BCD AUDIO

Broadcast Audio
Custom Manufacture
Design and Consultancy

LIMITS OF AUDIO SIGNALS

It is an often-neglected fact that all Audio electronics are in reality a huge combination of physical bits, all of which are obeying the laws of physics, and whilst the units purpose is to process and distribute high-quality audio, in physical terms it can only do the following:

Create heat:
As all circuitry consumes power in its own right. Power will also be passed out of the output ports, and used to power the displays and indicators. Interestingly, An AES-3 signal terminated into 110R could actually supply 200mW of power to that load!

Add noise:
All electronics will add physical noise to the signal. It is this process that limits the minimum signal levels that are achievable. The noise can be reduced by dropping the system impedances but this will create more heat and will reduce the maximum output levels achievable. There is a theoretical limit to the noise floor, which is based on the thermal noise of the electrons moving around, and this is the 'Boltzmann ' noise. This noise is proportional to the impedance, bandwidth and temperature. This theoretical noise limit normally only dominates the noise performance of high-gain paths, and so mostly affects microphone stages. At gains above 40dB it starts to define the noise floor. Boltzmann noise will also define the noise limit in EQ and mix stages, but the microphone path noise still dominates. Digital audio is limited by the quantisation noise of the converters, and with modern devices is similar to the best analogue systems. (That was not the case with early 16bit parts, and the cheaper Multimedia devices.)

Add distortion:
All electronics will contribute some non-linearity to the signal. Fortunately, modern devices are quite good in this respect. Note however that dropping the impedances will worsen the distortion. An often-neglected fact is that small value ceramic capacitors can easily produce large volumes of distortion! This is particularly a problem with modern surface-mount circuitry and BCD Audio always use good quality polyester capacitors in the filter stages, particularly around Digital to Analogue converters. Audio transformers were traditionally used as output stages; these have good system properties but create more distortion at low frequencies. Eventually the frequency is so low that the transformer saturates into mega-distortion; this effect can be as high as 100Hz in small transformers, and contribute 1 to 5 % distortion at low frequencies.

Limit the frequency response:
The ideal circuit would react like a piece of wire and would not be limited in frequency response, but real circuitry is not like that. A circuit that passes signals down to DC levels is undesirable, as any offset in the input signal will pass through the circuit, giving trouble at any switch or control. Limiting the response to around –3dB at 2Hz is therefore desirable, and at higher frequencies when some immunity to mains hum is desirable (for example in talkback chains.) If the circuit was not limited at higher frequencies it would be susceptible to RF interference and may even pass through signals that were not intended; for example 15KHz line rate whistle from Televisions, Infra-red remotes etc. As the frequencies rise, the electronics will eventually be unable to follow the signal and gross slew-rate distortion will occur. In addition, RF interference will use your audio system as an AM receiver, enabling you to listen to the local taxis! It is therefore desirable to limit the frequency response above 15KHz and definitely above 50KHz.

Limit the output level:
The level is limited by the power rails of the unit, and increasing these rails above sensible limits only adds to the heat generated. Available parts will also limit the power rails used. Lower rails will severely compromise the achievable output level and the 5V rated parts are notorious for adding

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distortion, noise and poor output drive capability. Balanced output and mixing techniques double (6dB) the output drive capability for given power rails. In practice 24 to 30V DC power rails are suitable for systems running at 0dB reference, and allow peaks to rise to nearly +26dB. These levels are a good match to real audio signals, which are calibrated for 0dB level, with peaks to +8dB and still have adequate headroom for the occasional overload or incorrect gain setting. Domestic equipment often runs on lower voltage rails, but the system level is dropped to –10dB to compensate, and is nearly always unbalanced.

Lose dynamic range:

As we have seen, the electronics define the maximum output level, its noise contribution with added thermal noise limit the minimum level, and the difference between these two figures is the dynamic range. Therefore, unless noise gating is employed, the only thing the overall system can do is lose dynamic range! Any given system will have a defined dynamic range, and the operating level is chosen to intelligently (!) use that range. Systems can be designed with a low operating level, with the advantages of greater headroom but worse noise performance, or with a high operating level, with advantages of better noise but lower headroom. Unprocessed signals (from Microphones, before the fader) generally need greater headroom, so unexpected peaks do not get clipped. Processed signals after the fader and during transmission can use less headroom as there is less chance of unexpected peaks. The top quality Analogue systems use 0dB reference, with +26dBU maximum level and system noise floor of around –85dBU giving a dynamic range of around 111dBU. This system setting is suitable at every stage in an audio path so could be argued to be the perfect compromise! In practice, mixing stages before the fader might use a lower reference level to gain more headroom, and transmission stages might use a lower maximum level and accept less headroom, but 0dB remains a good compromise.

The dynamic range of digital systems is firstly defined by the number of bits used, and each bit gives approximately 6dB as expected. Therefore 16bit systems have 96dB dynamic range, 20bit systems have 120dB dynamic range and 24bit systems 144dB dynamic range. Digital systems are generally defined from the peak level 0dBFS, and the system level has been determined by the industry to be either –18dB(BBC/EBU) or –20dB (USA). These calibration points were determined when the converters used were 16bit and the noise performances were much worse than the theoretical limits mentioned above.

The industry is now stuck with these limits, and the available headroom is not adequate and not as good as the earlier analogue systems with 26dB headroom. Modern 24bit converters in reality do not give results better than the theoretical 20bit performance (apart from low level distortion) , but in practice are adequate. The current setting of 18dB analogue maximum level , with 0dB nominal typically gives –91dB noise floors and a dynamic range of 109dB, but is light on headroom, and in practice levels are being pushed to the limits, with many overloads being present in final transmission signals. The current converter performance would enable the system level to be reduced, with the advantage of better headroom, and no appreciable increase in noise floor.

BCD propose a new headroom setting of +24dBU, which would represent the best compromise, and would be useable at all stages of the audio path. The current setting of 18dBU is only acceptable in the transmission stages. There are movements in the industry to use even worse headroom settings in the transmission path, and BCD do not understand why; there are no noise advantages in doing this now, and overloads will be more common.

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