BBC Design & Engineering

Audio Quality Information & Standards

for

BBC RADIO

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Summary of Requirements

BBC Radio delivers a diverse range of audio content across many different listening platforms. The use of appropriate audio formats and standards is important, both for compatibility and to ensure a consistent, high quality listening experience for our audience.

This document sets out the technical requirements for audio content in terms of parameters such as file type, digital bit depth, sample rate, peak level and Loudness. It also gives guidance on subjective sound quality elements such as lossy compression and dynamic range.

In addition to guidance for those delivering programmes to BBC Radio, relevant details of the transmission chain have been included. It is hoped that this will help contributors to understand how the audio is processed on its way to our listeners and why good quality source material is important.

The main default requirements can be found in the table below. Deviation from these requirements may be acceptable but must be agreed with the commissioning Network beforehand. For more details please refer to the relevant section(s) referenced in the right hand column.

General sound quality requirements	Content must be free from unintentional distortion, noise artefacts and gross inter-channel phase differences. Speech must be intelligible.	Sec 2
Digital audio format[1]	Linear PCM	Sec 3.1
Mode	Stereo	Sec 3.2
File type	.wav	Sec 3.1
Resolution (bit depth)	16 bit	Sec 3.1
Sample rate	48 ksps (kHz)	Sec 3.1
Maximum true peak level	-1 dBTP	Sec 3.3.1
Integrated loudness	-23 LUFS	Sec 3.3.3
Dynamic range	As appropriate for commissioning Network (in accordance with their 'House Style'). Extreme amounts of compression and/or limiting should be avoided.	Sec 2, Sec 3.6
Bandwidth	Full (nominally 20Hz to 20kHz) or as required by programme material and commissioning Network.	Sec 3.4

Notes:

[1] It is acknowledged that delivery using linear PCM may not be possible over certain links. If lossy bit-rate reduction is used, care must be taken that this is done at an appropriately high bit rate using a good quality codec. See Section 3.5 and Appendix 1 for more information.

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1 Background: Definition of *high quality* and *broadcast quality* audio

BBC Radio delivers a diverse range of programmes to its listeners via terrestrial and satellite transmission, online streaming and downloads. An underlying requirement for all programmes and delivery methods is that the audio quality heard by the listener is consistently high.

Audio quality may be defined by a combination of objective parameters – bandwidth, peak level, digital bit depth and so on – plus a range of subjective elements. A listener's judgement of audio quality will depend on their own listening environment, equipment, expectations and their sensitivity to various kinds of imperfections.

For audio content to be considered *high quality*, it must meet certain minimum objective standards and be free from subjective impairment that is likely to reduce listening enjoyment for a significant number of listeners.

The factors that affect audio quality can be divided into four main categories:

- Environmental (for example venue acoustic, ambient noise)
- Technical limitation (for example microphone quality, restricted signal path bandwidth)
- Operational practice (for example microphone placement, loudness, peak level control)
- Fault conditions (for example dropouts, clicks, hum)

There is naturally some interaction between these categories: For example a change of operational practice might be used to overcome a technical limitation and reduce the likelihood of a fault condition arising.

For speech signals the additional subjective parameter of *intelligibility* must be considered. Intelligibility will usually be aided by attention to the factors above, but deserves special mention as it is a common source of listener complaint. Intelligibility should be good for the majority of listeners – including those with a degree of hearing impairment.

In radio workflows, quality expectations vary according to the limitations inherent in each type of programme and each part of the broadcast signal chain. The aim is to achieve the best possible audio quality within these limitations. For example the speech quality of a vox pop recorded in the street is not expected to be as high as that in a pre-booked studio interview. Likewise, a live concert heard via one of our emission platforms is likely to reveal some quality impairment when compared with the audio captured at the venue. In both radio and TV operations, the phrase *broadcast quality* is sometimes used to define audio (and picture) quality as being sufficiently high that it is fit for purpose.

In the following sections we define the requirements for audio used in BBC Radio services and the recommended approach to ensure that quality remains as high as possible at each stage of the signal chain. These stages are:

Acquisition	(microphone to local mixer and recorder)
Links	(between sites, such as an OB venue and base)
Production	(mixing, editing, level and EQ adjustment during production)
Interim Storage	(local recording, delivery format)
Archiving	(long-term storage copy)
Playout	(studio playout to air)
Distribution	(audio routing of playout and live content to TX platforms)
Processing	(dynamics processing, EQ and limiting immediately prior to emission)
Emission	(AM/FM modulation, digital coding and multiplexing, streaming)

2 General Sound Quality Requirements

The basic requirements for broadcast quality audio have been defined by the Digital Production Partnership (DPP) in section 1.2 (Sound Quality) of their generic document: Technical Standards for Delivery of Television Programmes to UK Broadcasters[1]. These requirements have been amended here to take account of radio's specific use of dynamics and other processing in the creation of what is often called a 'House Style' for its services.

- Sound must be recorded with appropriately placed microphones, with consideration and full attention to background levels and noise, and without distortion.
- The audio must be free of spurious signals such as clicks, noise, hum and any analogue distortion.
- The audio must be reasonably continuous and smoothly mixed and edited.
- Dynamic range should be appropriate to the broadcast chain and take into account the full range of domestic listening situations.
- Stereo audio must be appropriately balanced and as free as is practicable from phase differences which cause significant cancellation in mono.
- The use of any audio compression and equalisation of speech – either solo or over music – should model the audio to a 'House Style' that is network, timeslot and genre dependent. The use of these tools should not cause listener fatigue or discomfort in their contribution towards intelligibility.
- House Style should be considered in detail and achieved by using the above tools, alongside a very considered approach to the relative level of speech to background (music or effects) so as to achieve the required density and intelligibility for the Network, and familiarity, consistency and enjoyment for the listener. House Style guides should be used as a reference to achieve the required relativity of speech to background, and density of audio in general. They will also give guide levels to help understand at what level the speech will ideally sit within a programme.
- The audio must not show dynamic and/or frequency response artefacts as a result of the action of processing or lossy bit-rate reduction systems*.

*Note on the use of lossy bit-rate reduction

Audio should be captured and transported throughout the broadcast signal chain in a lossless digital format (e.g. linear PCM, FLAC) wherever it is practicable to do so. Where circumstances require the use of lossy bit rate reduction, care must be taken to maintain acceptable subjective audio quality. See section 3.5.

3 Technical Requirements

3.1 Digital Audio Format

Audio files should be submitted in the RIFF/WAV format (.wav).
See Specification of the Broadcast Wave Format: A format for audio data files in broadcasting[2] for details.

If the audio file is programme material delivered for broadcast, then the audio format for the file will be: Linear PCM, 48 kHz, 16 bit (or greater, by prior agreement)

To reduce file size and so speed up file delivery over the Internet, the BBC will accept such wav files encoded to FLAC format (.flac).

Audio files delivered to the Radio Digital Archive (RDA) should be stored in their native format, normally 16 bit 48kHz. In some cases, audio recorded at a venue will be of higher quality than that delivered to the RDA via the normal route (via the playout and production system). In such cases, a mechanism should be in place to ensure that the higher quality version is also retained by the RDA. Where material requires archiving in a format other than 16 bit 48kHz, bespoke arrangements must be made with the Archive.

***Note on sample rates**
48kHz is the standard audio sample rate across both BBC Radio and TV. In common with much of the audio and radio industry BBC Radio at one time used a sample rate of 44.1kHz in production environments. This was helpful when audio CD and CD-R were the principal source and recording media, but was only an interim measure. Where content has been produced at 44.1kHz or other sample rates, high quality sample-rate conversion should be used to provide a 48kHz version at the point of delivery for broadcast.

3.2 Audio Mode

Stereo programmes must always be supplied as a single wav (or FLAC) file with the two channels recorded as A and B (i.e. left and right) not as M and S (i.e. sum and difference).

Stereo programmes must be recorded so as to be compatible for listeners in mono. In general signals should be in phase between channels. The S (difference) signal should rarely exceed the M (sum) signal (otherwise cancellation can result when the signal is heard in mono). Avoid extremes of stereo imagery or “out of phase” effects as these present problems with mono compatibility.

Surround, binaural or other ‘immersive’ audio modes must only be used by prior agreement with the commissioning editor.

3.3 Loudness, True Peak and Quasi-Peak Level

BBC Radio currently specifies that all programmes should comply with Loudness Recommendation EBU R 128[5]. This requires the use of Loudness (LUFS) and true peak-reading meters compliant with
ITU-R BS.1770[6].

Programmes produced using traditional level measurement such as PPM or peak-reading bar-graph meters so that they “peak to PPM6” will still be accepted, provided they comply with the requirements for maximum true-peak and quasi-peak level stated below. However it is BBC Radio’s intention to stop accepting programmes made to this standard in the near future.

3.3.1 True Peak Level

Applicable to <u>all</u> programmes, regardless of the meter used to monitor levels during production.

Not to exceed -1dBTP measured using a true peak (4x oversampling or higher) meter, compliant with ITU-R BS.1770[6].

3.3.2 Quasi-peak level

For programmes produced with the aid of a PPM (Peak Programme Meter according to EBU Specification Tech. 3205-E[3]): Peak level on A, B or M is not to exceed PPM 6.

3.3.3 Integrated Loudness

For pre-recorded programmes produced in accordance with EBU R 128[5], the integrated loudness for the programme or programme segment shall be -23 LUFS ±0.5 LU*.

For live programmes, the tolerance on this target is relaxed to ±1.0 LU.

*Note on loudness targets
The EBU recommended target loudness of -23 LUFS is currently not applicable to the emission stages of the radio broadcast chain. Agreement has yet to be reached within the UK radio industry on normalising loudness on AM, FM, DAB and DVB services. For online radio streams, the AES has proposed a higher target loudness of between -16 and -20 LUFS, primarily to ensure good audibility on personal music players used with earphones. See section 3.6.6 for more details.

Loudness data relating to our current services and emission platforms can be found in Appendix 3.

3.3.4 Loudness Range (for information only)

Loudness range (LRA) is a parameter defined by EBU TECH 3342[7]. It is intended to indicate the amount of long term loudness variation within a programme. The higher the measured value of LRA, the more likely it is that a listener will need to adjust replay volume to maintain audibility – particularly where there is ambient noise. In the context of radio broadcast, loudness range is often reduced by processing at the emission stage. This helps to ensure audibility for listeners in cars and other noisy environments.

EBU guidance does not define LRA targets, but suggests that broadcast content with a value exceeding 20 LU is likely to cause audibility issues for listeners in certain environments.

The example LRA values below are typical for source material intended for radio broadcast – they are not target values. In most cases these values will be further reduced by dynamics processing in the signal chain prior to emission.

Classical music:	10-20 LU
Drama:	6-11 LU
Rock & pop:	3-9 LU
News & sport:	3-6 LU

3.4 Audio Bandwidth

Full audio bandwidth is generally considered to be 20Hz to 20kHz, which is the range of frequencies audible to a young person. Most of our audio signal paths are capable of handling – and should deliver – at least this range of frequencies without significant loss or emphasis. Notable exceptions are where HF bandwidth is curtailed due to the use of lossy encoding (see below) or where EQ has been applied to reduce the subjective effects of microphone wind noise or other extraneous noises*.

*Note on extraneous noises

These are sometimes inaudible to those monitoring in the studio, only to cause problems further down the signal chain. A good example is TV line frequency interference (15.625kHz) from old CRT monitors, which is present on some archive recordings. This is sufficiently high in frequency as to be inaudible to many people over 40 years of age, but may be heard by younger listeners (or dogs!). Very low frequencies can cause similar problems as they are inaudible on small studio monitors. Some loudness / level meters include a spectral display that can be useful in highlighting such signal artefacts.

3.5 Lossy Audio Codecs

3.5.1 General Principles of Lossy Codecs

Lossy codecs reduce the bit rate and, consequently, the file size of audio content by applying psycho-acoustic principles. In so doing, they discard data beginning with that which is likely to have the least impact on perceived audio quality. The basic characteristics of lossy audio codecs are:

- For a given type of codec, there is a bit rate 'sweet spot' at which most listeners find it difficult to distinguish the lossy encoded version of the audio from the (linear PCM) source. Under these conditions the codec is sometimes referred to as being 'near-transparent'. Increasing the bit rate beyond this point is likely to bring diminishing returns in terms of higher sound quality (although it might help to lessen the effects of cascading – see below).

- All lossy codecs degrade the audio signal and introduce artefacts (distortion products). These artefacts are highly dependent on programme content and range from very obvious to imperceptible depending on the type of codec and its settings.

- 'Lossy' means just that: Once parts of the signal have been discarded by the codec, no amount of processing can recover it.

- The cascading of one lossy codec after another will usually cause further signal degradation. The belief that once data has been discarded, a second or third pass through a similar codec will cause no further damage to the audio is false. Whether the resulting degradation becomes audible will depend on the combinations of codec types and bit rates in use.

- Lossy codecs have evolved over the last 30 years to a point where the early codec types are fully mature (i.e. little or no further improvement is possible). Many of these early codecs (e.g. G.722, MPEG 1 Layer II) remain in use for reasons of compatibility and because their relative simplicity keeps latency to a minimum. Some codec types such as HE-AAC have been optimised for use at very low bit rates. This does not necessarily equate to greater transparency at high bit rates than for earlier types of codec.

As far as is reasonably practicable, the use of lossy audio codecs should be avoided during acquisition, editing, interim storage, archiving, internal distribution and dynamics processing.

The use of a lossy codec such as mp3 for material outside the programme production chain can sometimes be helpful. Examples might be distributing audio to an editorial team for compliance checks, emailing inserts to a reporter to aid with script writing, or providing programme contributors with a copy of the programme. Care must be taken to ensure that the lossy version of content is not inadvertently incorporated back into the production chain or supplied for transmission.

3.5.2 Guidance on Codec Types and Bit Rates to be used

There are two sections of the broadcast signal chain where the use of lossy audio codecs is inevitable:

1. Emission, for which lossy audio encoding is an integral part of our digital transmission platforms. The types of codec used for this are generally defined by the emission platform and the bit rates are constrained by the available transmission bandwidth. (See Appendix 2 for details of current emission rates.)

2. Links between OB venues and base, where cost or bandwidth limitations do not allow linear PCM to be used. Here, there will be certain sound quality and latency expectations according to the type of programme or contribution. Once minimum expectations have been defined, the exact codec and bit rate will depend on what is available and possible at the time.

The following paragraphs, together with the table in Appendix 1, give guidance on the types of lossy codec and minimum bit rates that should be used. When considering the use of lower bit rates than these, remember that cascading a codec with additional lossy codecs further down the chain will increase the risk of signal artefacts becoming audible to the listener. The effects of cascading are very difficult to predict, so it is advisable to maintain as high a bit rate as possible early on in the chain.

As stated in Section 1, audio quality expectations vary with programme (or contribution) type and duration. In order to provide guidance, three programme categories have been defined. These indicate the degree of audio quality reduction that can be accepted where the use of certain codec algorithms and/or lower bit rates offer significant benefits in other areas (e.g. the cost of a link). The allocation of programmes to particular categories should be by prior agreement between Radio Operations and the relevant production departments.

Category A
Stereo productions, where audio quality is likely to be a significant factor in listener enjoyment. Examples are live music concerts, festivals and recorded music programmes where the music is played out from a remote site rather than base. These are also likely to be programmes longer than half an hour.

Category A programmes demand very high quality and full bandwidth audio. Codec artefacts should be imperceptible to the majority of listeners, even via our highest quality emission platforms.

Linear PCM remains the preference for this category of programmes, but only where the link has sufficient bandwidth to support it without compromising resilience.

Category B
Stereo productions, where music is either absent, or features less prominently within a programme.

Speech programmes such as panel discussions, where stereo is useful in providing positional information for the listener.
Remote inserts in stereo, where it is important that the audio quality from the OB matches that of contributors in the studio.
As above, but in mono, allowing for a further reduction in bit rate.

Category B programmes require high quality audio. Slight bandwidth reduction and mild codec artefacts are acceptable.

Category C

Mono inserts to news, magazine programmes, etc., where speech clarity and intelligibility takes priority over audio quality

Category C programmes must retain good overall clarity, but some noticeable bandwidth reduction and codec artefacts are acceptable.
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A further consideration is whether a link is the main TX path or a backup route. It is usually acceptable to risk a slight reduction in audio quality relative to the main TX path if this provides a significant cost saving. However it is important that any degradation is not too great. Guidance is given in the table in Appendix 1.

3.6 Gain Adjustment and Processing

3.6.1 Microphone Processing

Microphone processing (or EQ of a mic feed) is highly personal and will normally be devised by or with individual presenters to give their voice a particular timbre on air. As with other types of adjustment applied in the studio, it needs to be done with care so that it complements rather than 'fights' any multiband processing applied further downstream.

3.6.2 Limiting and Compression

Limiting and compression are useful tools in production and may be applied at the discretion of the sound engineer or presenter, according to local guidelines and/or House Styles. In essence they should gently control the dynamic range of source material where this is considered excessive. High compression ratios and brick-wall limiters can lead to unpleasant 'pumping' if over used, so should be applied with care. It should also be remembered that most services apply multiband processing downstream of the studio that is appropriate for each emission platform. This equipment generally performs a better job of applying large amounts of compression and limiting than the equipment available in a typical radio studio.

3.6.3 Normalisation

Normalisation is the adjustment of the level or loudness of a piece of audio in order to make it conform to a particular target value. It is frequently provided as a utility in digital editing or other production tools.

Until recently level normalisation was the only option provided, in which the peak level within an audio file would be adjusted to hit a certain value and no higher. Very often, the value chosen would be

-10dBFS, in the mistaken belief that this would make a programme conform to the BBC requirement of 'peaking to PPM 6'. This is to misunderstand the dynamic behaviour of a PPM, for which -10dBFS only corresponds to PPM 6 on tone signals, not programme.
A second flaw in using peak normalisation is that, its effect is skewed by isolated transients within a programme. This can result in a lower average loudness for the whole programme than was intended.

Loudness normalisation offers a much better solution because it allows programme content to be matched in terms of subjective loudness. This is what matters to the listener because it reduces the need to constantly adjust replay volume. The EBU R 128[5] loudness target of -23 LUFS ensures that most programme material can be normalised without requiring any dynamic range adjustment. In essence, heavily compressed audio needs to be turned down in order to meet the target, whereas wide dynamic range material needs little adjustment or just a slight increase in gain. Some care is needed with very dynamic material, for which a small amount of peak limiting might have to be applied. Most loudness normalisation utilities cater for this.

Further information on loudness normalisation can be found in EBU TECH 3343: Guidelines for the Production of Programmes in accordance with EBU R 128[8].

3.6.4 TX (Emission) Processing

This topic is fraught with misconceptions, profound beliefs and a long legacy of *how it's always been done*.
In the beginning, AM radio broadcasts had to be louder than the noise inherent in the technology. The introduction of FM broadcasting reduced the noise floor to the point where it became inaudible in many domestic listening environments. Digital emission platforms have dropped the noise floor even further, to the point where it has effectively gone for all our audiences. However, similar multiband processing to that used on AM transmissions continues to be applied to FM platforms and to a lesser extent on digital platforms.

Some radio stations still tend to want a "Station Sound" and to be louder than their competitors, it seems. However in a multi-platform digital, on demand world, what should that "Station Sound" be like?

As engineers, we bow out at this point but respectfully suggest that in a digital world, new thinking is required as our audience becomes multi-platform - picking their way from on-demand streaming services to a curated broadcast, to locally stored file, wherever their individual whim takes them. Our industry has lived for decades on "being loudest on the dial". Is that still the right way to broadcast? Or is it to be the "most distinctive on the dial"? Is any of that achieved with hyper-compression, treble boost and distorted transients?

It is beyond the scope of this document to provide a definitive answer but here are some points to consider when reviewing "Station Sound":

- How consistent is the station's studio (i.e. unprocessed) output in terms of dynamics, average loudness and frequency balance? In other words, to what extent is processing required (or currently being used) to smooth out variations in the production environment?

- Does the station's output remain audible for listeners in noisy environments, without them needing to constantly adjust replay volume?

- Are processing artefacts such as pumping, distortion or frequency response errors likely to be obvious to many listeners?

- Is the amount of processing appropriate for each emission platform? For example is the amount of dynamic range reduction and equalisation weighted in favour of those listening on devices with small speakers and/or in noisy environments? Does this result in too much bass, distortion, etc, that spoils the quality for those listening on better equipment?

- Does the processing need to be 'day-parted' so that it takes account of likely listening situations and presentation styles? For example, breakfast and evening 'drive time' slots (lots of listeners in cars or public transport – moderate compression) versus specialist evening programmes (listeners predominantly at home – mild compression).

In particular, a 'set and forget' mentality is often applied to TX processing in spite of significant changes in production workflows or programme schedules. It is therefore important to occasionally review the sound of a station across all its emission platforms.

3.6.5 Emission Headroom

The misguided quest to be the 'loudest on the dial', often results in emission headroom being sacrificed. A quick scan of commercial DAB stations reveals some that peak to 0dBFS – in other words they are devoid of headroom!

Whilst occasional peaks to 0dBFS might be tolerable, the nature of TX processing means that peaks will in many cases be constantly 'hitting the end stops'. This can lead to distortion, particularly in receivers that have little or no headroom in their digital to analogue converters.

EBU R 128[5] recommends that true peaks do not exceed -1dBTP in the production environment. It further states that additional headroom is required at the input to lossy encoders. These can generate overshoots – peaks that weren't present in the source material – once the audio has been decoded for conversion to analogue. As lossy encoding is used on all our digital emission platforms, we need to factor this into the headroom calculations for our broadcast chain.

At present, most of our emission platforms are set up with a nominal 5dB of headroom – that is, processing is set to 'brick-wall' limit peaks at -5dBFS. In practice, this figure is eroded slightly due to TX processors failing to limit all peaks at exactly this level, or where overshoots have been introduced by equipment downstream of the processor. As a result the maximum true peak level measured at the input to our emission encoders is typically -4dBTP.

A notable exception to the above is Radio 3 digital. This has no dynamics processing applied to its output. To remove the possibility of clipping, a protection limiter prevents peaks exceeding -2dBFS from reaching the inputs to the emission encoders. Whilst this 2dB headroom is less than on our other services, the nature of Radio 3's output is such that the average peak level is significantly lower. The extended working range is only used to accommodate exceptional transients such as those on musical crescendos during live concerts.

3.6.6 Service Loudness

As mentioned in section 3.6.4 on TX processing, radio stations have a long history of competitive loudness. This has led to increasingly aggressive processing being applied, especially by rock and pop stations. The loudness of these is typically in the range -9 to -14 LUFS when measured off air.

At the other extreme, a station such as Radio 3 that broadcast wide dynamic range content may have an average loudness at the receiver of less than -23 LUFS. Although to some extent a step-change in loudness is expected when switching between a classical and a pop station, this is no longer desirable or incurable with digital platforms. In particular, the ability to stream internet services and build custom programme sequences makes loudness matching more important than ever.

EBU R 128[5] offers a potential solution to the problem of loudness control at the emission stage as well as in production. Unfortunately, there are currently two barriers to its implementation: Firstly the loudness rivalry between some stations persists, and secondly the difficult listening environments in which radio services are often consumed.

In particular, it has been found that the normalisation of audio material to -23 LUFS does not allow comfortable listening levels to be achieved on some devices, especially portable music players and phones. Although the solution to this problem ultimately lies with the device manufacturers, an interim solution is required for users of current devices.

Work by the AES, with input from the BBC and others, has proposed an interim target loudness of between -20 and -16 LUFS for internet radio streams and downloads. This 4 LU window would allow the likes of Radio 3 to be accommodated (at -20 LUFS) with minimal peak limiting, whilst rock and pop services could be normalised to the higher end of this target range. The result will be a maximum difference in average loudness between services of 4 LU which is much more satisfactory than the current window of up to 15 LU. For more information see AES TD1004.1.15-10: Recommendation for Loudness of Audio Streaming and Network File Playback[9].

3.7 Studio Monitoring

The last ten years has seen a large leap in the audio fidelity of the components used in modern devices, be they radios, smart phones, TVs or tablets – each is capable of producing exceptional audio quality. However, this technological leap has also seen loudspeakers get smaller and sometimes these are the weakest link in the chain from microphone to listener. It is also impossible for the programme maker to know the quality of the speaker their audience is using. It could be the speaker on the base of the smart phone, earbuds or headphones, or it could be the smart phone plugged into a dock in the living room driving a pair of hi-fi speakers.

Thus full range, analytical studio monitoring used at the optimum listening loudness remains key to the production of high quality audio - although the circumstances of a production often inhibit this ideal.

Either way, mixes should be checked on small, lo-fi speakers during production. The “check loudspeaker” remains as important today as it has ever been.

Recommendations on setting optimum monitoring loudness can be found in section 8.2 of EBU TECH 3343: Guidelines for Production of Programmes in accordance with EBU R 128[8].

Information on loudspeaker configuration for multichannel surround sound can be found in International Telecommunication Union Report ITU-R BS.2159-4: Multichannel sound technology in home and broadcasting applications[10].

3.8 Service Watermarking

Several proprietary techniques are available to embed audio watermarks in station output. As distinct from data carried separately from the audio, these watermark signals have the potential to affect audio quality. Although the manufacturers of such devices strive to minimise possible audibility of the embedded signals, the watermarks must also be sufficiently robust to allow reliable decoding. In other words there will always be a trade-off between the two factors. Careful listening tests must be conducted – using the exact same configuration as proposed for the final installation – before watermarking is applied to live services.

References

[1] Digital Production Partnership: Technical Standards for Delivery of Television Programmes to UK Broadcasters

[2] Specification of the Broadcast Wave Format: A format for audio data files in broadcasting

[3] EBU Tech. 3205-E: The EBU Standard Peak Programme Meter for the Control of International Transmissions

[4] BBC R&D White Paper WHP202: Terminology for Loudness and Level, dBTP, LU and all that

[5] EBU Recommendation R 128: Loudness Normalisation and Permitted Maximum Level of Audio Signals

[6] ITU-Recommendation BS.1770-4: Algorithms to measure programme loudness and true peak audio level

[7] EBU TECH 3342: Loudness Range: A measure to supplement EBU R 128

[8] EBU TECH 3343: Guidelines for Production of Programmes in accordance with EBU R 128

[9] AES TD1004.1.15-10: Recommendation for Loudness of Audio Streaming and Network File Playback

[10] International Telecommunication Union Report ITU-R BS.2159-4: Multichannel sound technology in home and broadcasting applications

Appendix 1: Lossy Audio Codecs for Contribution Links: Minimum Requirements

Codec Type	Link Status	Programme Category					
		A		B		C	
		Mode	Bit Rate	Mode	Bit Rate	Mode	Bit Rate
G.722	Main	N/A	N/A	N/A	N/A	N/A	N/A
	Backup	N/A	N/A	N/A	N/A	Mono	64
MPEG LII	Main	Stereo	384	Stereo / Mono	192 / 160	Mono	64
	Backup	Stereo	256	Stereo / Mono	128 / 96	Mono	64
MPEG LIII	Main	N/A	N/A	Stereo / Mono	128 / 96	Mono	64
	Backup	Stereo	128	Stereo / Mono	112 / 96	Mono	64
aptX	Main	Stereo	384	Stereo / Mono	192 / 160	Mono	64
	Backup	Stereo	256	Stereo / Mono	128 / 96	Mono	64
AAC-LC	Main	Stereo	320	Stereo / Mono	128 / 96	N/A	N/A
	Backup	Stereo	192	Stereo / Mono	128 / 96	N/A	N/A
HE-AAC	Main	N/A	N/A	Stereo / Mono	96 / 64	Mono	32
	Backup	Stereo	128	Stereo / Mono		Mono	32
AAC-LD	Main	Stereo		Stereo / Mono		Mono	64
	Backup	Stereo		Stereo / Mono		Mono	64
Opus	Main	Stereo		Stereo / Mono		Mono	32
	Backup	Stereo		Stereo / Mono		Mono	32

Notes

(i) N/A means that this codec is not suitable for the programme category and link status.

(ii) Where Stereo (St) is specified, this must be full stereo mode and not joint-stereo where this option is offered in the codec settings.

Appendix 2: Emission Codecs and Bit Rates (current values for information only)

Platform	Codec	Radio 1	1 Xtra	Radio 2	6 Music	Radio 3	Radio 4 FM	Radio 4 LW	4 Extra	5 live	Sports Extra	Asian Network
DAB	MPEG LII	128 js	128 js	128 js	128 js	192 s / 160 js	128 js / 80 m	64 m	80 m	80 m / 64 m	64 m	64 m
DVB	MPEG LII	192 s	160 s	192 s	160 s	192 s	192 s	96 m	160 s	96 m	96 m	128 js
Online Profile 1	AAC-LC	320 s	320 s	320 s	320 s	320 s	320 s	320 s	320 s	320 s	320 s	320 s
Online Profile 2	AAC-LC	128 s	128 s	128 s	128 s	128 s	128 s	128 s	128 s	128 s	128 s	128 s
Online Profile 3	HE-AAC v1	96 s	96 s	96 s	96 s	96 s	96 s	96 s	96 s	96 s	96 s	96 s
Online Profile 4	HE-AAC v1	48 s	48 s	48 s	48 s	48 s	48 s	48 s	48 s	48 s	48 s	48 s
Online Profile 2a	MP3	128 s	128 s	128 s	128 s	128 s	128 s	128 s	128 s	128 s	128 s	128 s

Notes
Where alternative bit rates are shown for DAB services, the lower value is used when additional, regular services such as Sports Extra join the multiplex. Short-term 'pop up' DAB services are normally accommodated by reducing the bit rates of R1, 1X, R2 and 6M from 128k js to 112k js.

Online bit rates are dependent on device and prevailing connection speed. Most desktop PCs, smartphones, tablets and internet radios are capable of decoding profiles 1-4. Profile 2a is used by some low-end internet radios and third party apps.
For more information see: http://iplayerhelp.external.bbc.co.uk/radio/streaming_codecs_bitrates

DAB services bit rates can be found at: https://intranet.gateway.bbc.co.uk/designengineering/distribution-and-business-development/Pages/DAB.aspx

Appendix 3: Service Loudness (current typical values for information only – these are <u>not</u> targets!)

		Radio 1	1 Xtra	Radio 2	6 Music	Radio 3	Radio 4 FM	Radio 4 LW	4 Extra	5 live	Sports Extra	Asian Network
Pre-Processor	**Loudness** (LUFS)	-17	-17	-19	-19	-27	-22	-24	-23	-21	-23	-18
	Max True Pk (dBTP)	-4	-1	-3	-3	-6[1]	-3	-6	-1	-6	0	0
	LRA (LU)	8	6	7	8	15	6	7	8	6	8	7
DAB	**Loudness** (LUFS)	-14	-13	-15	-14	-23[2]	-18		-19	-18	-15	-13
	Max True Pk (dBTP)	-4	-4	-4	-4	-2[2]	-5		-5	-5		-3
	LRA (LU)	3	2	5	3	20	5		7	4		2
DVB	**Loudness** (LUFS)	-14	-13	-15	-14	-27	-18		-19	-18	-15	-13
	Max True Pk (dBTP)	-4	-4	-4	-4	-6	-5		-5	-5		-3
	LRA (LU)	3	2	5	3	15	5		7	4		2
Online	**Loudness** (LUFS)	-14	-13	-15	-14	-27	-18	-19	-19	-18	-15	-13
	Max True Pk (dBTP)	-4	-4	-4	-4	-6	-5	-4	-5	-5	-4	-3
	LRA (LU)	3	2	5	3	15	5		7	4		2

Notes

1 This is typical peak level for Radio 3. There is normally no peak level limiter applied in the studio so pre-processor level can in theory peak to 0dBFS

2 Radio 3 DAB has 4dB gain applied prior to the encoder. Peak level protection limiting is set to -2dBFS