BBC Digital Audio—A Decade of On-Air Operation

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The BBC has unique experience in the benefits and problems of digital audio, having used digital systems in the normal distribution of sound signals from London to various transmitters around the British Isles continuously for a period of some ten years. The BBC also evolved a digital stereo tape recorder in 1971, followed by a multitrack recorder and a working digital sound control desk in the late seventies. Experiments in how the broadcast transmission of digital audio is affected by difficult reception conditions have also been conducted. At a time when the BBC as well as other users and manufacturers of audio equipment are contemplating larger scale excursions into the use of digital techniques, it is appropriate to discuss our experience in the subjective and objective evaluation of such systems during their design, acceptance, and continuing use. Some of the impairments which may arise are not disclosed by conventional distortion measuring techniques and although they may only be detected subjectively on a limited variety of program material under ideal listening conditions, if uncorrected they may lead to the generalized criticism of digital sound which exists in some areas. It is also important to consider the repercussions on associated analog components of the introduction of digital processes into parts of the audio chain, such as the performance of analog limiters, the limitations of existing level indicating meters, and even the criteria for acoustic noise levels in studios.

0 INTRODUCTION

Most of the speakers addressing this conference have concerned themselves, very appropriately, with looking into the near future of digital audio to propose solutions to the various problems that may arise. At this stage it may be helpful to survey the BBC's experience of digital audio over a wide field during the last ten or eleven years and to note the reasons for adopting certain digital options and deferring others.

The BBC adopted digital techniques to solve some specific problems of program distribution in the early seventies. We did not choose them just to obtain improved frequency response or signal-to-noise ratio, though these were a useful bonus. At the time there was no practicable analog solution to the problem of distributing three stereo networks 24 hours per day to transmitters up to 500 miles away, or routing the sound portion of our television services on the same circuits as the video.

I will begin by running through a little of the history of digital audio in the BBC to clarify our reasons for introducing digital techniques into specific areas. On this experience we base our comments on the current state of progress and our plans for future expansion into the digital field.

1 SOUND-IN-SYNCS

This was the first BBC-designed digital system to enter broadcast program use in 1971. Up to that date television sound was carried on analog lines, which could have different routings from the vision signal, so operational errors could arise. Also the sound quality, by the time it had traveled 400 or 500 miles, was rather disappointing. The digital system adopted used 10-bit coding plus analog compansion (compression and expansion, controlled by a line-frequency pilot tone). By sampling at twice line frequency we obtained a 14-kHz bandwidth and a 65-dB signal-to-noise ratio. (I will continue throughout this paper to use the old system of peak-signal to peak-noise measurement using a quasipeak indicator.) The digital signal was then inserted into the bottom of the sync pulse at the rate of two samples per line.

2 13-CHANNEL SOUND DISTRIBUTION SYSTEM

Over the first 50 years of the BBC's growth a complex network of "Post-Office lines" had evolved to distribute the various radio programs to the hundreds of transmitters around the British Isles. Initially physical copper wire lines were used with one channel per pair. Later a variety of carrier systems were incorporated. A long line, made up from various arbitrary links, might only offer a 5-kHz response and a 45–50-dB signal-to-noise ratio, plus many thousands of degrees of phase rotation varying with frequency. The requirement for a mono-compatible stereo distribution could not be met by such a network, so we developed a 13-channel pulse-code-modulation (PCM) system to distribute all the radio services we could foresee in 1972. This used 13-bit linear coding, giving a 15-kHz response. It has continued in use 24 hours a day ever since, with very few faults. The main reason for contemplating its replacement is the need for more program channels—it is by current communication standards rather wasteful of bandwidth with a bit rate of 6336 kbit/s—rather than any significant improvement in quality in more recent techniques.

3 DIGITALLY COMPANDED SYSTEMS

In order to reduce bandwidth requirements while keeping the same, or hopefully better, quality, it is necessary to use digital compression and expansion. A long series of evaluations of near-instantaneous compansion systems, such as the BBC's own NICAM, and instantaneous compansion systems, such as A-law, have led to the adoption of our NICAM-3 system for extensions of our distribution system and for outside broadcast (OB) contribution circuits. This has been described in detail elsewhere and uses a 14-to-10-bit compansion, permitting six mono channels on 2048 kbit/s.

4 DIGITAL RECORDING

In parallel with the development of the PCM distribution systems, the BBC Research Department evolved a digital magnetic tape recorder. The first practical machine, demonstrated in 1971, was a stereo recorder using 16-track heads on 0.5-in (13-mm) tape running at 15 in/s (381 mm/s) and recording a density of 5 kbit/in.

This was followed in 1976 by a 10-channel multitrack recorder, using 40 tracks on 1-in (25-mm) tape at 20 in/s (501 mm/s). The success of this encouraged discussions with various potential manufacturers, which led to the well-known 3M machines.

Beyond this stage we have left further development of complete recording machines to the various professional equipment manufacturers. As each new type of machine appears, we explore its potential and its problems in operational use. We broadcast a fairly regular sequence of one or two digitally recorded programs per week on Radio 3, our serious music network, although, of course, this does not necessarily offer our listeners a spectacular quality improvement since a large proportion of our concerts in this service are broadcast live.

We started with the Sony PCM 1600 and U-matic VCRs installed in a convenient vehicle. A VCR is a very inconvenient machine for slick operation, and we are now using Telefunken Mitsubishi 0.25-in (6.2-mm) machines, while awaiting developments from Studer and others with interest.

5 DIGITAL CONTROL DESKS

In working on our 10-channel recorder we soon felt the desirability of manipulating the signal in digital form, so in 1976 we developed a digital control desk as a test bed for investigating the problems of fading, frequency response adjustments, and so on, in the digital mode. While never seen as a practicable operational desk, as it required certain operational program changes to be fed in on paper tape, the concept of associating assignable controls with digital mixing soon became obvious. Because of limitations imposed by commercially available computers of practicable size, we developed our COPAS system (computer for processing audio signals) to provide economically the very-high-speed multiple-multiplication ability necessary for real-time audio processing.

As in the recording field, our technical involvement is primarily in investigating the validity of new techniques and indicating suitable avenues of development rather than in building operational equipment, so construction work based on BBC digital mixing desk concepts is currently in the hands of commercial manufacturers. We have now placed an order with Neve for a full-scale 48-channel mixing desk based on field trials of a simple assignable desk recently evaluated. This should be delivered shortly and will be installed in a large vehicle, together with digital and analog recording equipment, to allow countrywide operation.

6 EVALUATION OF DIGITAL SYSTEMS

Quite apart from BBC's own research, already described by Mr. Gilchrist, into the fundamental principles and applications of digital technology, we have a wide operational experience of commercial digital equipment. We investigate and evaluate each new technique or item of equipment as it becomes available, but will only adopt it as a firm part of our very large broadcasting system if it offers a facility previously not practicable or gives worthwhile audible improvement to some link, and so long as it introduces no detectable degradation relative to current techniques.

It is, of course, possible to design bad analog-to-digital (A/D) converters or antialiasing filters, just as it is possible to design bad transistor amplifiers or even valve amplifiers; also, even the best equipment can develop a fault. So if in a demonstration of some long awaited, perhaps heavily publicized new development, some listener claims to hear a "metallic" sound on the digital link, or a change in the ratio of direct to reverberant sound, or some other unlikely musical defect which the designer knows can't really happen, it is vital for the future of digital audio that the mystery be investigated thoroughly and a credible explanation found and published. Final purchasers of audio equipment and recorded or broadcast programs are already being persuaded that copper wire or adhesives exhibit musical effects far outside their normally accepted physical properties. They will easily accept claims from the antidigital critics that all digital systems introduce insidious, unmusical effects which defy measurement.

It is important that specialist, professional listeners, who can speak the engineering language, identify any such unexpected impairments quickly and help the equipment designers to eliminate them long before the public or the hi-fi press get a chance to deduce that all digital systems introduce subtle degradations which are absent in the analog equivalent. Certainly the BBC would not have introduced a PCM distribution to serve our VHF FM networks if there was any such inherent risk.

In my experience any defect which is consistently audible to a professional listener can be demonstrated in (possibly complex) objective measurement, with the possible exception so far of loudspeaker defects which we are not considering here. Nevertheless many defects are far easier to identify aurally than to measure, and they would never show up in exhaustive measurements unless pinpointed in a listening test. For example, a 2-kHz sine-wave tone is audible even behind program material at a level 80 dB below that program, perhaps 20 dB below the "noise level" as normally measured. It can be traced by spectral analysis and correlation techniques once one knows the frequency to look for, but an experienced listener will spot it in seconds. Some of the impairments which may arise in digital equipment will be as difficult to measure as this and may be new types of noise or distortion not previously encountered, so even the experienced listener will not spot them at first.

This brings us back yet again to the vital question of comparing analog and digital systems. Such comparisons arise in three fields: the comparability of specifications, objective (measured) tests, and subjective (listening) tests. There are similar hazards in all three of these fields, and they cannot be considered in isolation. In a perfect world there would be no ambiguities, and equipment would meet its specifications whether assessed objectively or subjectively. The problem is that we are seldom comparing like with like.

Let us look at the basic parameters of an audio system: frequency response, distortion, and signal-to-noise ratio—all apparently capable of simple, unambiguous measurements which should correlate with audible effects. However, closer inspection shows that simple measurements may be misleading or even impossible to conduct validly, even on analog equipment, and they may not fully confirm all audible impairments. Traditional total harmonic distortion figures are not even adequate for a simple amplifier, where the relative levels of harmonics of increasing order need weighting to agree with their subjective annoyance, and recent preoccupations with transient intermodulation distortion (TID) or slew rate necessitate special measurements.

With more complex links in the chain, the interpretation of specifications or the making of measurements to check them can lead to great ambiguity, which may in turn lead to incorrect adjustments when setting up for subjective tests. Analog magnetic tape recording is a good example of such ambiguity, and since listening panels all around the world are busily comparing analog and digital recorders and sometimes are deducing that the digital process introduces nasty noises, it is essential that even such simple concepts as peak level are properly understood.

The peak level that can be handled by a digital system is precisely defined to a fraction of a decibel by the chosen coding parameters. Attempts to feed higher levels than this into the coder will usually lead to severe clipping effects or the operation of some protective circuitry. Overloads lasting significantly less than 1 ms may provoke these serious distortions, that is, within the overshoot period of any conventional protective limiter and outside the resolution of conventional level indicating meters. Only certain types of test program will contain these transient effects, but they must be explored.

By contrast the generally accepted peak level on analog tape is set around the 1–2% total harmonic distortion point, a figure that is audibly acceptable because the harmonics are "musically" related and decrease rapidly in level with increasing order.

The comparison is further confused by the peculiar behavior of the magnetic tape process at high frequencies. The frequency response of analog tape machines is normally measured at a level 10–20 dB below the chosen peak level to avoid confusing the issue with the effect of "crushing," the loss of high-level high-frequency components. Until recently the effect was considered inaudible or at least insignificant on program material, and so this rather dishonest low-level response has been accepted as valid. Of course, high-frequency crushing provides a subtle, mellow modification of the signal at just those high levels when the ear is getting ready to flinch at the onset of severe harmonic distortion or even clipping in other links in the chain, which may explain its widespread acceptability. Indeed tape crushing is audibly a very elegant way to avoid overloading later links, such as FM radio, where preemphasis has been adopted on the basis of out-of-date spectral analysis. In the absence of tape crushing it is necessary to adopt more obtrusive electronic means of protection, such as limiters which must take account of preemphasis. But the point at issue is that in a comparison between analog and digital recording the lack of high-frequency crushing on the digital route may cause critics to accuse it of "hard" or "metallic" quality.

The very brief transient peaks already mentioned will usually be rounded off or passed without much distress by the analog process, but must be seriously considered in any digital system. Program levels are usually controlled with reference to either a vu meter or a peak program meter (PPM), which is more strictly a quasipeak device. The vu meter is sometimes claimed to give readings which agree more nearly with subjective impressions of loudness than the PPM, but it underreads by 5–10 dB on different types of program material. The integration time constant of the BBC PPM was set many years ago at a value that ignores very-short-duration peaks of the order of 1–2 ms, as these did not produce significant distortion on the "limiting" links then existing, such as AM transmitters and disk recorders. Such links can be adequately protected by limiters with similar attack times (about 1–5 ms), and the distortion generated is seldom disturbing. But digital systems cannot comprehend the word "overshoot" and so must be preceded by a device to prevent even the briefest excursion beyond the defined peak. In the absence of an overshoot-free limiter, it is necessary to use a level indicator that will register peaks of only hundreds or even tens of microseconds. The effect of adopting such a true peak indicator instead of a PPM is to reduce the average level by some 6 dB or more, and if it replaces a vu meter, the reduction may be more than twice this figure. In other words, the top two or three bits of the digital coding are unused for 99% of the time.

While this neglect may be considered acceptable for a recording process with nominal 16-bit coding, it is quite out of the question in both bandwidth and economic terms for telecommunication links, such as our PCM distribution system with 13-bit or, more recently, 14-bit coding. So the BBC retained the PPM as a very realistic level control reference, but preceded the A/D converter with a specialized limiter, which incorporates two significant advantages over conventional limiters. By using an analog delay line of some 300 μs combined with a 250-μs attack time, it can perform any necessary gain reduction before a transient arrives at the variable gain element. This not only prevents any overshoot, but avoids almost all the distortions normally inevitable with fast attack times. The second feature is to allow for the fact that both the final FM transmitter and the PCM link itself use preemphasis, not by incorporating preemphasis weighting in the side chain which would cause broadband gain ducking effects, but by including a second stage giving variable deemphasis. Thus only those high-level high frequencies which would need limiting because of preemphasis are actually reduced. It is a very unobtrusive process, rather similar to tape crushing, but rather embarassing to admit to in the new digital era. Moreover as other links in the chain are improved, we are becoming aware of the audibility of the fast attack on certain types of program, so we are working to modify the process.

Having studied at length the precautions necessary to ensure strict comparability of peak level and frequency response in both objective and subjective comparison, it is necessary to consider also the question of background noise as a possible source of unconventional impairments.

The normally quoted noise figure for an analog tape system, to relate to the rather arbitrary peak signal level, is the noise in the absence of a signal. But tape introduces quite significant inherent signal-modulated noise, while so-called "noise-reduction systems" actually generate this effect. It is surprisingly seldom considered a serious problem, though it is the main reason for the BBC's decision not to adopt any of the more powerful noise-reduction systems. The rather similar audible effects of quantizing noise or granular distortion in the digital process are rightly and widely publicized as a serious problem, even if they are only audible on very-low-level piano notes. These level-dependent noises may be measured with slight difficulty, but their numerical values do not always correlate well with subjective annoyance. Depending on the noise spectrum, the defects can sound more like a defective musical instrument, or even a poor performance, than an identifiable technical fault. Unlike constant hiss or "classic" harmonic distortion, such impairments may well distress the musician far more than the purely technical listener.

In assessing a new digital system it is usually necessary to make two different comparisons, with the equivalent analog system and with a theoretically perfect link. The former usually leaves the digital system a clear winner, subject to some of the considerations already raised, but the latter allows a very detailed exploration of the coding and decoding processes in the absence of any convenient analog masking effects.

In contemplating the introduction of our 13-bit PCM distribution system ten years ago, we were in no doubt of its superiority over the limited-bandwidth, high-noise, variable-phase-shift line system it was to replace. But we also compared the output of the PCM decoder with the actual source material, first on all types of recorded material and finally on a live piano. On just a few piano notes, at a particular level through the system, we thought we could detect an effect that was not present at the system input. It could easily have been a minor mechanical vibration in the piano itself and would certainly have been obscured in an analog recording process. We persisted until we found a means of measurement which confirmed what we thought we could hear.

The cause eventually proved to be a very minor inaccuracy in setting up the 64-to-1 ramp currents in the A/D converter, which was not detectable with existing measuring equipment. This is a perfect example of the need to believe critical listeners, even when they disagree with measurements. Luckily we identified and cured this problem before the equipment went on air.

When the PCM system did enter service and at long last replaced the tenuous links of Post Office lines from London to Scotland, we received a new type of complaint. For 50 years the listeners there had heard their programs on a 5-kHz or perhaps 8-kHz bandwidth, with telephone dialing and speech crosstalk perhaps 50 dB below peak signal. Now that they had a response flat to 15 kHz and almost 70 dB signal-to-noise ratio, they deluged us with complaints about tape hiss, ventilation rumble, and the announcer's clock ticking. Digital tape, and in another sense digital clocks, should shortly remove these two sources of annoyance, but acoustic noise is a very big problem. The problem is not helped by the fact that digital recording systems do not generally provide the extreme low-frequency rolloff inherent in most analog tape machines. To judge from many of the commercial "digital" disks I have heard, the BBC is not alone in using concert halls and studios with an elderly air-conditioning plant or built too close to a railway or highway. As digital techniques replace analog links with their convenient masking noises, we are having to contemplate improving our studio background noise criteria by some 8–10 dB.

So quite apart from the many manufacturers of audio equipment represented at this conference, who are hoping that the digital revolution will keep the production lines rolling, the various acoustic consultants of the world together with their suppliers of antivibration mountings, duct attenuators, and solid-mass concrete must all be looking forward to this profitable side effect of the new world of digital audio.

THE AUTHOR

David Stripp joined the BBC as a junior control room engineer in 1955, after qualifying with a Bachelor of Science degree in mathematics and physics at London University. From 1958 to 1972 he supervised the progressive growth of stereo techniques throughout BBC Radio.

In his present post he is resonsible to the chief engineer of Radio Broadcasting for all sound quality considerations, from studio acoustics through the choice of microphones, monitoring loudspeakers, and processing, distribution and modulating equipment, with particular emphasis on adequate subjective assessments.