# REPORT ITU-R BS.2054

# Audio levels and loudness

(2005)

## 1 Introduction

This Report describes advice for the broadcasting of television programmes which include pre-recorded television advertisements (commercials) with particular reference to audio levels and loudness. It considers the several processes of studio production, recording onto storage media, transport of the media, and broadcast via a television presentation and transmission system. This descriptive material is provided as guidance. It describes one administration's approach to dealing with the ingest and transmission of television soundtracks, in particular the factors which contribute to loudness.

Question ITU-R 2/6 *decides* the following Questions should be studied:

**1** What audio metering characteristics should be used to provide an accurate indication of signal level in order to assist the operator to avoid overload of digital media?

**2** What audio metering characteristics should be used to provide an accurate indication of subjective programme loudness?

While the work of the Rapporteur on Level Metering within Working Party 6P is progressing, some administrations are providing interim measures to address audio levels and loudness. For instance, Australian television broadcasters have established a common alignment level, –20 dBFS, in accordance with SMPTE RP 155. Guidelines have been introduced specifying that volume compression where used after the final mix of a television commercial soundtrack, be restricted to a slope of 2:1 with an onset point of –12 dBFS.

## 2 Simulcasting of analogue and MPEG-1 layer II digital audio

Many administrations are currently progressing through or planning a transition to digital broadcasting. During the transition a requirement exists for simultaneous broadcasting in analogue and digital form.

The parameters of the analogue and digital broadcasting systems have different limits. These limits determine the operation of audio processors and, consequently, have an impact on the potential loudness of the sound.

## 3 Loudness considerations

Within most broadcasting systems television broadcasters transmit material of varying programme genres contiguously and interspersed with inserted material, including advertisements. There is a potential for variations in the perceived loudness of adjacent audio segments.

The factors contributing to perceived loudness are complex but the correct alignment of audio levels through the various stages of production and transmission and the careful management of dynamic range and spectral content are key factors in preventing extreme variations in loudness.

## 4 Operating in the analogue domain

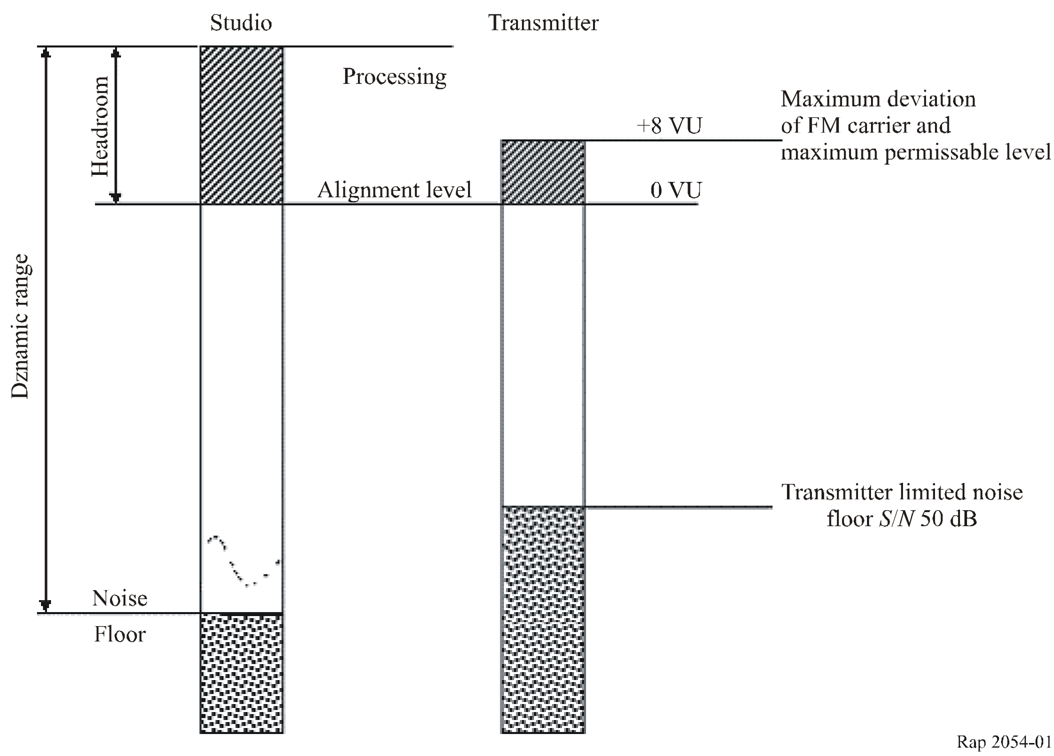
In the analogue domain, the amount of headroom available in recording devices is limited by the amount of distortion that may occur at high levels of audio.

Magnetic recordings will be limited by the integrity of the magnetic image at high levels of coercion.

The analogue television transmission system is limited by the allowable deviation of the FM sound carrier.

Figure 1 describes the parameters and limits of the analogue television transmission audio system as adopted in Australia.

FIGURE 1




The lower limit of the analogue studio audio system is determined by the level of the system noise. Practically, this is maintained at least 50 dB below the alignment level providing some 70 dB of dynamic range. Although a studio analogue system and recording devices may provide adequate headroom for dynamic production, the limiting factor in an analogue broadcasting chain is the allowable deviation of the FM transmitter and broadcasters usually apply some limiting of the audio level at the transmitter input to avoid over deviation.

It is necessary to define the nature of peak level. Peak level in this context and as used throughout this document, really means what is commonly called "quasi peak" levels – the levels as measured by a PPM type meter having (typically) an integration time of 10 ms.

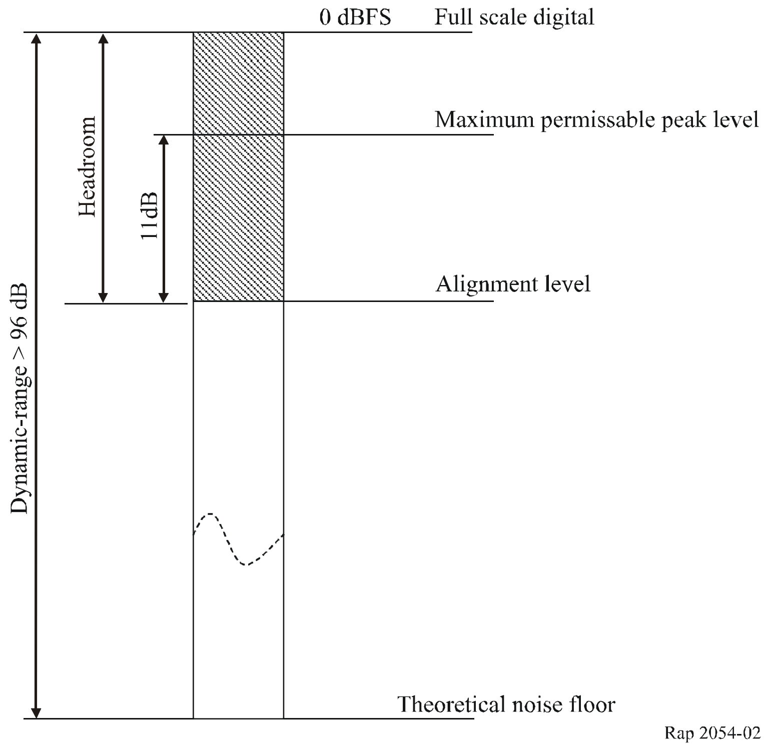
## 5 Operating in the digital domain

Where production, post-production, switching and mixing of television advertisements is carried out in the digital domain, the digital audio system is aligned using a reference signal 20 dB below full-scale digital "0 dBFS" (i.e. 20 dB below the level at which digital clipping commences).

The alignment level –20 dBFS is usually equated to zero on the station's VU meters or "4" or "TEST" on PPM metres.

Figure 2 describes the parameters and limits of the digital audio system as adopted by Australia.

FIGURE 2




In a 16-bit digital audio system where there is a theoretical dynamic-range of some 96 dB, if the average audio level is zero VU, there will be 20 dB of headroom before digital clipping occurs. The audio peaks should not normally exceed a level 11 dB higher than the alignment level.

## 6 Use of audio meters

### 6.1 VU meters

Standard VU (Volume Unit) meters are commonly used in television facilities in many administrations to adjust audio levels during the recording, playback and transmission of audio material. A VU meter will indicate an "average" audio level, but cannot indicate the instantaneous level of audio peaks. The standard VU meter has an integration time of 300 ms and thus its "average" reading relates to that particular integration time.

A VU meter scale is not capable of directly representing the full dynamic range of audio signals in analogue or digital systems but VU meters provide a convenient means of ascertaining that the audio level is within normal parameters relative to the alignment level and they provide some limited indication of loudness.

It should be noted that in production and broadcasting establishments there is a wide range of audio level indicators including LED and on-screen devices commonly referred to as "VU meters". Although most audio level indicators will return consistent measurements under steady state conditions such as that produced by a sinusoidal alignment tone, there may be inconsistencies between instruments where they are used to measure programme levels in dynamic material.

### 6.2 Peak meters

Where meters such as Peak Programme Meters (PPMs) are used, they indicate peak levels more accurately than the VU meter because their internal time constants are optimized for such measurement.

Since Peak Programme Meters are somewhat better indicators of peak audio level they are more useful in the management of system headroom but are of limited use in assessing loudness.

### 6.3 Loudness meters

The development of new meters capable of measuring sound energy in a way that accurately correlates to the human perception of loudness continues.

It is envisaged that such meters will eventually provide producers and broadcasters with an objective means of comparing the loudness of adjacent audio segments.

## 7 Harmonization of audio alignment levels for digital programme exchange – Adoption of SMPTE RP 155

Some television broadcasters have adopted SMPTE RP 155 audio levels for digital audio records on digital television tape recorders.

For television recordings a sinusoidal steady state tone at 1 kHz representing the alignment level of –20 dBFS should precede programme material presented for broadcast. This level is usually equated to zero (zero VU) on the station's VU type audio level meters and is used to align the broadcaster's recording and transmission equipment to the same reference level as the originating equipment.

When measured with a VU type meter, the normal audio level of the programme material that follows the alignment signal should be approximately zero VU.

When measured with a PPM or digital equivalent type meter, the normal audio level of programme material will be indicated typically in the range of +8 to +11 dB above alignment level.

## 8 Peak audio level

In the digital domain audio peaks should not exceed –9 dBFS, i.e. peak excursions should not be more than 11 dB above the alignment level. It must be understood that +11 dB in this context is not a deliberate aim point for production levels, but is a technical limit to be observed. This limit will help ensure that short-duration peaks do not reach 0 dBFS (full scale).

In an analogue transmission system, audio peaks should not exceed the alignment level by more than 8 dB.

These levels are recommended for optimum use of the available headroom in the analogue and digital systems.

## 9 The studio environment

Studio analogue sound systems are capable of mixing, recording and reproducing material with dynamic ranges extending from the level of the audible system noise to the level at which distortion is unacceptable. For practical purposes, this represents a dynamic range of some 70 dB.

Studio digital sound systems typically operate with dynamic ranges of more than 90 dB. The lower limit in a digital audio system is determined by the theoretical digital noise floor where there is no meaningful data. This lower limit is principally determined by the audio word length (16, 18, 20, 24 bits).

The upper limit in a digital audio system is defined as the full-scale digital level, 0 dBFS. At that point, digital clipping occurs because the audio signal cannot be adequately represented by the number of data bits available.

Using a VU metering system, programme audio material should be recorded such that the normal programme level is around zero VU with occasional louder passages allowed to exceed this level by 2 or 3 dB (+3 dB being the limit on most VU meters). In a normal broadcast audio mix (not a proprietary multichannel surround mix), the dialogue level will typically fall at around –2 to –3 dB below the alignment level. For significantly processed material such as commercials or pop music the VU meter reading should not be permitted to exceed zero VU.

It is common to employ some form of audio processing during the production phase to compress the dynamic range of the material so that the loudest and softest passages of the material can be enjoyed without the need to adjust the receiver sound controls. For example, soundtracks which are mixed for a cinema environment will require some compression for comfortable listening in a home environment.

The consequence of compression is reduction of the ratio between the peak and “average” level of the content. Increasing the “average” level will increase the apparent loudness. The human ear tends to be more sensitive to frequencies in the mid range and if these frequencies are artificially boosted, then again the apparent loudness will increase. The use of audio processing must be judicious so that the compression of the dynamic range of the soundtrack plus any other processing employed does not produce excessively loud or strident material.

Soundtrack production studios often employ gates to attenuate or eliminate the sounds below a lower threshold, and peak limiters to prevent audio exceeding the level that causes distortion, or digital clipping. These devices should not be used for the purpose of increasing the relative loudness of the material.

## 10 Application of volume compression in post-production following the final mix of a television commercial soundtrack

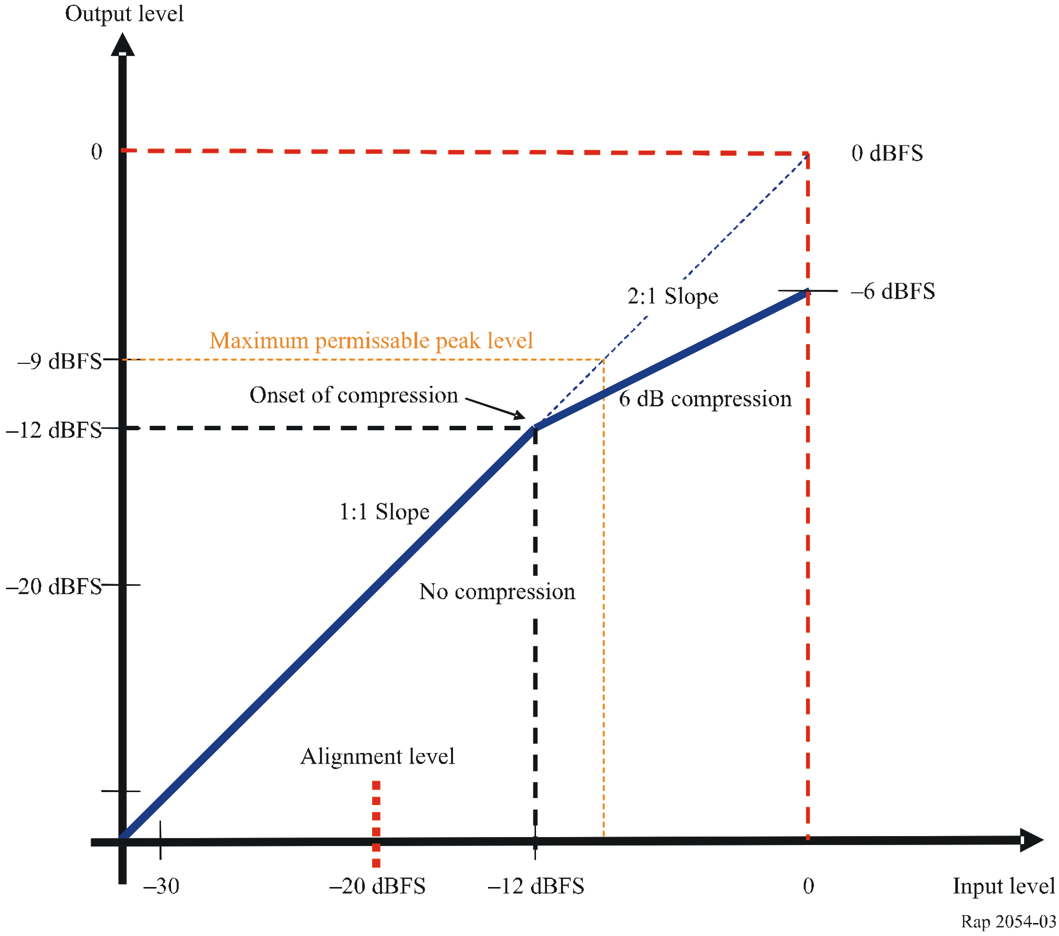
Australian free-to-air commercial television broadcasters have introduced guidelines specifying that volume compression should, where used after the final mix, be restricted to a slope of 2:1 with an onset point of –12 dBFS.

Figure 3 provides a diagrammatic representation of this simple profile. In this profile, an onset of compression at –12 dBFS allows for gentle compression of the upper 3 dB of the signal before reaching the maximum permissible peak level. If any further peak limiting were to be necessary, it would be provided automatically by the broadcaster’s transmission processor.

The elements of a soundtrack, namely dialogue, music and effects are subject to various processes during production. Where these elements sit in the final soundtrack, with respect to audio levels and loudness, is the result of a final mix and effectively it is here that the loudness of the soundtrack will be principally influenced.

The processes that take place during the production of a soundtrack are not the only contributors to an increase in the loudness characteristic of the sound. It is in the final mastering of the track, where the “relativity” of sound characteristics of each component is defined, that loudness is most affected.

FIGURE 3




Material that has been compressed may sound louder, even though there is no increase in peak level. This is because compression of a soundtrack may raise the energy content of the sound by reducing the dynamic range (i.e. the difference between the loudest and softest levels of the sound) thereby making it more dense.

Many modern processors are not calibrated in dB, have constantly varying compression ratios and are likely to be multiband devices which apply different amounts of compression in different frequency bands. This makes it difficult for soundtrack producers to measure and quantify how much compression is applied to a soundtrack.

## 11 Ingest of soundtracks into the television broadcasting chain

As noted previously, audio material delivered for transmission should be preceded by a sinusoidal audio alignment signal of 1 kHz at a level of –20 dBFS. The receiving station will align its systems to that signal so that it is equivalent to 0 VU, i.e. the level 20 dB below the point of digital clipping in the broadcasting station’s audio system. Where PPMs are in use, this level is usually equivalent to “4” or the “TEST” level. A VU meter aligned to this reference level should read programme material in accordance with section 9 depending on the type of sounds and dynamic range of the material.

As far as it is practicable, all stages of the broadcasting system should have unity gain and operate at the recommended levels for optimum headroom. It is intended that material provided to broadcasters should not require any level adjustment other than aligning the reference signal on the material to the broadcaster’s zero reference.

In the analogue transmission chain, broadcasters must limit the extent of audio peaks to ensure that the FM sound carriers are not deviated beyond the allowable limit of 50 kHz.

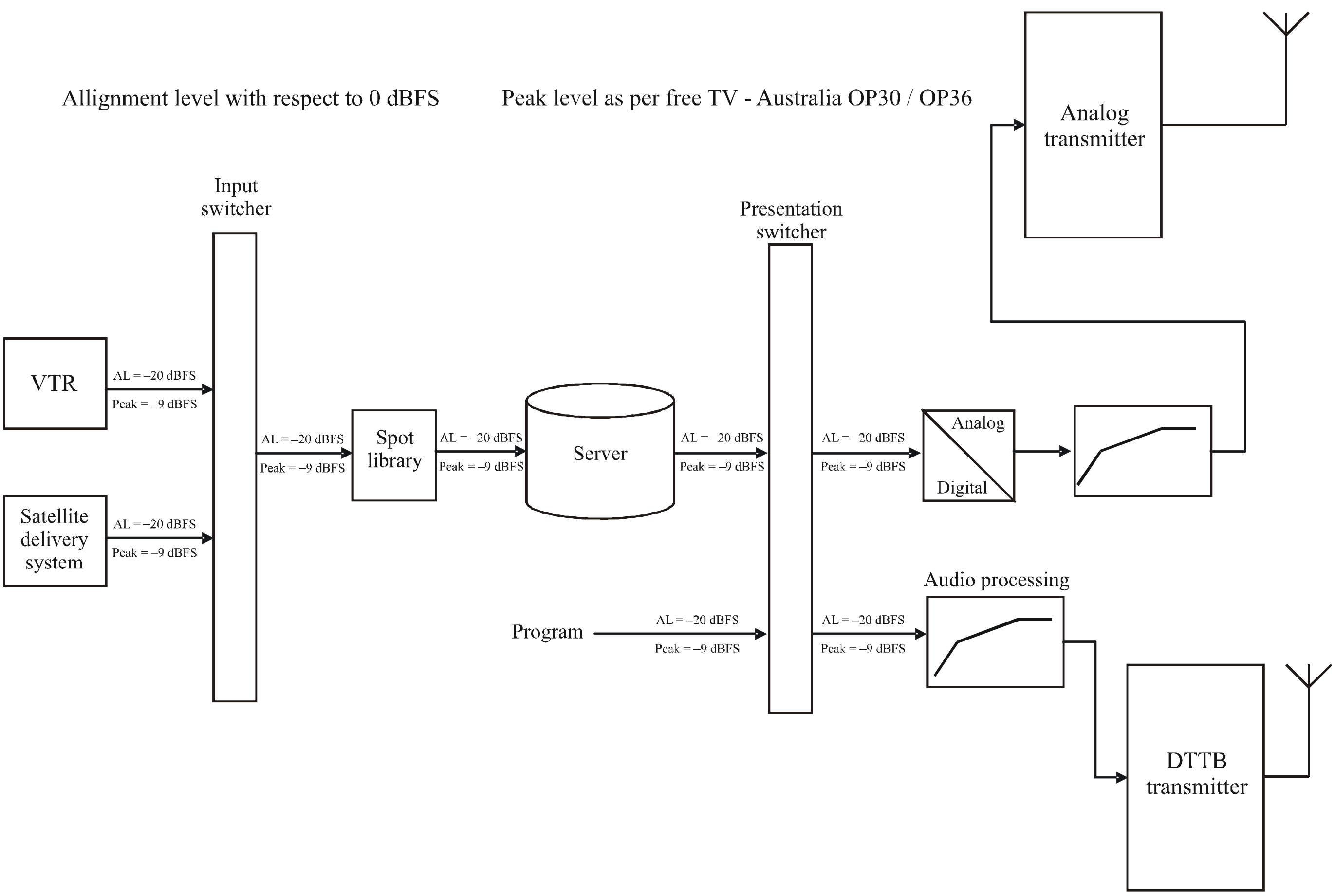
Broadcasters may also compress the dynamic range of the audio signal at the broadcast station output in the digital transmission chain, to ensure that the audio levels are consistent and that listeners can enjoy the softest and loudest passages of sound without having to adjust their volume controls beyond a comfortable setting.

Where broadcasters use an audio-processing system, it is strongly recommended that it provide the following functions:

– automatic gain control (AGC);
– single or multiband compressor;
– capacity to adjust the attack and release time of the compressor;
– limiter (matched to the transmitter pre-emphasis in analogue transmission);
– adjustments to limit the range of AGC and compressor action to limit the gain applied to low-level passages; and
– ability to modify the action of the AGC and compressor to match future loudness measurement and control algorithms.

Figure 4 provides an Australian depiction of a typical television broadcasting station audio transmission chain.

FIGURE 4

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## 12 Summary

Some free-to-air commercial television broadcasters have adopted the following principles for programme and advertising material provided to a television broadcaster:

a) Programme and advertising material shall be preceded by an audio alignment signal as specified below. The audio content as measured on a VU-type level meter shall in general be consistent with the alignment signal level. Ideally, it is intended that the television station equipment settings should remain fixed, so that there is a unity relationship between the alignment signal on the material, the ingest process and the transmission process.

b) In digital systems the alignment level will be 20 dB below full-scale digital, i.e. –20 dBFS, in accordance with RP 155. The audio peak level should nominally not exceed 11 dB above the alignment level.

c) In analogue systems the alignment level is equivalent to the digital alignment level of –20 dBFS. In an analogue transmission system, audio peaks should nominally not exceed a level of 8 dB above the alignment level.

The television station alignment level of –20 dBFS[1] (at a frequency of 400 Hz for transmitters with pre-emphasis) will be the level that causes reference modulation in the station's analogue and digital transmitters under test conditions (transmitter limiter bypassed). It will also ensure optimum operation and headroom in the station's analogue and digital recording equipment.

Audio-processing techniques employed during the production of audio material must not produce passages of audio which are strident or excessively loud. Broadcasters should process the transmission audio in the analogue and digital streams to maintain the dynamic range within a range of sound control settings that are comfortable for the listening audience.

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[1] Normal alignment level in a studio domain is normally set using a 1 kHz tone. Analogue TV transmitters employ pre-emphasis which results in a 0.5 dB gain. Transmitter alignment level must therefore be set using a –20 dBFS tone at a frequency of 400 Hz.