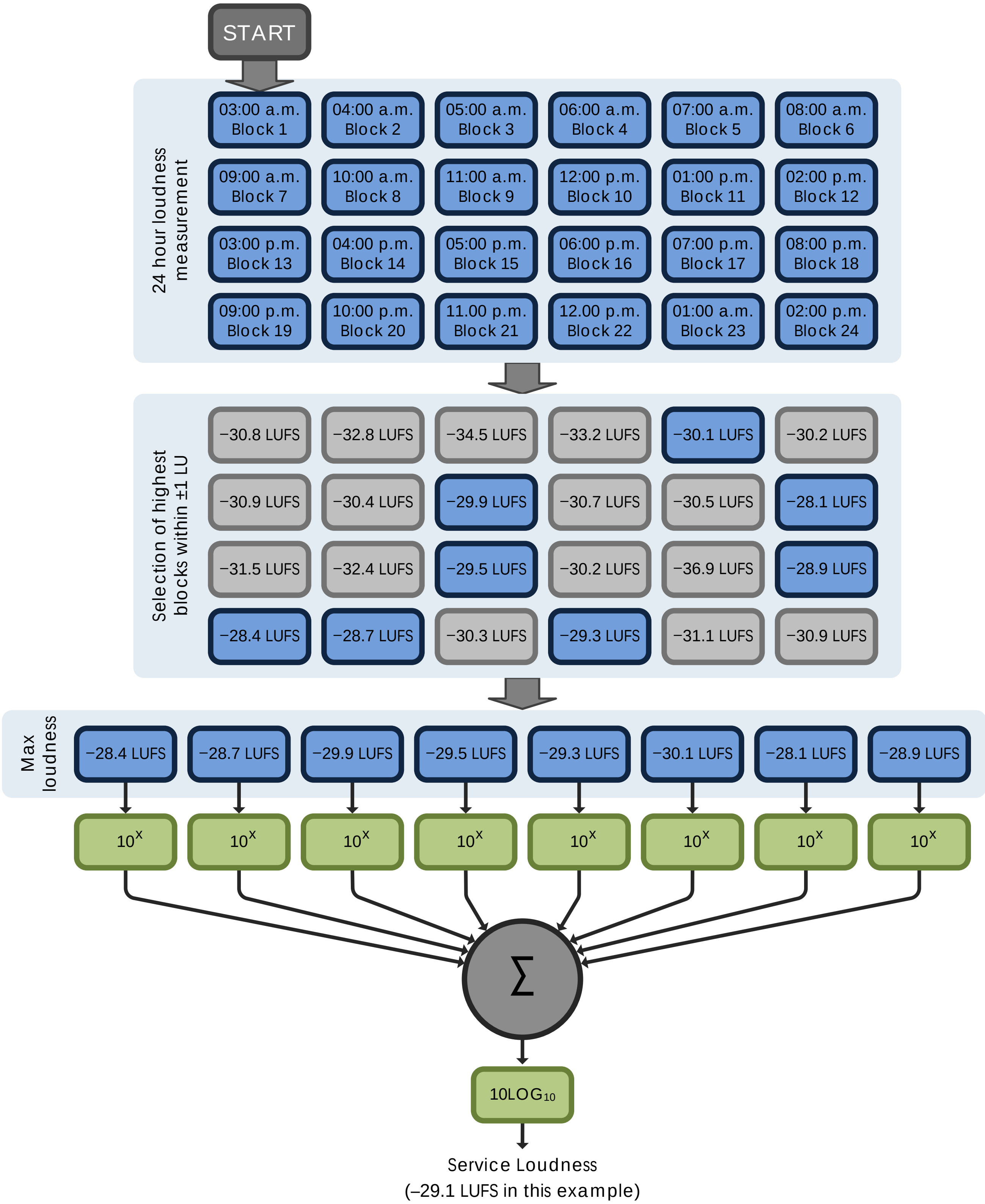

Figure 3: Service Loudness measurement for codecs carrying loudness indicating metadata

3.4 Head-ends

Figure 4 shows an example of a typical DVB head-end; its application to other platforms, for example IPTV distribution, can be derived from this figure. The acquisition multiplexer combines services into transport streams. A loudness adaptation device can be present if necessary to correct services that have a Service Loudness which deviates heavily from the EBU R 128 Target Level (see Note 1 and § 3.3 for details).

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 [5]. 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. As indicated in Figure 4, the added advantage is that the same signal meant for digital television can be used to feed analogue modulation.

Alternatively, the broadcast station can 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 signal. Because of reasons described in § 4.2 on modulation levels for analogue systems, 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.

Note 1: In case of loudness adaptation based on changing the scale factors in MPEG-1 Layer II codec systems and on changing the loudness indicating metadata in other codec systems, it is recommended to restrict the correction of reproduced loudness level to attenuation only, by default. This is suggested to avoid potential clipping at the output of the decoder in MPEG-1 Layer II systems and at the output of the down-mix decoder of multichannel codecs. If amplification of the reproduced loudness level nevertheless is applied, it is strongly recommended that detection of overloads is continuously included in the consideration to raise the level over a long period of time. In normalisation systems that feed analogue modulation equipment, it is recommended to allow both attenuation and amplification in order to optimise the signal to noise ratio.

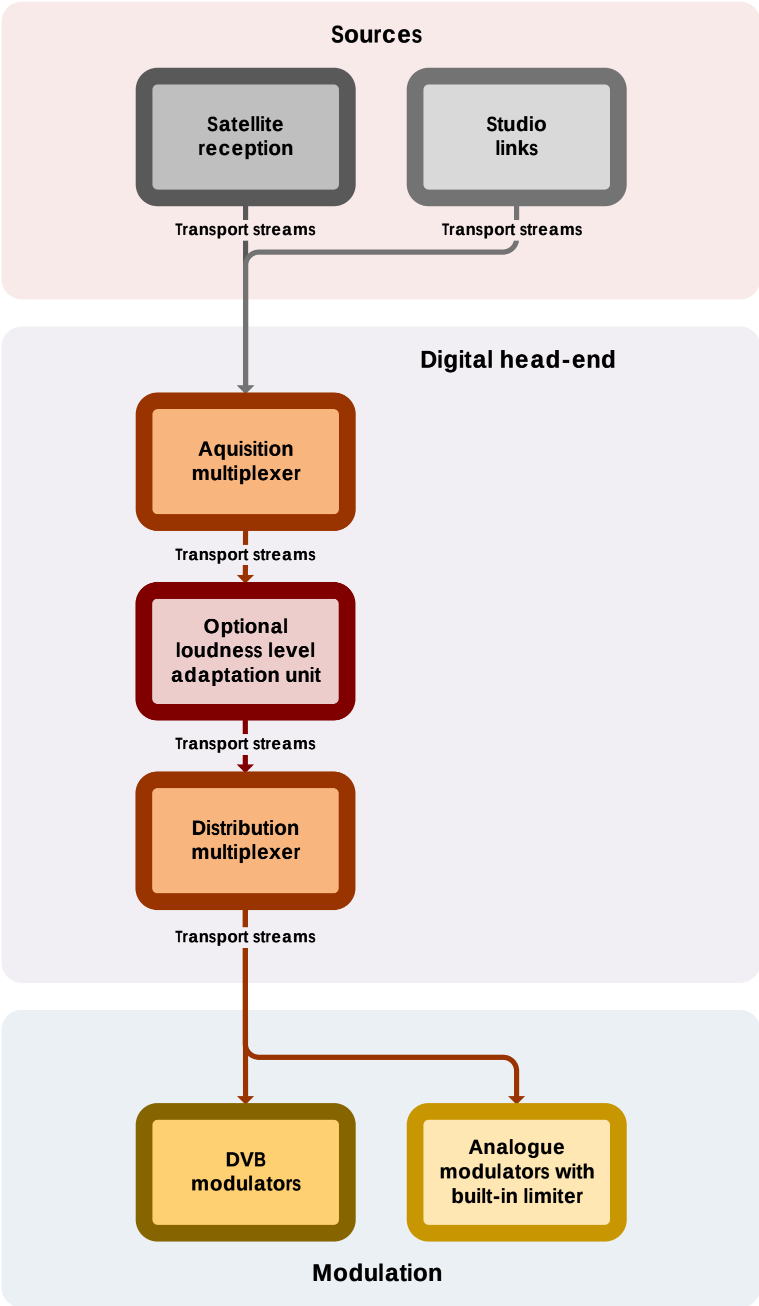

Figure 4: Audio processing in the head-end

3.5 Terrestrial analogue television and terrestrial 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. 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/DAB+, or it is recommended that the approach is followed for analogue cable distribution as 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 and needless attenuation of loudness levels due to the effect of pre-emphasis processing is avoided. Alignment and modulation levels are as for those of cable networks and can be found in § 4.

The loudness level paradigm for FM radio on cable networks described in this document can also be used for terrestrial FM radio transmission and can serve as an alternative for terrestrial planning standard ITU-R BS.412 [6]. Although the described approach can neither achieve a strict limit nor maintain a target of an MPX power level if the MPX power application is replaced by loudness normalisation, the goals of the terrestrial planning standard remain to be served as on average the power levels are more or less restricted. An offset to the alignment is possible if a higher MPX power level on average is desired, depending on local rules. It is recommended that future legislation shall not only be based on maximum total FM deviation or on 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.

3.6 Loudness matching in mobile DAB/DAB+ and FM radio networks

EBU Tech 3344 includes the alignment of digital radio and FM radio via cable systems and terrestrial transmission. On terrestrial FM radio and DAB/DAB+ networks the loudness level of the same radio service has, historically, been very different. Although loudness normalisation on terrestrial networks according to EBU Tech 3344 is recommended, a workaround is included in the specification for FM radio and DAB/DAB+ receivers in § 6.10 that establishes adaptive loudness matching of linked services.

4. Level alignment in analogue and digital distribution systems

4.1 Level alignment between systems and interfaces

This section gives an overview of the level alignment for European television and radio transmission systems. The alignment schemes are compliant with CENELEC EN 50049 [7]; 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).

4.2 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.

A 1 kHz sine wave is used as a reference, according to CENELEC EN 50049. 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 both left and right channel results in 50 kHz FM deviation. −12 dBTP using a 1 kHz sine wave in phase on both left and right channel results in 27 kHz FM deviation.
Limiter threshold (1, 2)	−6.7 dBTP referenced to 1 kHz (−7 dBTP can be used as practical value).
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 both left and right channel results in 96% AM modulation depth. −12 dBTP using a 1 kHz sine wave in phase on both left and right channel results in 54% 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 [8]
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 (1, 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)	−9.7 dBTP using a 1 kHz sine wave in phase on both left and right channel results in 65 (75) kHz FM deviation. −12 dBTP using a 1 kHz sine wave in phase on both left and right channel results in 50 (60) kHz FM deviation.
Limiter threshold (1, 2)	−9.7 dBTP referenced to 1 kHz (−10 dBTP can be used as practical value).
Pre-emphasis limiting	50 μs
Low-pass filter	15 kHz
Radio system	**ITU-R BS.450-3**
Modulation	FM mono (no pilot, no RDS)
Level alignment (1, 4, 5)	−8.5 dBTP using a 1 kHz sine wave results in 75 kHz FM deviation. −12 dBTP using a 1 kHz sine wave results in 50 kHz FM deviation.
Limiter threshold (1, 2)	−8.5 dBTP referenced to 1 kHz (−9 dBTP can be used as practical value).
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 to the following causes it could be necessary to decrease the pre-emphasis limiter level **a few tenths to several dB** to avoid exceeding 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 [9] 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 filtering while 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 influences the maximum headroom available for audio and the corresponding limiter level. In all cases, the loudness reference for the measurement and the alignment 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 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 remains the same.

Note 5: -12 dBTP corresponds to an analogue relative level of +6 dBu0s, as specified in ITU-R BS.645 [10]. 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.

4.3 Graphical representation of level alignment between systems and interfaces.

The figures in Annex A show a graphical representation of the level alignment between systems and interfaces.

4.4 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 EN 60728-5 [11]. 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.

TECH 3344 PART 2 – REPRODUCTION

5. Reproduction

§§ 5 & 6 describe the guidelines for audio reproduction of radio and television signals.

5.1 Tech 3344 compliance list for Reproduction

The following items are specified to achieve compliance with this Tech Doc. Involved parties: content distributors and consumer electronics equipment industry.

Subject	Section	Requirement
Level adaptation −23 LUFS	6.4	Mandatory
Level adaptation −31 LUFS	6.4	Mandatory
Level adaptation −27 LUFS	6.7	Optional
Level alignment between interfaces	6.5	Mandatory
Audio processing in mobile devices	6.9	Mandatory
User Volume Control	6.11	Mandatory
Transcoding loudness level and metadata consistency	6.12	Mandatory
Bitstream pass-through consistency	6.13	Mandatory
Dynamic Range Control	6.14	Mandatory
Down-mixing behaviour of multi-channel audio	6.15	Mandatory
Behaviour of mono audio at stereo and multi-channel outputs	6.16	Mandatory
Audio preference settings	6.17	Mandatory
HDMI E-EDID dependency	6.18	Optional
Analogue level alignment	6.20	Mandatory

5.2 Loudness and down-mix consistency in consumer equipment

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. Even if the channels leave the broadcast studio with the correct loudness levels, it is difficult to maintain these levels through the chain. Due to a multiplicity of working methods, reproduction can be spoiled by loudness jumps of 11 dB or more when switching from one service to another, using for example a set-top box connected to a Home Theatre Device. This occurs if one service is for example using MPEG-1 Layer II audio and the other one is delivering AC-3. MPEG-1 Layer II audio is usually experienced louder in that case. The use of the internal decoder of the set-top box or the television set can lead to a loudness jump of typically 3 dB, but now in the opposite direction, which means that the AC-3 signal is perceived louder than a service that supplies MPEG-1 Layer II. The difference in level and the opposite direction of the loudness jumps that occur between the codecs depending on the reproduction situation does not only lead to confusion, but also means that the broadcast station cannot solve this problem simply by adjusting the input level of the encoders.

In practice these variations can increase or decrease existing loudness differences between services. Because the outcome can also vary between different brands and models of set-top boxes, it is impossible for a broadcast station to transmit with a guaranteed reproduction result in such a situation. To make matters even worse, 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.

Last but not least, set-top boxes and IDTVs rarely offer the option to choose the stereo down-mix format. This is either Left only/Right only (Lo/Ro) or Left total/Right total (Lt/Rt). One would expect that if there is no user choice, the device follows the preference set by the broadcaster and transmitted by metadata. However, in many cases the device is programmed by the manufacturer to use either one of the schemes. This results in a rather different reproduction of the stereo signal and occasional differences in loudness. Again, the system design can differ between brands and between models of the same brand. To counteract all these problems, this document contains extensive guidelines for consumer equipment.

6. Audio processing and level alignment in reproduction equipment

6.1 Application

The guidelines described in this section are applicable to Integrated Receiver Decoders (IRDs), Integrated Digital Televisions (IDTVs), Media Players (including mobile devices such as Personal Music Players), Home Theatre Equipment and DAB/DAB+/FM radio receivers. A set-top box is referred to as an IRD.

6.2 Device types

For the purpose of defining the appropriate reference loudness level, device types are divided into two groups:

- Devices that can be characterised as being stereo, such as TV sets, Hi-Fi systems and Personal Music Players. The loudness reference for this kind of device is −23 LUFS.
- Devices that can be characterised as being multi-channel surround sound, such as Home Theatre Equipment. The loudness reference for this kind of device is −31 LUFS.

Note 1: More advanced Home Theatre Equipment can include a separate decoder which applies a PCM loudness level of -23 LUFS for output to stereo analogue RCA interfaces or stereo PCM to SPDIF and/or HDMI interfaces. Signals from other sources shall have a gain of +8 dB applied to the stereo outputs, as shown in Figure B.6. Distortion due to numerical overload can be prevented by a True Peak limiter as shown in the same figure.

Note 2: A multi-channel surround sound 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.

Note 3: A so-called Soundbar device is considered to be Home Theatre Equipment.

6.3 Codec types

Audio coding systems used for digital television and radio broadcasting applications are, but not limited to:

- MPEG-1 Layer II according to ISO/IEC 11172-3 [12]
- AC-3 and E-AC-3 according to ETSI TS 102 366 [13]
- AC-4 according to ETSI TS 103 190 Part 1 [14]
- MPEG-4 AAC/HE-AAC according to ISO/IEC 14496-3 [15]
- MPEG-H 3D Audio according to ISO/IEC IS 23008-3 [16]

Some of these codecs make use of accompanying metadata and a Programme Reference Level (PRL) parameter while others, such as MPEG-1 Layer II, do not. In AC-3, E-AC-3 and AC-4 this indicator is known as 'dialnorm'. MPEG-H 3D Audio refers to it as 'program loudness'. For MPEG-1 Layer II and MPEG-4 AAC/HE-AAC sent without a PRL parameter, the audio signal is assumed to be delivered with an average loudness of−23 LUFS. For the other codecs it is assumed that the PRL parameter has been set correctly.

6.4 Loudness alignment

Figure 5 shows a graphical representation of a radio and television head-end feeding consumer equipment by means of a cable distribution network or a satellite path. 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.

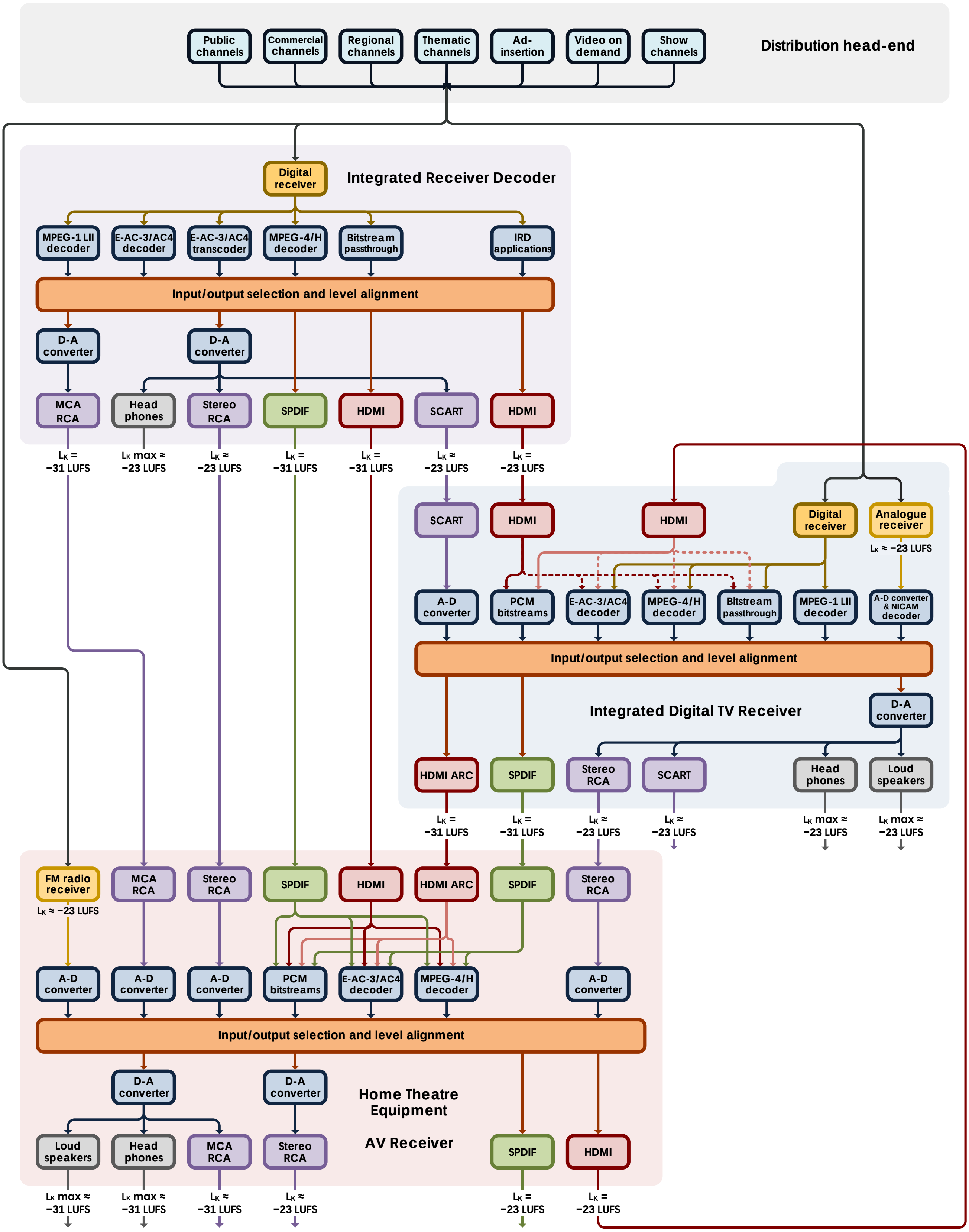

Figure 5: Interconnection of consumer equipment with indicated loudness levels

Figure 5 is set up as follows: After transmission over the 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, depending on brand and model, some of these devices are able to apply internal decoding of codec bitstreams supplied to the HDMI input whilst others are not. This difference is one of the reasons for quite different and confusing behaviour regarding loudness levels between television sets if they are connected to, for example, an IRD. Apart from this, the TV set can be connected to the IRD directly, but can also receive its input signal from the Home Theatre Device. The radio signal is received by the Home Theatre Device, by the IRD or by a separate FM or FM/DAB/DAB+ tuner.

The correct loudness levels are indicated at the outputs. The term ‘LK’ refers to loudness (Loudness, K-weighted; K-weighting is the filter curve used in ITU-R BS.1770 at the core of the loudness measurement algorithm). On analogue outputs, the term ‘LK ≈’ refers to the PCM equivalent relative loudness of the decoded signal based on the mappings specified in this document between the levels in the analogue and digital domain. The word ‘max’ in the term ‘LK max ≈’ refers to the use of a volume control in the device where the maximum setting corresponds to the indicated loudness level, before optional power amplification that may be present on that particular output.

The way that loudness alignment must be achieved depends on the codec used to deliver the audio and the connected device.

6.4.1 MPEG-1 Layer II

- For a reference loudness level of −23 LUFS no alteration to the level of the signal is required.
- For a reference loudness level of −31 LUFS apply a gain of −8 dB to the signal.

6.4.2 AC-3 and E-AC-3

- For a reference loudness level of −23 LUFS use RF Mode, then apply a gain of −3 dB to the signal.
- For a reference loudness level of −31 LUFS use Line Mode.

6.4.3 AC-4

- For a reference loudness level of −23 LUFS use Flat Panel Mode.
- For a reference loudness level of −31 LUFS use Home Theatre Mode.

6.4.4 MPEG-4 AAC and HE-AAC

- For a reference loudness level of −23 LUFS set ‘target_level’ to −23 dBFS.
- For a reference loudness level of −31 LUFS set ‘target_level’ to −31 dBFS.

6.4.5 MPEG-H 3D Audio

- For a reference loudness level of −23 LUFS set ‘targetLoudness’ to −23 dBFS.
- For a reference loudness level of −31 LUFS set ‘targetLoudness’ to −31 dBFS.

6.5 Level alignment between interfaces

To avoid loudness jumps when switching between PCM output and codec bitstreams on the SPDIF and HDMI outputs, it is strongly recommended to include selections via a user interface. The purpose of this is to allow choosing what type of sink device is connected to the source device, such that the source device can use the appropriate reference loudness level. Two modes can be distinguished in this Tech Doc:

Mode	Description
TV Mode	The output is connected to a sink device that is based on a loudness reference level of −23 LUFS. The menu dependent attenuator[1] is set to 0 dB.
Home Theatre Mode	The output is connected to a sink device that is based on a loudness reference level of −31 LUFS. The menu dependent attenuator1 is set to 8 dB so that an incoming loudness level of −23 LUFS is attenuated to −31 LUFS which is the appropriate reference level for Home Theatre Equipment.

6.6 Graphical representation of the audio processing within devices

The figures in Annex B show a graphical representation of the audio processing and level alignment within the devices.

6.7 Home Theatre Equipment referenced to −27 LUFS

It appears that a substantial number of Home Theatre Devices process PCM input signals with a fixed 4 dB offset compared to the reference loudness level of −31 LUFS. 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 these Home Theatre Devices, an alternative offset mode may optionally be included in the IRD, IDTV and Media Player where the loudness of PCM output signals is adjusted to −27 LUFS. The level alignment figures in Annex A show these levels as 'HTM Offset'.

6.8 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 § 3 and § 4 for implementation guidelines.

6.9 Audio processing in mobile 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 mobile devices like Personal Music Players and phones, it is strongly recommended that equipment like that has a volume control stage shown in Figure B.5 with a range that includes sufficient positive gain to allow a Target Level of −23 LUFS. A limiter after the positive gain circuit but before the digital to analogue converter and headphone output prevents distortion due to numerical overload.

Safety requirements to prevent hearing loss can be addressed by software simulated measurement of the acoustic output level. The dose limiter shown in Figure B.5 continuously measures and controls the output signal and enables protection for the user at all times, irrespective of the loudness level of the media content. This core system is supposed to be part of the firmware of the player, but can potentially work in tandem with more sophisticated loudness features in codec systems and/or Media Player applications. In that way the protection can be fine-tuned over the duration of a programme, a song or an entire album.

6.10 Loudness matching in combined DAB/DAB+ and FM radio receivers

Receivers that use service linking as described in ETSI TS 103 176 [17] shall contain a feature that is able to match the loudness of the FM radio receiver to that of the DAB/DAB+ receiver. Figure B.7 shows a simple block diagram of the recommended loudness-matching mechanism. Each audio output of the FM and DAB receiver of the linked service is analysed for audio programme loudness integrated over a certain time, for example one minute. The results are kept in a memory, which is

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[1] See figures in Annex B for clarification.

assigned to the station-ID. The loudness difference between DAB and FM controls a static gain stage in the FM path, allowing loudness matching of the signal presented to the listener. Service-dependent audio delays are required to compensate for different propagation delays of the two distribution paths. A manual setting for the 'gain stage' block can override the automatic process for FM services that cannot be linked to DAB. It is strongly recommended to implement a changeable default gain setting in the user menu, indicated by the block 'Manual loudness setting', that can be applied on newly received FM services.

In general, it is recommended to leave the output level of the DAB decoder unchanged to stimulate the use of an integrated loudness level of −23 LUFS on DAB transmission networks. However, if the manufacturer wishes to implement an offset gain setting for DAB, it is strongly recommended to place it in front of the block 'Loudness & sync analysis'. The alignment for FM radio will subsequently follow this manual gain offset automatically.

For DAB/DAB+ receivers featuring Internet and/or network access, loudness jumps can spoil the quality of experience as these audio sources 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 apply the identical approach as for FM radio.

6.11 User Volume Control

A volume control provided to a user by a source device should not affect the signal level on its digital outputs (SPDIF and HDMI). 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.

6.12 Transcoding consistency

- When transcoding from MPEG-4 AAC to AC-3, the level of the audio should be preserved and the metadata should be transcoded to ensure the correct reproduction level on decoding.
- When transcoding from MPEG-4 AAC to DTS, audio should be decoded for a reference loudness level of −31 LUFS before being encoded into DTS.

6.13 Bitstream pass-through consistency

When a codec bitstream is passed-through, no changes shall be made to the audio data or the metadata.

6.14 Dynamic Range Control

Dynamic Range Control (DRC) can optionally be used in some codecs. The audio decoder shall follow the metadata, which means that if, for example, the encoder uses no DRC, the decoder shall not apply any control other than overload protection. The way that DRC is employed depends on the codec used to deliver the audio and the reference level and is described in the following sections.

6.14.1 MPEG-1 Layer II

DRC is not included in this codec.

6.14.2 AC-3 and E-AC-3

- For a reference loudness level of −23 LUFS on stereo outputs, use RF Mode. Scaling of DRC gain words is not allowed.
- For a reference loudness level of −31 LUFS on multi-channel outputs, use Line Mode. If a down-mix of multichannel audio is performed, scaling of negative gain words is not permitted. Otherwise, scaling of DRC gain words is allowed.

6.14.3 AC-4

- For a reference loudness level of −23 LUFS on stereo outputs, use Flat Panel Mode. Scaling of DRC gain words is not allowed.
- For a reference loudness level of −31 LUFS on multi-channel outputs, use Home Theatre Mode. If a down-mix of multichannel audio is performed, scaling of negative gain words is not permitted. Otherwise, scaling of DRC gain words is allowed.

6.14.4 MPEG-4 AAC and HE-AAC

The decoder shall apply DRC depending on the DVB 'drc_presentation_mode' as defined in ETSI TS 101 154 [18], Annex C, section C.5.2.2.3. If 'drc_presentation_mode' is not signalled or set to '00', it should be assumed to be either mode 1 or mode 2 according to regional specifications.

If DRC presentation mode 1 is signalled:

- For a reference loudness level of −23 LUFS on stereo outputs, apply the DRC metadata 'compression_value' as described in ETSI 101 154 Annex C.5.2.2.3. If this metadata is not present, the decoder shall revert to the DRC metadata in the dynamic_range_info() field of ISO/IEC 14496−3. Scaling of DRC gain words is not allowed.
- For a reference loudness level of −31 LUFS on multi-channel outputs, apply the DRC metadata in the dynamic_range_info() field of ISO/IEC 14496−3. If a down-mix of multichannel audio is performed, scaling of negative gain words is not permitted. Otherwise, scaling of DRC gain words is allowed.

If DRC presentation mode 2 is signalled:

- For a reference loudness level of −23 LUFS on stereo outputs, apply the metadata in the dynamic_range_info() field of the ISO/IEC 14496-3 bitstream. Scaling of negative gain words is not allowed.
- For a reference loudness level of −23 LUFS on a monophonic output, apply the DRC metadata 'compression_value' described in ETSI 101 154 Annex C.5.2.5. If this metadata is not present, the IRD shall revert to the DRC metadata in the dynamic_range_info() field of ISO/IEC 14496−3. Scaling of DRC gain words is not allowed.
- For a reference loudness level of −31 LUFS on multi-channel outputs, apply the DRC metadata in the dynamic_range_info() field of ISO/IEC 14496−3. If a down-mix of multichannel audio is performed, scaling of negative gain words is not permitted. Otherwise, scaling of DRC gain words is allowed.

6.14.5 MPEG-H 3D Audio

- For a reference loudness level of −23 LUFS on stereo outputs, apply the DRC set associated with a target loudness level of −23 LUFS. Scaling of negative DRC gain words is not allowed.
- For a reference loudness level of −31 LUFS on multi-channel outputs, apply the DRC set associated with a target loudness level of −31 LUFS. If a down-mix of multichannel audio is performed, scaling of negative gain words is not permitted. Otherwise, scaling of DRC gain words is allowed.

6.15 Down-mixing behaviour 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 levels 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. The way that DRC is employed depends on the codec used to deliver the audio and the reference level and is described in the following sections.

6.15.1 AC-3, E-AC-3 and AC-4

It is recommended that the IRD, IDTV and Media Player include the following user setting for the down-mix mode:

Item	Choice	Description
DOWN-MIX MODE	AUTO	The IRD follows the Preferred Stereo Down-mix Mode signalled in metadata, if present in the bitstream.
	Lo/Ro	The IRD applies the Left only/Right only down-mix mode.
	Lt/Rt	The IRD applies the Left total/Right total down-mix mode.

If the user setting for the down-mix mode is not implemented, the device shall follow the Preferred Stereo Down-mix Mode signalled in metadata.

6.15.2 MPEG-4 AAC and HE-AAC

The decoder shall apply the down-mix parameters according to ETSI 101 154 Annex C 5.2.4, down-mixing_levels_MPEG-4 (the parameter with increased resolution over that in ISO/IEC 14496-3).

6.15.3 MPEG-H 3D Audio

The decoder shall apply the down-mix parameters of DownmixMatrixSet( ) for the target loudspeaker layout, if present. If no down-mix parameters are available, the default down-mix of the format converter in the decoder shall be applied.

6.16 Behaviour of mono audio at stereo and multi-channel outputs

The equipment shall ensure that a correctly signalled mono audio signal with a loudness level of -23 LUFS will have the same loudness measured at the stereo or multi-channel outputs as a correctly signalled stereo signal with a loudness level of -23 LUFS. If this is not performed in the internal decoder unit itself, the general approach is to reduce the level of the decoder output by 3 dB before passing it to the left and right channel outputs.

6.17 Audio preference settings

A service can supply more than one audio signal. In general, the user may or may not have a preference for AC-3, E-AC-3, AC-4, MPEG-4 or MPEG-H encoded bitstreams (if they are supplied with the service) instead of MPEG-1 Layer II. It is strongly recommended to include a general setting in the user preference menu to assist the user in automatically choosing the preferred setting if a new service is added and to implement a service dependent setting stored in non-volatile memory that overrides the general setting and is applied again once the user chooses the same service the next time.

6.18 HDMI E-EDID dependency

The use of the HDMI E-EDID to override the loudness level selection is not recommended, but optional.

6.19 Interactive applications

Interactive applications on a device 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 device 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.

6.20 Analogue output level alignment

The following level alignment shall be applied to analogue input and output interfaces:

Level alignment for the analogue RCA and SCART input & output (1, 2, 4)	−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, 2, 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: 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 EN 50049 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 3: 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.

Note 4: The alignment for analogue modulation systems can be found in § 4 and Annex A.

Annex A: Graphical representation of level alignment between systems & interfaces

A.1 Clarification

The figures in § A.2 - § A.7 show a graphical representation of the level alignment between systems and interfaces. This signal level relationship is only valid for all systems using the alignment sine tone on 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. If there is no purple line visible, the red line is at that same level. 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 and distortion. Based on measurements of the transmission, it might therefore be necessary to decrease the limiter level in front of the codec system, for example if relatively low bitrates are used which in general cause a rise of overshoots. On high bitrates, a higher limiter level might be possible. Grey lines emphasize the alignment relationship at levels that have a particular meaning:

- −12 dBTP corresponds with the reference level as specified in CENELEC EN 50049. 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.

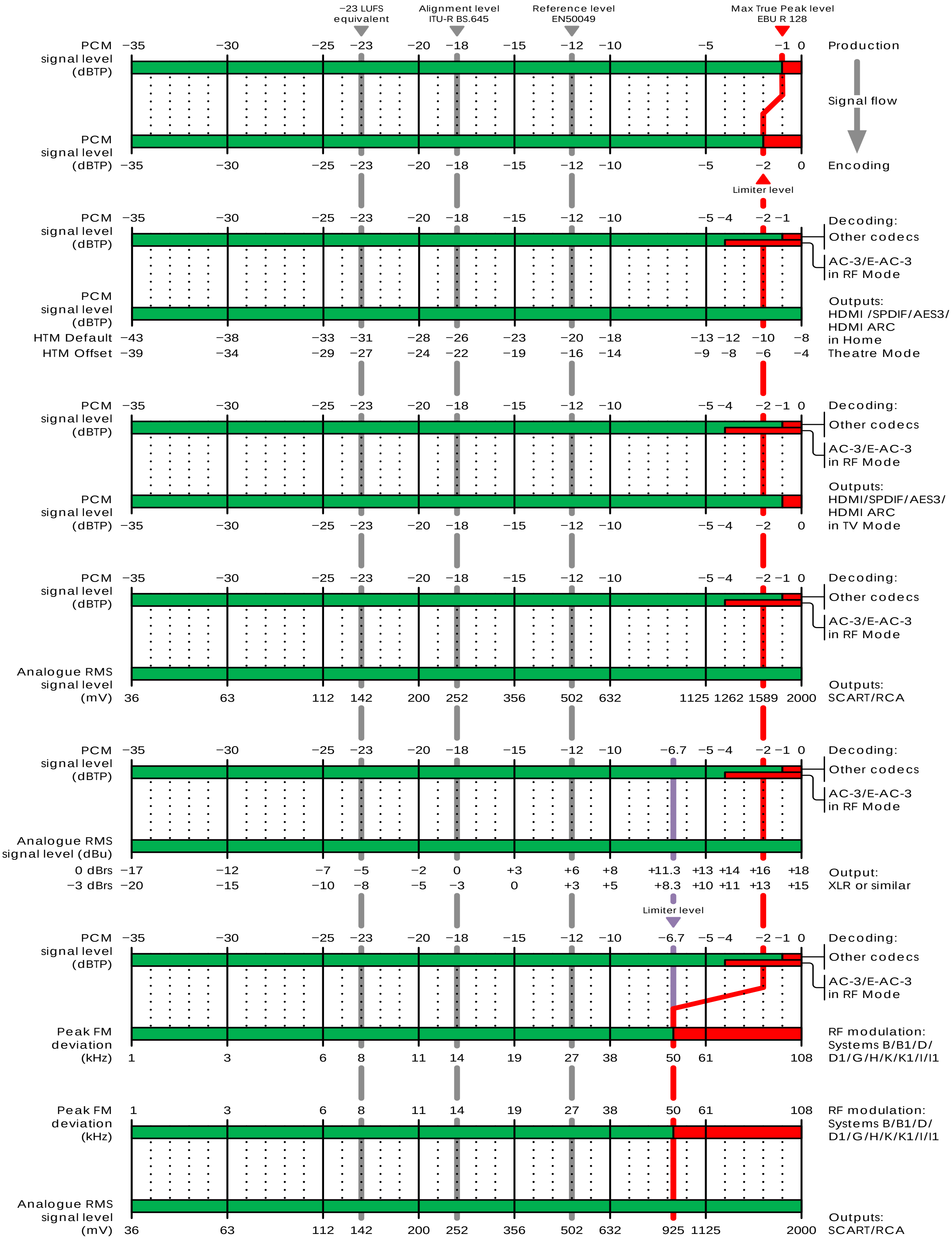
The following table describes the signal level bars from top to bottom:

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).
Encoding level	This is the digital signal level measured in dBTP of the encoded audio using one of the codecs mentioned in this document, with usually applied bitrates. 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 −2 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. On high bitrates, a higher limiter level might be possible.
Decoding level	This is the digital signal level measured in dBTP of the decoded audio using one of the codecs mentioned in this document. 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

Subject	Description
	compare the measured levels with the specified maximum levels to check if it is necessary to decrease the limiter level. For the AC-3/E-AC-3 decoder in RF Mode, different maximum peak levels are shown. Due to the internal loudness reference level of -20 LUFS of the AC-3/E-AC-3 decoder being decreased to -23 LUFS by 3 dB, the actual clipping level using a 1 kHz sine wave is -3 dBTP. The maximum recommended <u>output</u> level of the AC-3/E-AC-3 RF Mode decoder for programme material is therefore indicated as -4 dBTP (1 dB lower). The recommended studio limiter level of −2 dBTP exceeds that maximum threshold. Nevertheless this is not an error. As the AC-3/E-AC-3 system contains its own internal overload protection, the maximum peak level for encoding does not have to be decreased.
Outputs: HDMI/SPDIF/AES3 HDMI ARC/ in Home Theatre Mode	This is the digital PCM signal level of the decoded audio, measured in dBTP at the HDMI, the HDMI ARC (Audio Return Channel), the SPDIF or the AES3 interface if the IRD, IDTV or Media Player operates in Home Theatre Mode (§ 6.5 can be checked for details). There are two rows displaying the output levels, 'HTM Default' or 'HTM Offset' (see § 6.7 for more information). Support for the offset level is optional.
Outputs: HDMI/SPDIF/AES3/ HDMI ARC/ in TV Mode	This is the digital PCM signal level of the decoded audio, measured in dBTP at the HDMI, the HDMI ARC, the SPDIF or the AES3 interface if the IRD, IDTV or Media Player operates in TV Mode (§ 6.5 can be checked 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 (-7 dBTP is a practical setting as limiters can suffer from some amount of overshoot). 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 § 4.2, it could be necessary to decrease the limiter level.
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 purple line indicates the alignment for the limiter level. The limiter could also be built into the modulator itself. Because of reasons described in § 4.2, it could be necessary to decrease the limiter level.
RF Modulation: NICAM 728	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

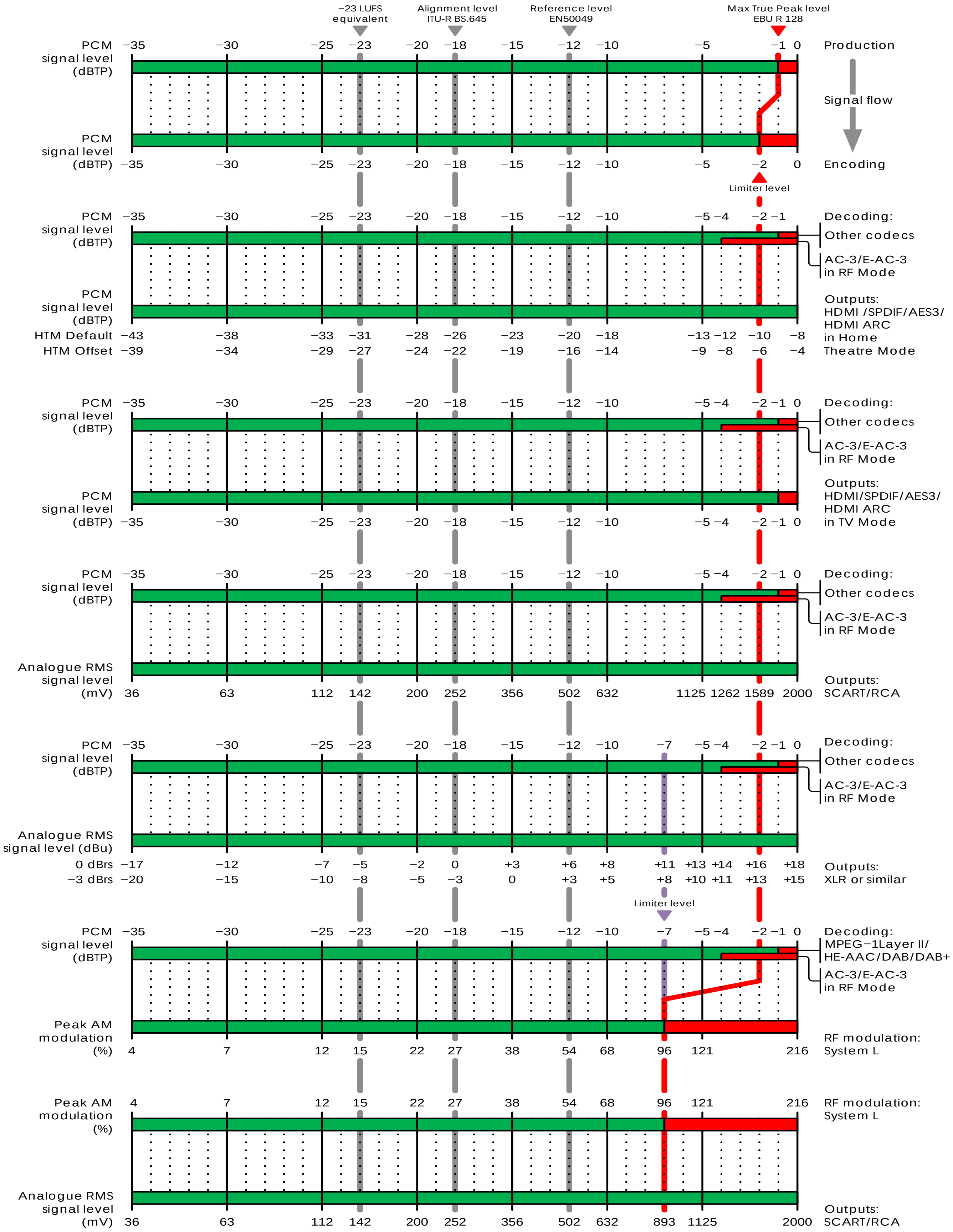
Subject	Description
systems B, B1, D1, G, H, K1 and L	connected to the input of the NICAM RF modulator. The red line represents the maximum peak level. In practice, this means that the signal shall be passed through an end-stage pre-emphasis peak limiter set to a level of -2 dBTP before being applied to the modulator. The purple line indicates the alignment for the limiter level. The limiter could also be built into the modulator itself. Because of reasons described in § 4.2, 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 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 -9.7 dBTP before being applied to the modulator (-10 dBTP is a practical setting as limiters can suffer from some amount of overshoot). 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 § 4.2, 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 -8.5 dBTP before being applied to the modulator (-9 dBTP is a practical setting as limiters can suffer from some amount of overshoot). 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 § 4.2, it could be necessary to decrease the limiter level. Any added signal in the spectrum 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.
Outputs: SCART or RCA interface	This is the analogue RMS signal level, measured in millivolt, 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 or DAB/DAB+ receiver.

A.2 Level alignment in systems B, B1, D, D1, G, H, K, K1, I and I1



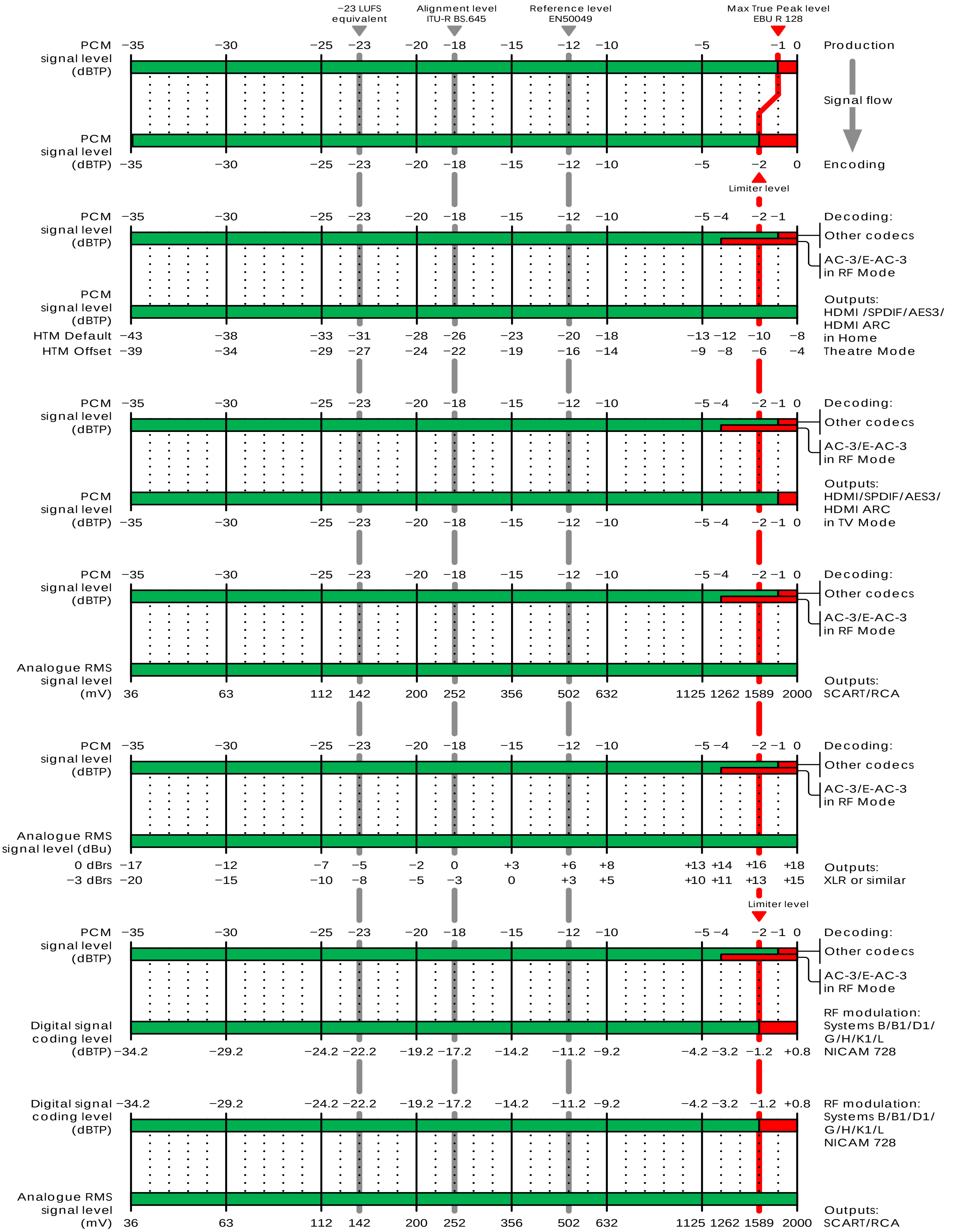
Conditions: 1 kHz sine wave in phase on Left and Right channel only, Dialnorm = −23, Mode = RF, DRC = None, PRL = −23, TL = −23

A.3 Level alignment in television system L



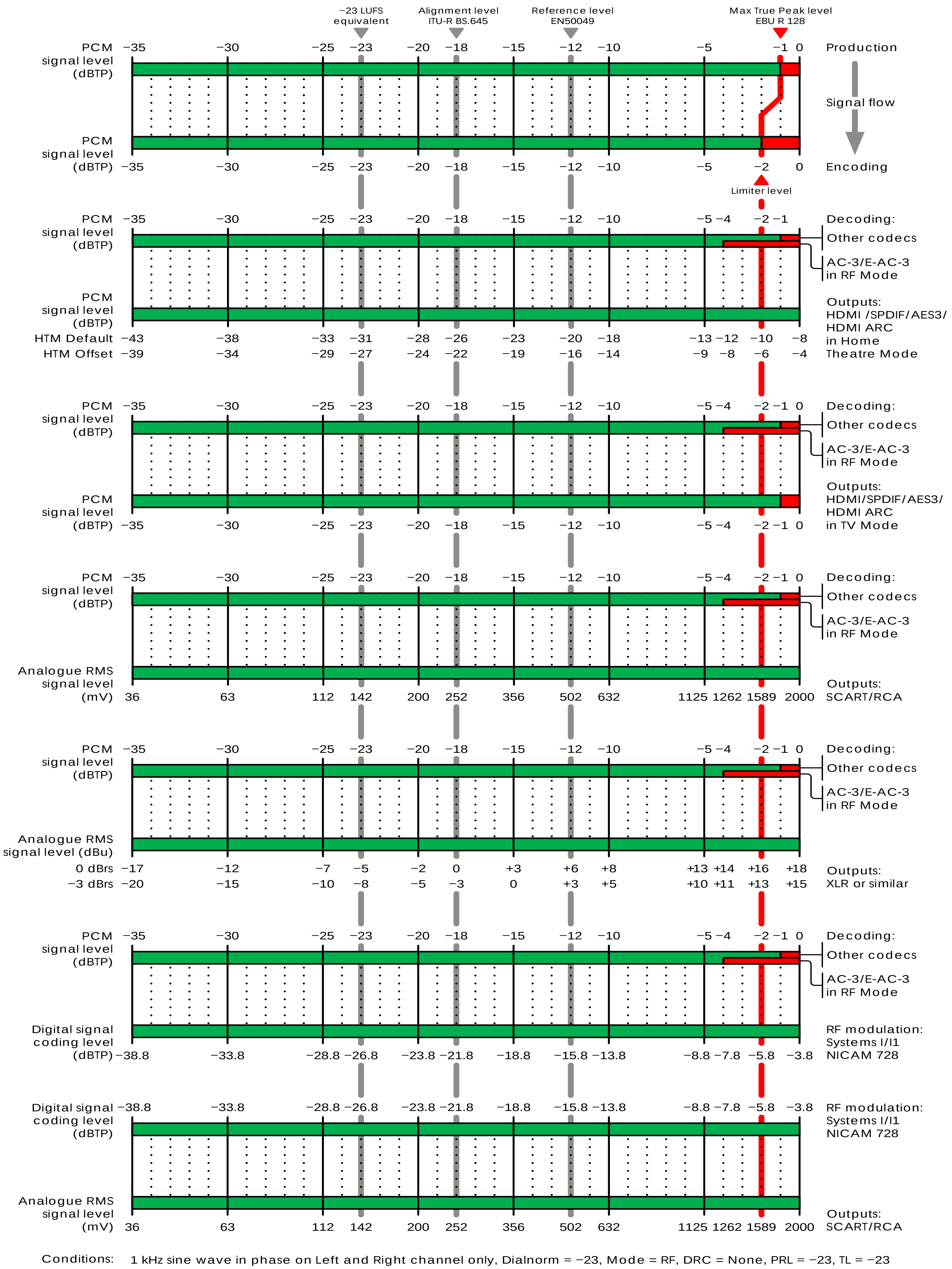
Conditions: 1 kHz sine wave in phase on Left and Right channel only, Dialnorm = −23, Mode = RF, DRC = None, PRL = −23, TL = −23

A.4 Level alignment in NICAM television systems B, B1, D1, G, H, K1 and L

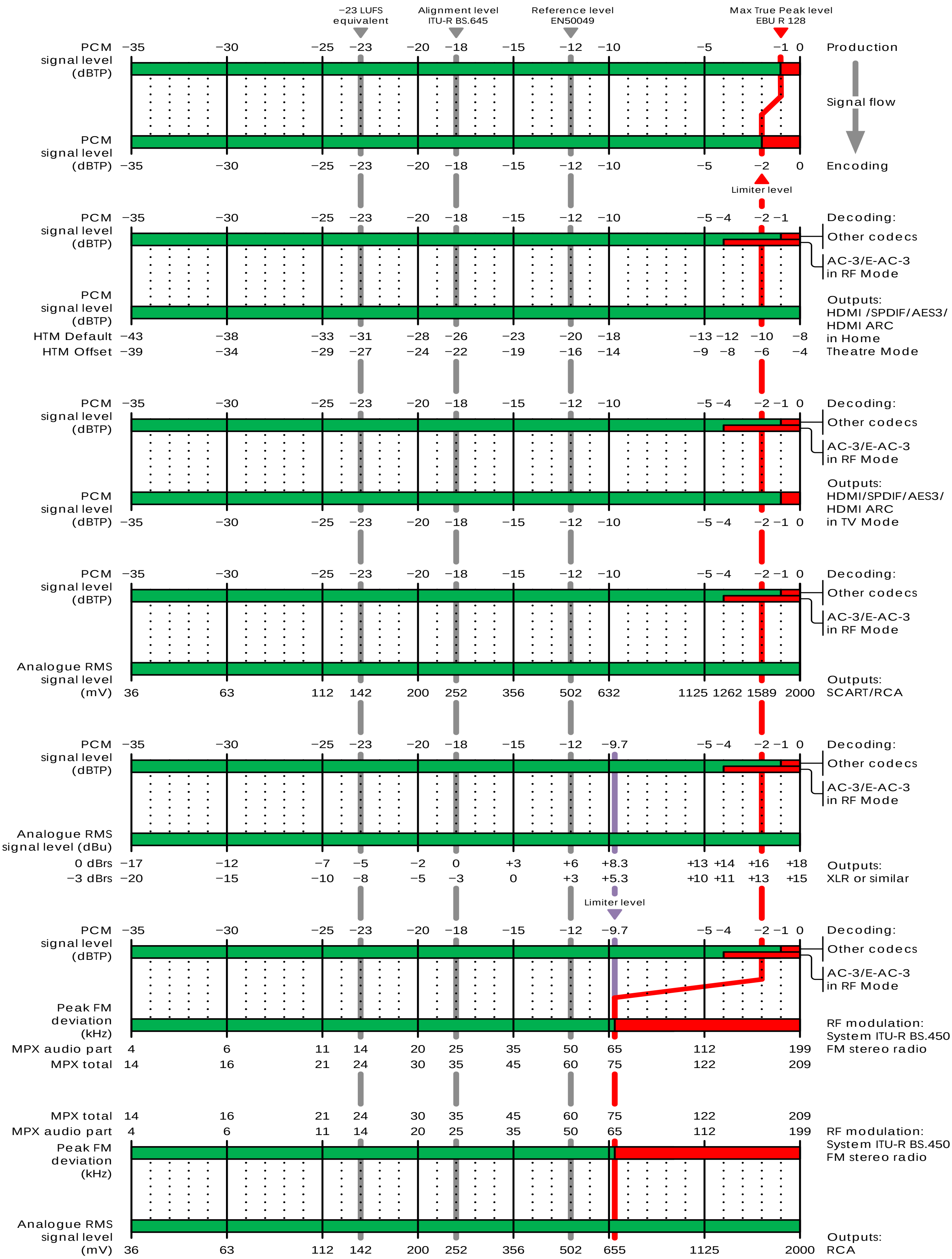


Conditions: 1 kHz sine wave in phase on Left and Right channel only, Dialnorm = −23, Mode = RF, DRC = None, PRL = −23, TL = −23

A.5 Level alignment in NICAM television system I and I1

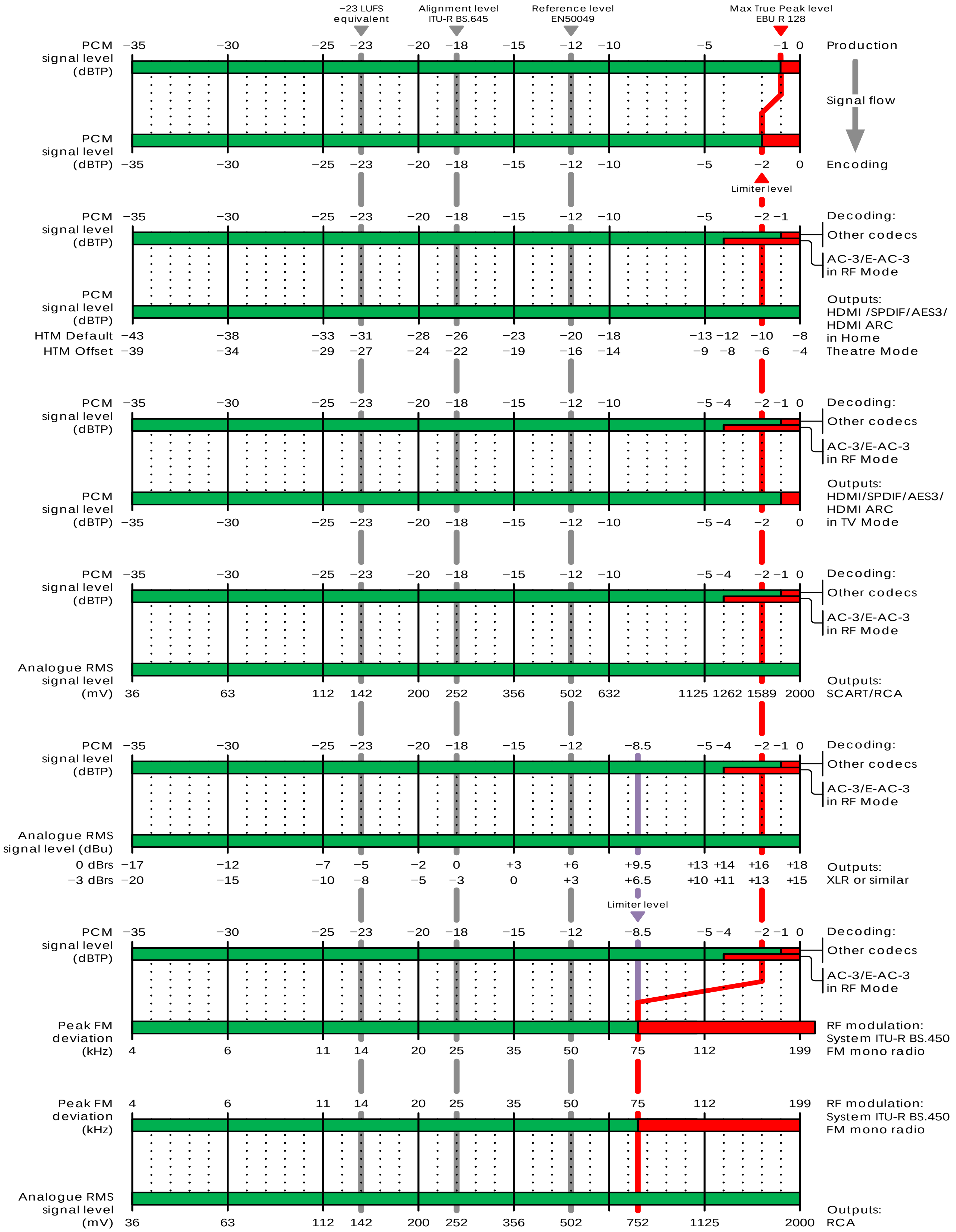


A.6 Level alignment in FM stereo radio



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

A.7 Level alignment in FM mono radio



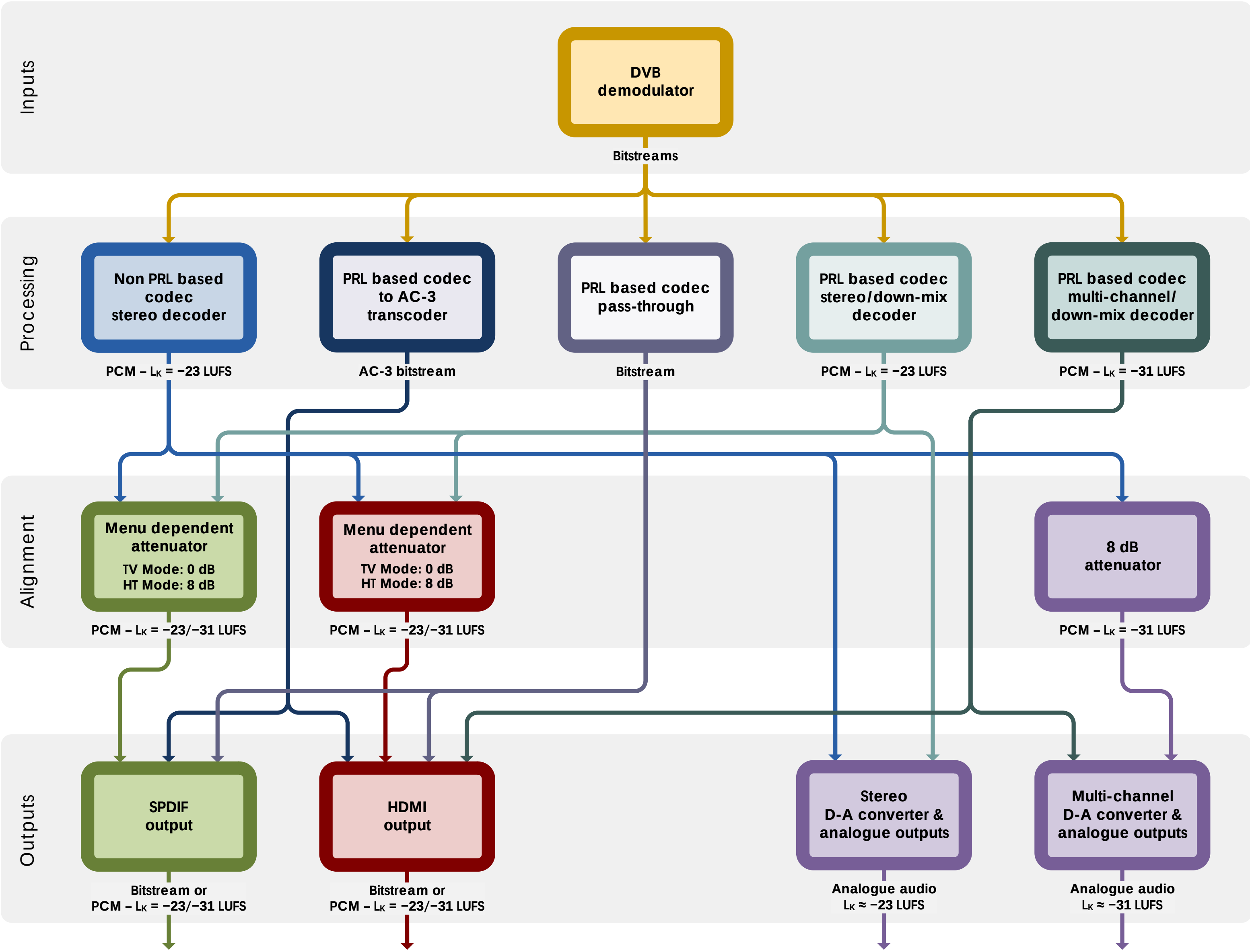
Conditions: 1 kHz sine wave in phase on Left and Right channel only, Dialnorm = −23, Mode = RF, DRC = None, PRL = −23, TL = −23

Annex B: Graphical representation of audio processing within the device

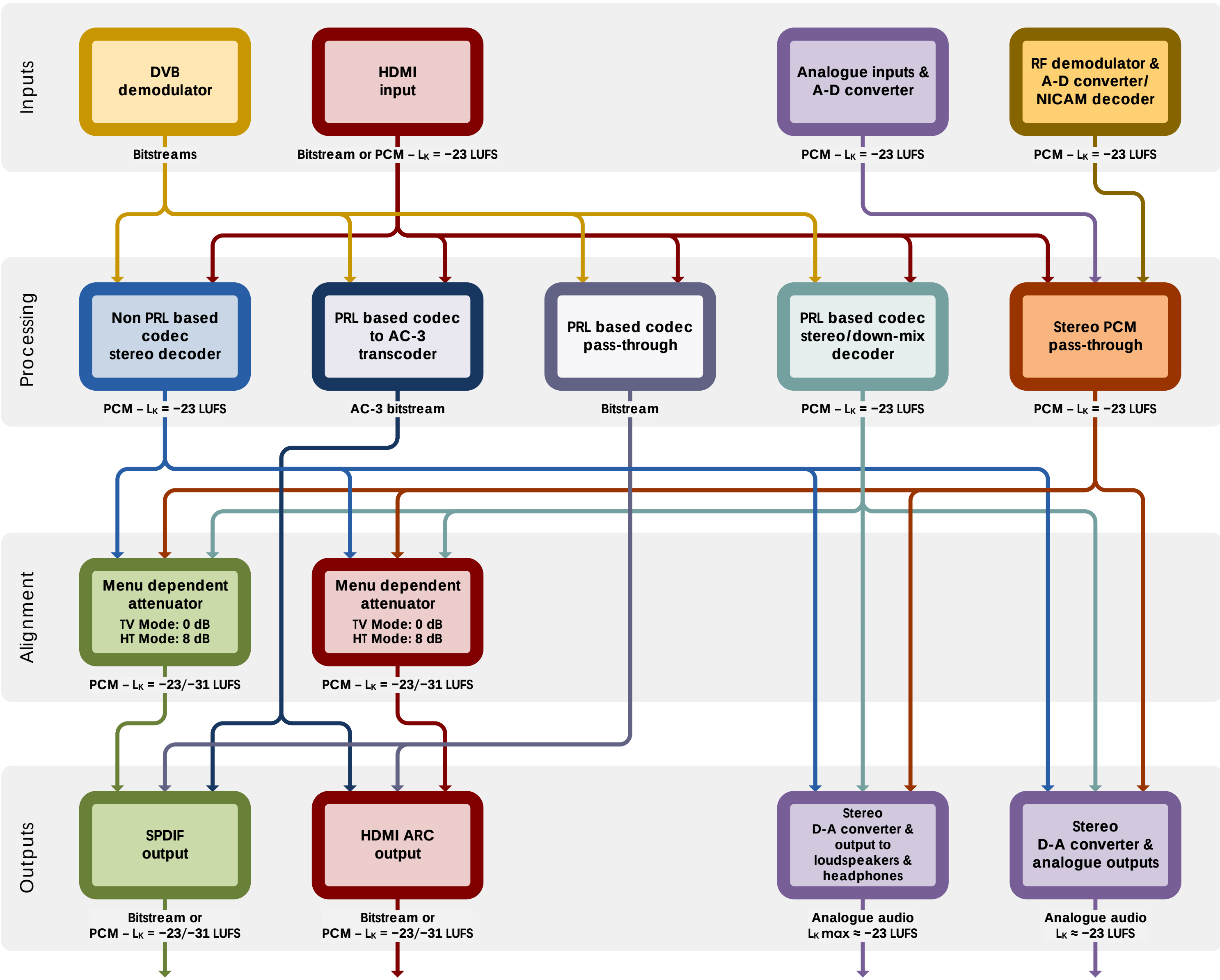
B.1 Clarification

The following sections show a graphical representation of the audio processing and level alignment within the device. Its application to devices which operate in a different transmission system, which offer fewer or more input and output interfaces, decoders and features, or which offer Audio Description by using two decoders, can be derived from this figure. The block 'Menu dependent attenuator' in these figures applies reduction of the PCM audio level depending on the user setting for the connected device as described in § 6.5.

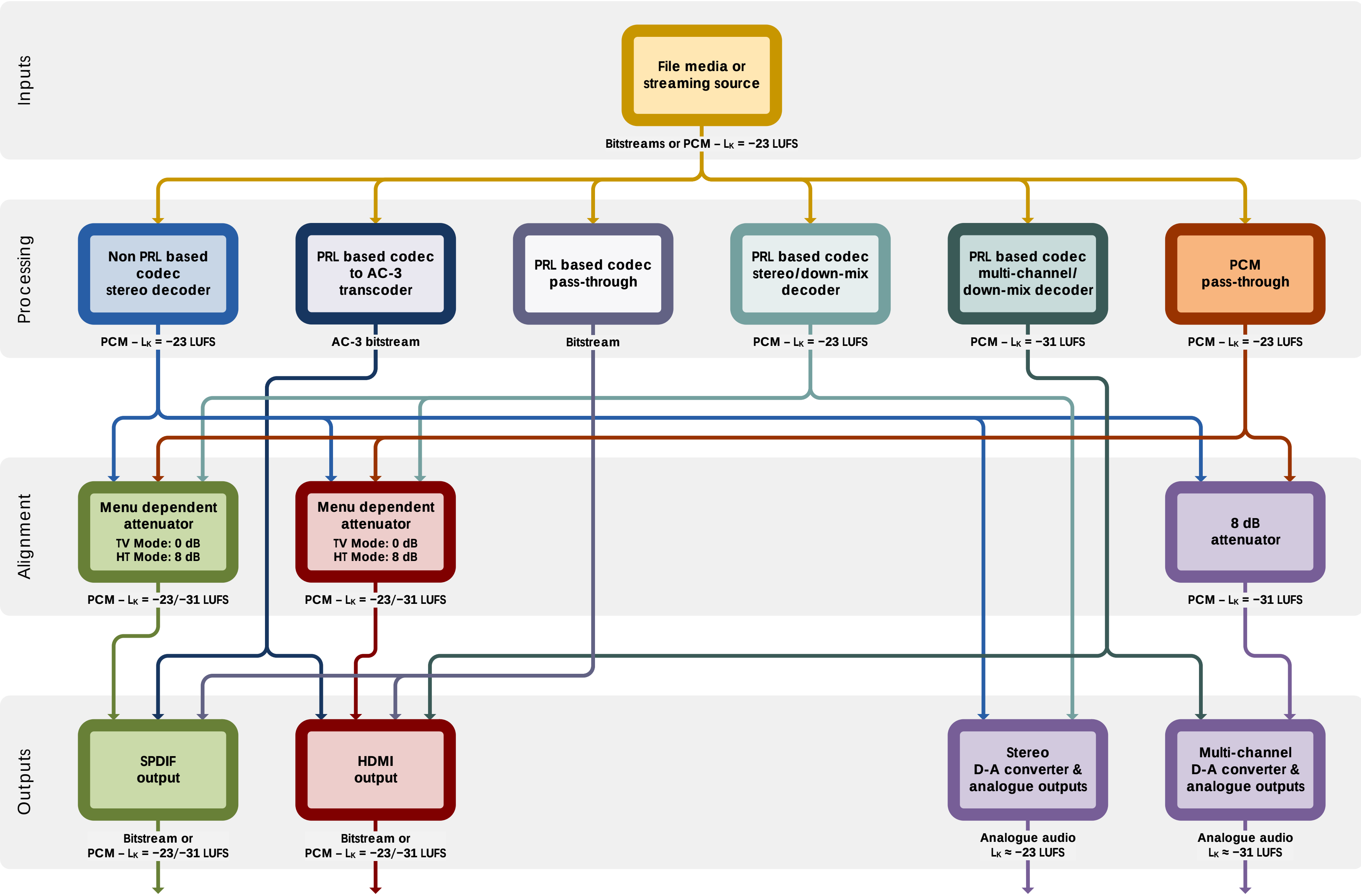
B.2 Level alignment in the IRD



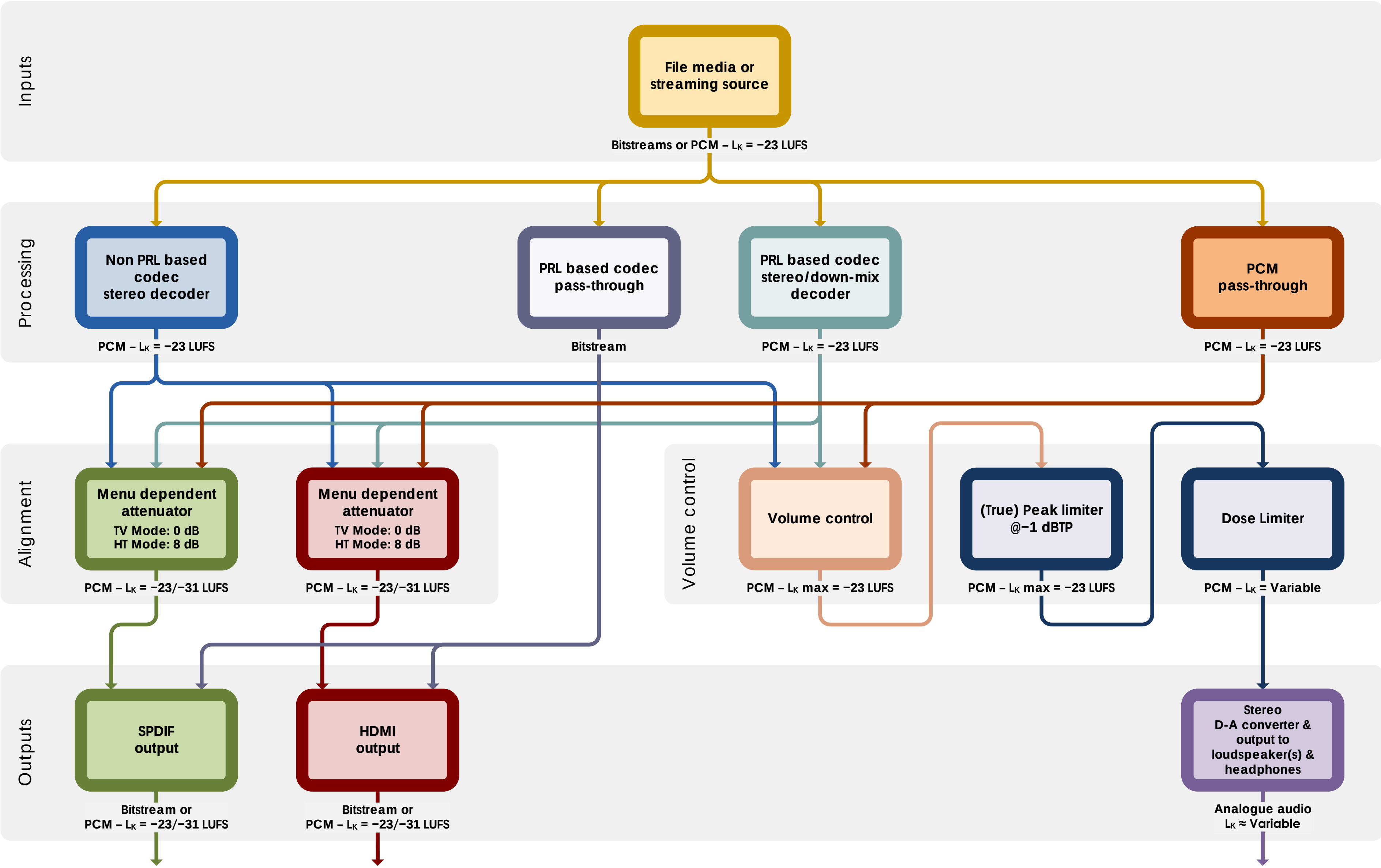
B.3 Level alignment in the IDTV



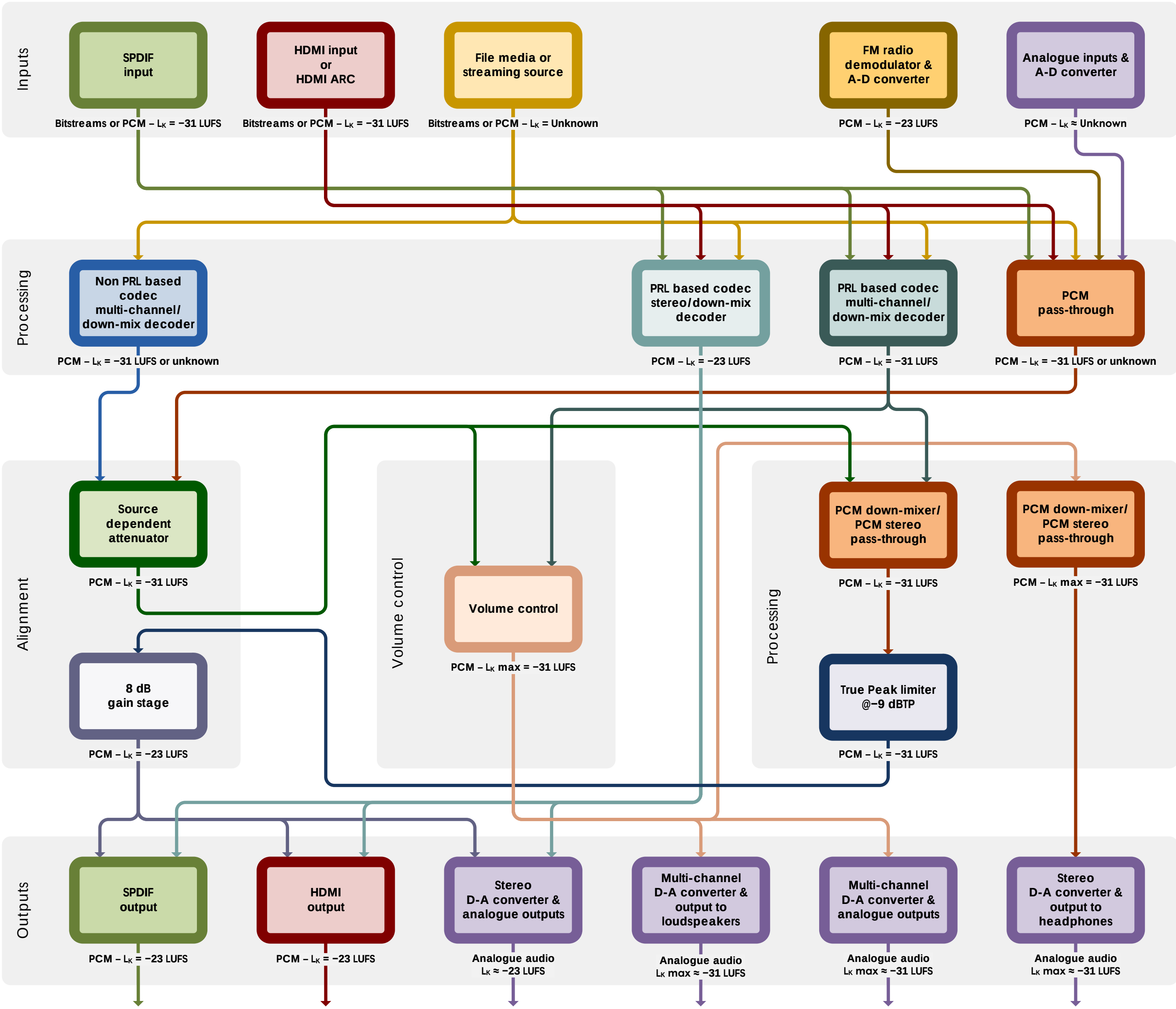
B.4 Level alignment in the Media Player



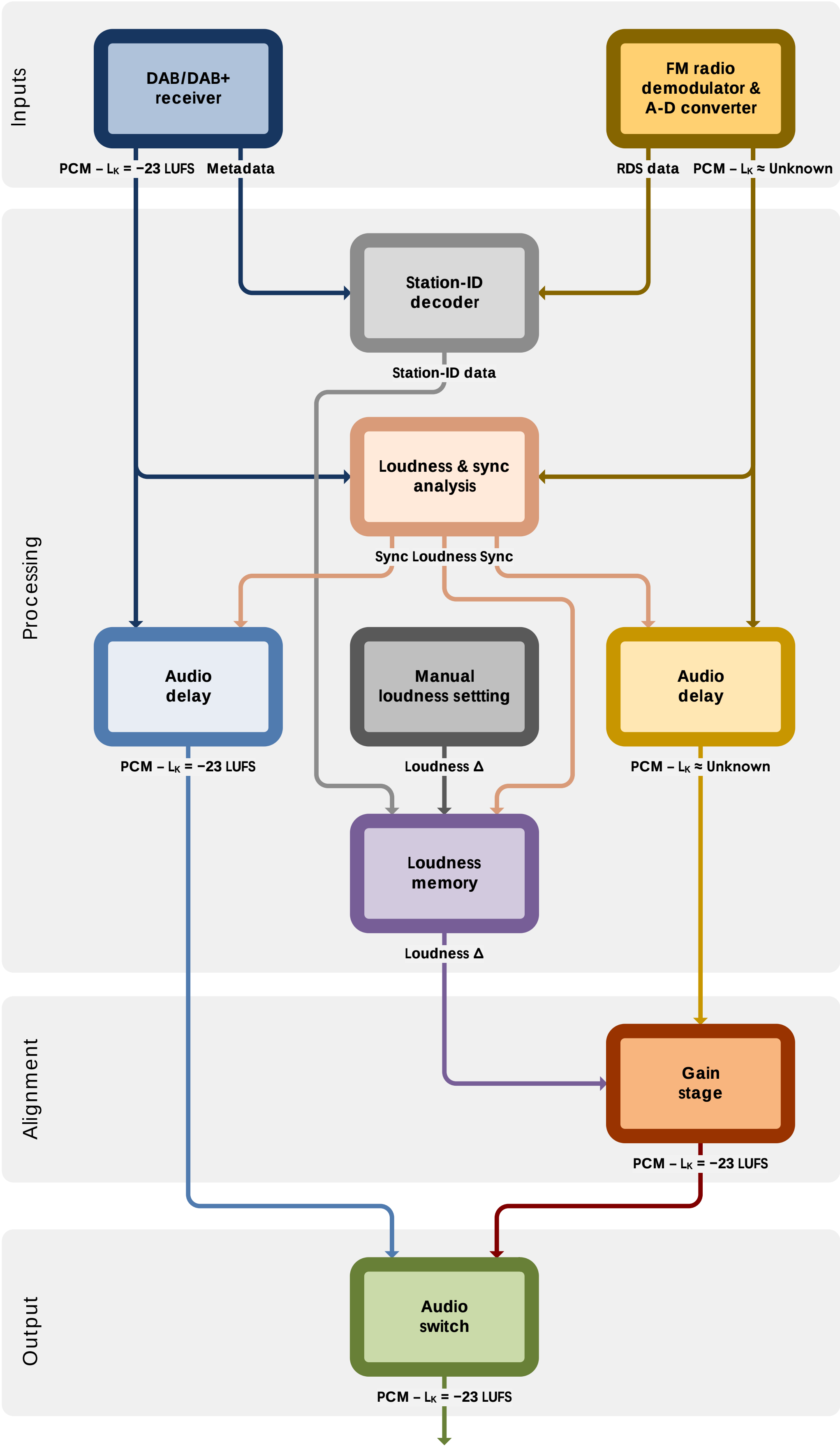
B.5 Level alignment in the Personal Music Player



B.6 Level alignment in Home Theatre Equipment



B.7 Level alignment in combined DAB/DAB+ and FM radio receivers



Annex C: References and Abbreviations

C.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 (2014)	Loudness normalisation and permitted maximum level of audio signals.
[2]	ITU-R BS.450-3 (2001)	Transmission standards for FM sound broadcasting at VHF.
[3]	ITU-R BS.707-5 (2005)	Transmission of multi-sound in terrestrial television systems.
[4]	ITU-R BT.2043 (2004)	Analogue television systems currently in use throughout the world.
[5]	ITU-R BS.642-1 (1990)	Limiters for high quality sound programme signals.
[6]	ITU-R BS.412-9 (1998)	Planning standards for FM sound broadcasting at VHF.
[7]	CENELEC EN 50049 (2000)	Domestic and similar electronic equipment interconnection requirements: Peritelevision connector.
[8]	ITU-T J.17 (1988)	Pre-emphasis used on sound-programme circuits.
[9]	ITU-R BS.1770-4 (2015)	Algorithms to measure audio programme loudness and true-peak audio level.
[10]	ITU-R BS.645-2 (1992)	Test signals and metering to be used on international sound programme connections.
[11]	IEC EN 60728-5 (2008)	Cable network equipment for television signals, sound signals and interactive services — Head-end equipment.
[12]	ISO/IEC 11172-3 (1993)	Information technology — Coding of moving pictures and associated audio for digital storage media at up to about 1.5 Mbit/s — Part 3: Audio.
[13]	ETSI TS 102 366 v1.3.1 (2014)	Digital Audio Compression (AC-3, Enhanced AC-3) Standard.
[14]	ETSI TS 103 190-1 v1.2.1 (2015)	Digital Audio Compression (AC-4) Standard, Part 1: Channel based coding.
[15]	ISO/IEC 14496-3 (2009)	Information technology — Coding of audio-visual objects — Part 3: Audio.
[16]	ISO/IEC 23008-3 (2015)	Information technology — High efficiency coding and media delivery in heterogeneous environments — Part 3: 3D audio.
[17]	ETSI TS 103 176 v1.1.2 (2013)	Digital Audio Broadcasting (DAB); Rules of implementation; Service information features.
[18]	ETSI TS 101 154 v2.2.1 (2015)	Specification for the use of Video and Audio Coding in Broadcasting Applications based on the MPEG-2 TS.

C.2. List of abbreviations

AAC	Advanced Audio Coding
AC-3	Audio Coding 3 (also known as Dolby Digital)
AC-4	Audio Coding 4 (Dolby next generation universal audio format)
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 electro technical standardisation)
Codec	Encoder and decoder system (coder/decoder)
DAB	Digital Audio Broadcasting
DAB+	DAB using the AAC codec
dB	decibel
dBTP	decibel True Peak
DRC	Dynamic Range Control
DTS	Digital Theatre Systems (refers both to DTS Inc. and to a family of coding standards)
DVB	Digital Video Broadcasting
E-AC-3	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
MPX	Multiplex
PCM	Pulse Code Modulation

PRL	Programme Reference Level
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

C.3. Further reading

EBU Tech Doc 3341 (2016)	Loudness Metering: 'EBU Mode' metering to supplement EBU R 128 loudness normalisation.
EBU Tech Doc 3342 (2016)	Loudness Range: A measure to supplement EBU R 128 loudness normalisation.
EBU Tech Doc 3343 (2016)	Guidelines for Production of Programmes in accordance with EBU R 128.