BBC RD 1974/11

1968
THE QUEEN'S AWARD
TO INDUSTRY

RESEARCH DEPARTMENT

REPORT

DIGITAL SOUND SIGNALS: subjective effect of timing jitter

W.I. Manson, B.Sc.(Eng.)

Research Department, Engineering Division
THE BRITISH BROADCASTING CORPORATION March 1974

BBC RD 1974/11
UDC 534.86
621.376.56

DIGITAL SOUND SIGNALS: SUBJECTIVE EFFECT OF TIMING JITTER

W.I. Manson, B.Sc.(Eng.)

Summary

In a pulse-code modulation (p.c.m.) sound system, the decoding process at the receiving terminal to reconstitute the original analogue signal should ideally be carried out at precisely the same regular rate as the coding process at the sending terminal. However, in a practical system, the timing of the decoding process may be subject to fluctuation – that is, timing errors or timing 'jitter' may occur – and the quality of the reproduced sound may consequently be impaired.

Apparatus has been built to simulate the effect of timing errors in a sound p.c.m. system. Subjective tests have been carried out to enable tolerance limits to be estimated for both sinusoidal and random jitter on high-quality monophonic programmes. Tolerances are proposed for jitter on a practical sound system, based on the results of the present work and other previous investigations. The tolerance varies from a timing jitter amplitude of 3·5 milliseconds (r.m.s.), for sinusoidal variation over a period of 50 seconds or more, to 35 nanoseconds for jitter frequencies of 2 kHz or higher. Comparable figures apply to timing jitter with a random-noise variation.

Issued under the authority of

Head of Research Department

Research Department, Engineering Division,
BRITISH BROADCASTING CORPORATION
March 1974
(EL-87)

DIGITAL SOUND SIGNALS: SUBJECTIVE EFFECT OF TIMING JITTER

DIGITAL SOUND SIGNALS: SUBJECTIVE EFFECT OF TIMING JITTER

W.I. Manson, B.Sc.(Eng.)

1. Introduction

In a p.c.m. sound system, the analogue signal at the sending terminal is normally sampled and coded at a regular rate. The decoding process which reconstitutes the analogue signal from the digital information at the receiving terminal should ideally be carried out at the same regular rate. In practice, however, the decoding process may be subjected to timing errors as a result of, for example, noise on the bearer circuit, the transfer of signals from one level of multiplexing to another in a communications network or to variation of clock phase arising in the process of data regeneration.

In a single p.c.m. link, timing errors, if comparable in magnitude with the bit-period, would be likely to cause bit errors. However, a different situation can arise when, in order to maintain immunity from noise on the bearer circuit, the p.c.m. data is regenerated repeatedly in the course of transmission. In these circumstances, it is possible for small errors at each regeneration process to combine to give timing errors in excess of the bit period without causing associated bit errors.

The process of regenerating p.c.m. data is normally controlled by a local clock signal, commonly derived from a high-Q resonant circuit driven by the incoming p.c.m. data itself. The phase of the clock signal may be affected by the bit-pattern density, so that variations in this pattern may give rise to variations in the clock timing, leading to timing errors or timing 'jitter'. Where each data regenerator in the chain is subjected to the same data pattern, the effect is systematic and the resulting timing jitter tends to be cumulative, at least at low jitter frequencies.[1] At high jitter frequencies the use of high-Q resonant clock circuits in the data-regeneration process imposes a limit on the amplitude of accumulated jitter. The maximum rate-of-change of clock phase is determined by the bandwidth of those resonant circuits; the amplitude of rapid changes of clock phase, and hence of the timing jitter at the higher frequencies, is therefore restricted. Thus, in a p.c.m. link containing a large number of regenerators, the amplitude of the systematic jitter may be relatively large at low jitter frequencies, but will tend to decrease at higher frequencies.

Timing jitter, adjustable over the range to be explored, could not readily be produced in the laboratory by modification of any available digital sound equipment; apparatus, described in the following Section, was therefore built to simulate the effect of jitter for the present investigation.* It was so arranged that the amplitude of the timing jitter introduced was proportional to an applied 'jitter control signal', the effects of different kinds of jitter could thus be studied simply by changing the nature of this control signal.

The investigation into the subjective effect of timing jitter on high-quality monophonic sound signals was carried out with sinusoidal jitter of various frequencies, and with random jitter. For the latter, the jitter control signal comprised a 30 Hz to 16 kHz band of random white noise, applied through a low-pass filter with a 6 dB-per-octave cut-off rate to simulate the restriction of jitter amplitude at high frequencies (due to repeated regeneration) as discussed above. Tests were carried out with the low-pass filter set, in turn, to different cut-off frequencies or 'break-points'.

The investigation was carried out in two groups of tests, separated by a period of about 18 months; the first group was devoted predominantly to sinusoidal jitter, and the second to random jitter. For the second group of tests the opportunity was taken to use programme material, not available at the time of the first tests, which provided a slightly more critical condition. In addition programme low-pass filters required in the apparatus were improved slightly to provide a bandwidth more representative of current practice. Some tests using sinusoidal jitter were therefore included in the second group to check the effect of these changes.

Finally, tolerance limits for timing jitter as a function of the frequency of jitter are suggested in this report, based partly on the results of the investigation described, partly on previous work by other authors and partly on typical standards for wow and flutter in sound-signal recording systems.

2. Instrumentation

Fig. 1 shows in block schematic form the experimental apparatus built to simulate timing jitter in a digital sound system. The sound signals to be subjected to jitter were first passed through a low-pass filter, to exclude components above half the sampling frequency, then to a sample-and-hold circuit working at a regular, 32 kHz, rate. The resulting waveform ('box-car' waveform) was re-sampled using clock pulses whose timing was perturbed by the jitter

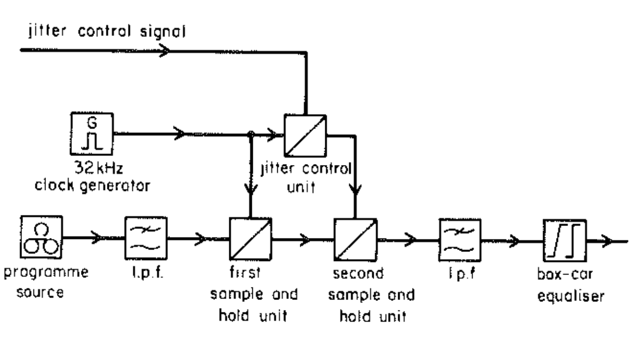

Fig. 1 - Apparatus used for tests

---

* The experimental apparatus was constructed by G.C. Wilkinson and R.L. Deane, who also conducted much of the subjective investigation.

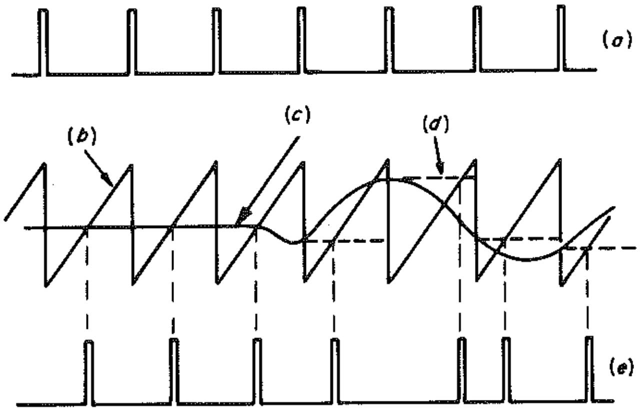

Fig. 2 - Timing diagram of apparatus to simulate timing jitter in a digital system
(*a*) Regular, 32 kHz, control pulses for first signal sampling process (*b*) Ramp waveform (*c*) Applied jitter control signal
(*d*) Box-car waveform derived from jitter control signal (*e*) Control pulses for second signal sampling process

control signal, and the box-car waveform from the second sample-and-hold unit was passed, through a further low-pass filter and an equaliser, to the monitoring loudspeaker. In both the signal-sampling processes the level of each sample was held until the beginning of the next.[2]

The timing of the pulses controlling the second sampling process could be perturbed by up to about ±15 μs, relative to a mean position in the middle to the hold period of the first sampling process. Fig. 2 shows the method used to derive, from a regular clock pulse series, a second series with timing jitter superimposed, delayed, on average, by half a clock period. The regular pulses (*a*), used in the first signal sample-and-hold process, are used also to trigger a ramp waveform, (*b*), and, further, to sample-and-hold an applied jitter control signal, (*c*), to produce the box-car waveform, (*d*). The control pulses, (*e*), required for the second signal-sampling process are generated as the rising ramp voltage (*b*) crosses waveform (*d*).

For the first series of tests, 14 kHz low-pass filters were used in the programme circuit; these were replaced by 15 kHz low-pass filters for the second part of the investigation, as discussed in the previous Section.

3. Subjective investigation

The subjective effect of timing jitter depends greatly on the nature of the programme being conveyed. A range of programme material was therefore assessed by a few experienced listeners before the first series of tests, and two test passages, one of piano and one of glockenspiel, were selected as representative of the most critical material available. For the second series of tests two newly-acquired passages of programme, again one of piano and one of glockenspiel, were considered slightly more critical and were selected for use.

The subjective investigation was carried out using a wide-range monitoring loudspeaker in a listening room having a volume of about 85 cubic metres and a mean reverberation time of about 0·3 second.

The first group of tests was carried out by only six listeners; in the second group the effects of random jitter was judged by thirteen listeners, but some of the repeat tests with sinusoidal jitter were made with only seven listeners. All of the listeners had some experience of assessing the quality of sound programmes — most indeed had considerable experience.

The tests were carried out by the listeners, one at a time. The listener was provided with a control so that he could adjust the listening level as desired. He was also given two further controls — one enabling him to alter the level of the jitter control signal, and the other to interrupt it completely so that he could readily compare impaired programme with programme not subjected to jitter. He was asked, for each test condition, to set the level of the jitter control signal to give the threshold of perceptibility of jitter, making use of the facility for direct comparison with unimpaired programme as desired.

4. Results

4.1. Sinusoidal jitter

Figs. 3 and 4 show, for test passages of glockenspiel and piano respectively, the average results obtained for sinusoidal jitter in the first series of tests. Here, the r.m.s. amplitude of jitter required for the threshold of perceptibility is plotted against the frequency of the jitter control signal. The standard deviation of the observations is indicated for each plotted result.

For both the test passages in the first series of tests the listeners were relatively tolerant of jitter at low audio frequencies and less tolerant for higher frequencies in the range from about 2 kHz to 5 kHz for glockenspiel and 5 kHz to 10 kHz for piano. For the most critical jitter frequencies the mean level of jitter for the threshold of perceptibility was of the order of 200 ns r.m.s. for both glockenspiel and piano.

Figs. 3 and 4 show also the results of the check

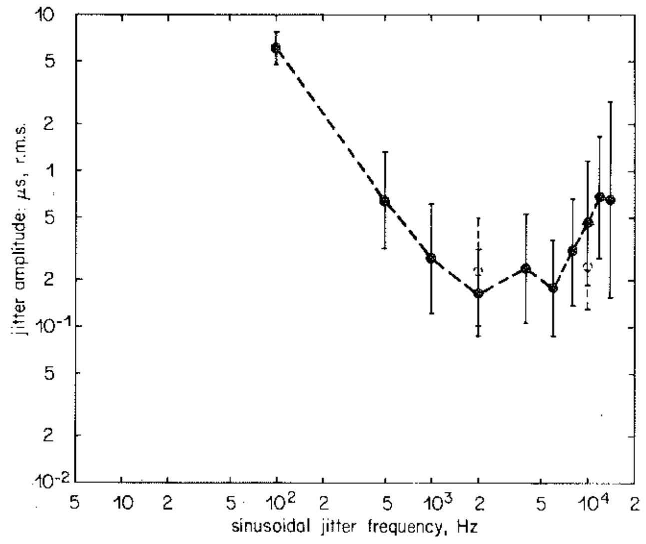

Fig. 3 - R.M.S. amplitude of sinusoidal timing jitter for the threshold of perceptibility on glockenspiel test passage, mean results

First series of tests --●--●-- Second series of tests ○ ○

±one standard deviation of observations I

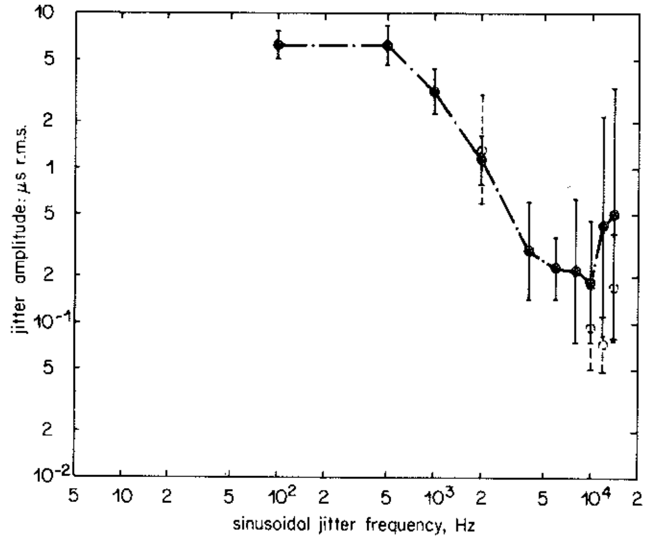

Fig. 4 - R.M.S. amplitude of sinusoidal timing jitter for the threshold of perceptibility on piano test passage, mean results

First series of tests -●-·-●-·-● Second series of tests ○ ○

±one standard deviation of observations I

measurements carried out using sinusoidal jitter in the second group of tests. The results for 2 kHz jitter are in reasonable agreement with those of the earlier investigations but considerable differences are indicated for some of the tests with higher jitter frequencies (10 kHz for glockenspiel and 10, 12 and 14 kHz for piano) where the standard deviations were large in the earlier investigation. If the most critical points on the curves were taken as a criterion, the results would indicate that, for the same impairment, the new, more critical, test material required the jitter amplitude, to be less than half that required for the earlier test material. However, if the amplitude of jitter estimated to give impairment perceptible to say, only 5% of the observers, were to be taken as a criterion, the tolerance limits estimated from the two sets of data would, in fact, be very similar.

4.2. Random jitter

Figs. 5 and 6 show the average test results for the investigation using random jitter control signals for glockenspiel and piano respectively; the r.m.s. amplitude of random jitter for the threshold of perceptibility being plotted as a function of the cut-off frequency of the 6 dB-per-octave low-pass filter used to shape the noise spectrum of the jitter control signal. Again the standard deviation of the

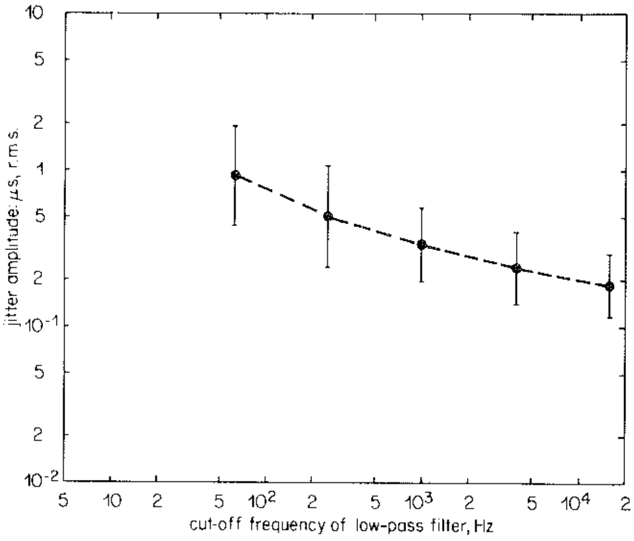

Fig. 5 - R.M.S. amplitude of random timing jitter for the threshold of perceptibility of glockenspiel passage, mean results

± one standard deviation of observations I

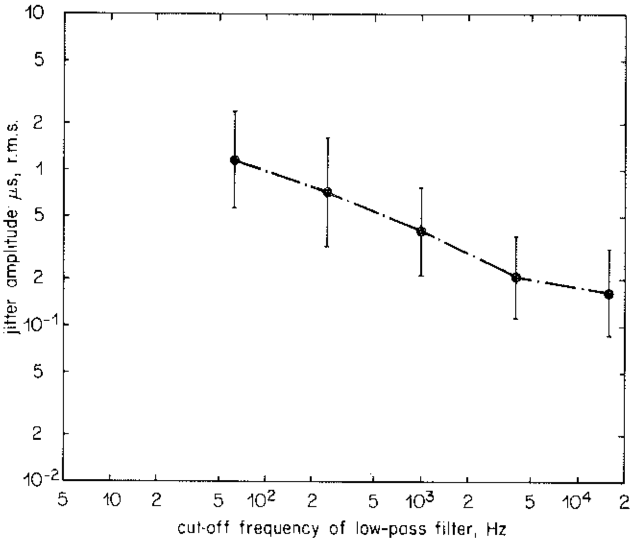

Fig. 6 - R.M.S. amplitude of random timing jitter for the threshold of perceptibility on piano passage, mean results

±one standard deviation of observation I observations is shown for each plotted result. The results indicate that the most critical jitter control signal investigated is the unshaped 30 Hz to 16 kHz band of white noise, where the average result is in the range 150 ns to 200 ns r.m.s. jitter for each test item. As the cut-off frequency of the noise spectrum shaping filter is reduced, the level of the random jitter control signal for the perceptibility threshold, measured at the output of the shaping filter, eventually rises at a rate of approximately $\sqrt{2}$ in amplitude for every halving of the filter cut-off frequency. This result is of interest because it represents the law for which noise power per unit bandwidth above the cut-off frequency of the filter is kept constant; it suggests that, over the range of conditions examined, the listeners were influenced mainly by the higher-frequency components of the random jitter. The effect on the noise spectrum of altering the cut-off frequency of the spectrum shaping filter is discussed further in the Appendix.

5. Tolerance limits for timing jitter

5.1. Sinusoidal jitter

With the apparatus specially constructed to introduce timing jitter for the present work, the scope of the investigation was restricted by the inherent limitation on jitter amplitude to about ±15 μs peak. No information could be obtained for jitter frequencies less than some 50 Hz to 100 Hz, since jitter at lower frequencies, even at the maximum available amplitude, produced no perceptible effect. However, relevant data can be derived from an earlier investigation (1955) by A. Stott and P.E. Axon[3] into the subjective effect of 'wow' and 'flutter' in recording systems. In this work the effect of frequency modulation of sound signals was investigated for modulation frequencies from about 0·5 Hz to 10 kHz, mechanical movement of record/replay heads, used with a drum recorder, being used to provide the modulation at the lower frequencies.

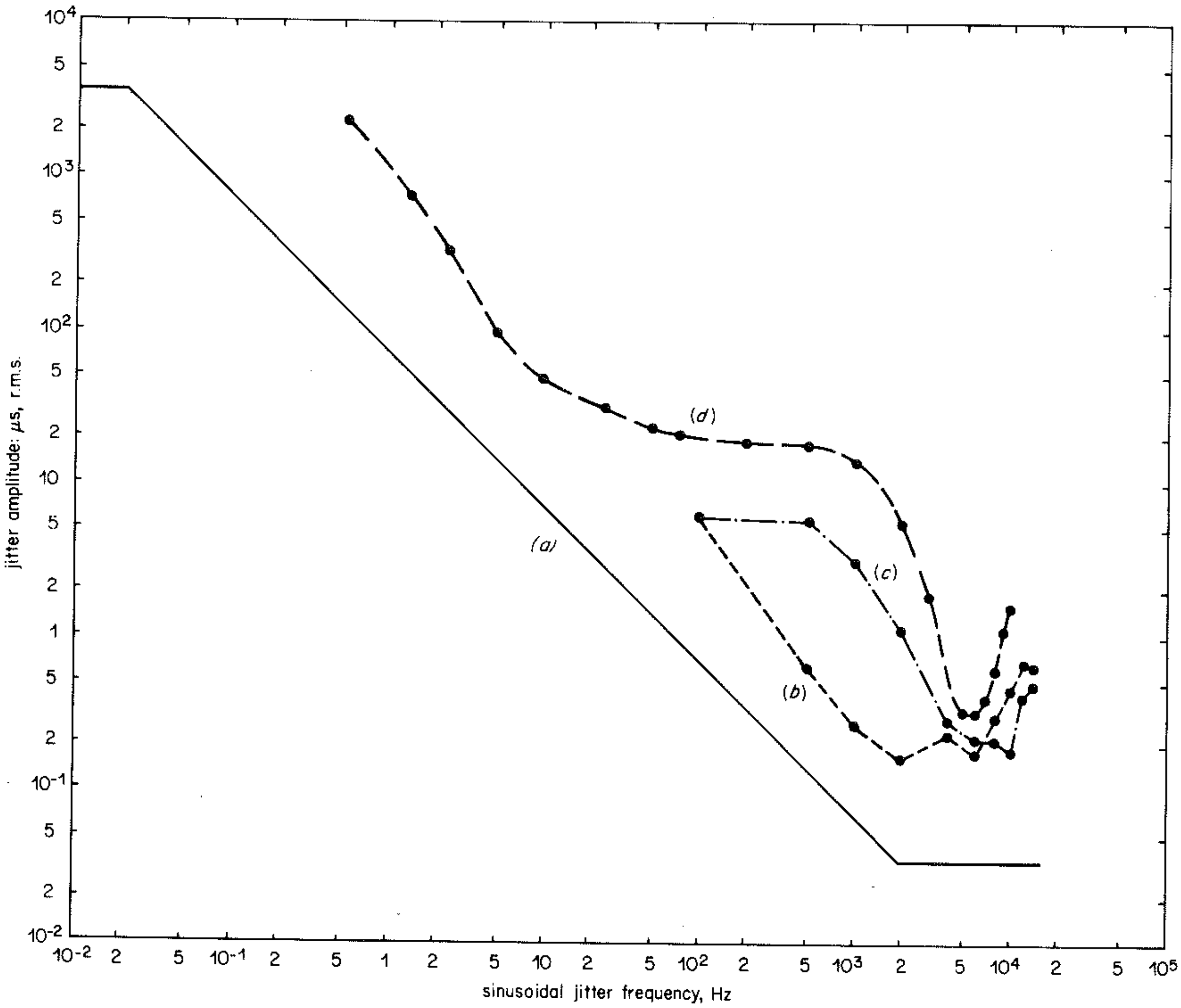

Fig. 7 - Proposed maximum allowable r.m.s. amplitude of sinusoidal timing jitter in high-quality sound signals

(*a*) Proposed tolerance limit ———— (*b*) Threshold, mean results, glockenspiel, 1971 -------

(*c*) Threshold, mean results, piano, 1971 —·—·— (*d*) Threshold, mean results, piano, Stott and Axon, 1954 — — — —

For jitter at very low frequencies, tolerance limits may be estimated from the requirements to maintain reasonable synchronism between sound and vision in a television broadcast when the two signals are conveyed by separate p.c.m. multiplexes.

The above data, together with the results of the present work, are summarised in Fig. 7. They have been used to construct a proposed tolerance curve for sinusoidal jitter, shown in Fig. 7, curve (*a*).

5.1.1. Sinusoidal jitter at the higher frequencies

The results presented in Figs. 3 and 4 show the jitter amplitude, for impairment at the threshold of perceptibility, for jitter frequencies above about 100 Hz; these results correspond to impairment that is perceptible to about 50% of experienced listeners. From the distribution of the results for the tests it is possible to estimate tolerance limits for which impairment would be perceptible to a smaller proportion of listeners.

A possible tolerance limit for jitter at the higher audio frequencies might be based on a level of impairment that is perceptible to no more than, say, 5% of listeners. On this basis, it can be estimated that the amplitude of sinusoidal jitter should not exceed about 35 ns r.m.s. in the more critical frequency range explored, above, say, 2 kHz; this criterion is represented by that part of curve (*a*), Fig. 7, lying above this frequency. (In fact the earlier results for piano indicate that impairment at this limit would be perceptible to slightly more than 5% of the listeners for jitter at 12 and 14 kHz. However, in view of the very large standard deviations for these points, they have been disregarded and preference given to the results of the later tests.)

The results for the glockenspiel, Fig. 3, indicate that listener tolerance of jitter increases as the jitter frequency is reduced below 2 kHz, and follows roughly a $^1/f$ law in this region. The portion of the proposed tolerance curve, Fig. 7(*a*), between 500 Hz and 2 kHz represents this law – the limits set again corresponding to impairment perceptible to an estimated 5% of observers.

For reference, the mean results given in Figs. 3 and 4 for glockenspiel and piano respectively are repeated in Fig. 7, curves (*b*) and (*c*).

5.1.2. Sinusoidal jitter in the range 1 Hz to 100 Hz

As indicated in a previous Section, the present investigation of sinusoidal jitter was restricted to jitter frequencies above about 100 Hz; in arriving at a proposed tolerance for jitter at lower frequencies, use was made of the results of earlier work.

Fig. 7(*d*), shows the results obtained by Stott and Axon for sinusoidal jitter on a piano test passage. Their complete composite curve of 'threshold, percentage frequency fluctuation', peak, plotted against fluctuation frequency down to about 0·5 Hz, has here been re-drawn in terms of r.m.s. timing jitter, taking r.m.s. jitter as $(w\ 10^4)/(2\sqrt{2}\ \pi f_w)$ μs, where w = % peak wow, and f_w = wow frequency.

Over the region of frequency common to the recent investigation, Fig. 7(*c*), and the earlier one by Stott and Axon, Fig. 7(*d*), both for a piano test passage, there is a substantial measure of agreement in the shape and even in the level of the curves.

The curve due to Stott and Axon, extending down to about 0·5 Hz, again suggests a $^1/f$ law, indicating that, for constant impairment at the threshold of perceptibility, the permissible jitter amplitude is, very roughly, inversely proportional to jitter frequency. Since both the present investigations and the earlier work by Stott and Axon both suggest an approximate $^1/f$ law it seems appropriate that a tolerance limit for sinusoidal jitter in the low- to medium-frequency range should also follow a $^1/f$ law.

The $^1/f$ portion of curve (*a*) Fig. 7 proposed in the previous section for jitter frequencies between 500 Hz and 2 kHz corresponds to a 'flutter' of 0·044% r.m.s. Stott and Axon state that 0·1% r.m.s. wow has been accepted as a good criterion for many years, Moir[4] indicates that equipment of the highest quality may exhibit speed variations in the region of 0·05%, Olson[5] suggests a limit of 0·4% r.m.s., while a specification of about 0·1% peak (0·07% r.m.s. for a sinusoidal variation) seems typical of modern, high-quality audio tape recorders.

The figure of 0·044% r.m.s. tolerance of flutter – suggested as a possible tolerance in the region 500 Hz to 2 kHz as a result of the present work – is thus in reasonable agreement with accepted standards for lower frequencies, and seems appropriate as a tolerance limit over a wider range of frequencies. This limit for flutter, 0·044% r.m.s. – corresponding to a sinusoidal jitter amplitude of about $(100)/(\sqrt{2}\ f_j)$ μs r.m.s., where f_j is the jitter frequency – is indicated by the full $^1/f$ region of the tolerance curve, Fig. 7(*a*) extending downwards from 2 kHz till the tolerable jitter magnitude reaches a limit set by factors to be discussed in the following Section.

5.1.3. Sinusoidal jitter at the lowest frequencies

For jitter at the very low frequencies, tolerance limits may be set by the need to maintain acceptable synchronisation between sound and vision in a television broadcast when the two signals are transmitted in different p.c.m. multiplex systems. It has been judged that timing errors of 70 to 140 ms may cause 'just perceptible impairment' to 50% of listeners.[6] If, arbitrarily, a maximum timing difference between sound and vision of 10 ms were to be accepted as a tolerance limit, this could be interpreted as an absolute limit for jitter of 5 ms peak (i.e. about 3·5 ms r.m.s.) for both the sound and the vision channels.

This criterion is represented by the section of the tolerance curve, Fig. 7(*a*), below 0·02 Hz.

5.2. Random jitter

Figs. 5 and 6 showed the average results for random

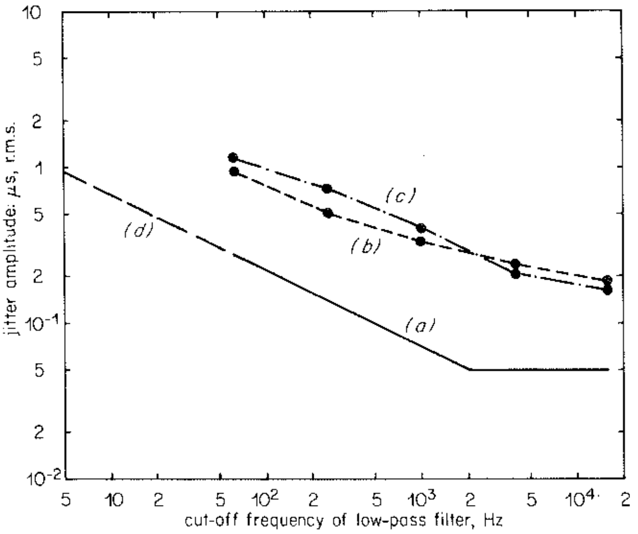

Fig. 8 - Proposed maximum allowable r.m.s. amplitude of random timing jitter in high-quality sound signals

(*a*) Proposed tolerance limit ————

(*b*) Threshold, mean results, glockenspiel, 1973 - - - - - - -

(*c*) Threshold, mean results, piano, 1973 —·—·—·—

(*d*) Tentative tolerance limit for break-point in noise spectrum below 63 Hz — — — — —

NOTE: For very low frequencies, see text

jitter giving impairment at the threshold of perceptibility, using a jitter control signal comprising 30 Hz to 16 kHz white random noise subjected to a 6 dB/octave high-frequency cut-off commencing at various frequencies. As already indicated for sinusoidal jitter, it is possible to estimate tolerance limits such that impairment is perceptible to a given small proportion of the listeners. Fig. 8(*a*) shows a possible tolerance limit curve, related to the cut-off frequency of the noise-spectrum shaping filter, for which the data indicates that impairment would be perceptible, on the test items used, to no more than 5% of listeners. The curve indicates a tolerance limit of about 50 ns r.m.s., provided that the filter cut-off frequency exceeds some 2 kHz, the limit rising at a rate of $\sqrt{2}$ for every halving of the cut-off frequency down to about 50 Hz.

For reference, the average curves for glockenspiel and piano are repeated in Fig. 8, curves (*b*) and (*c*) respectively.

Investigation of the effects of lower-frequency jitter were restricted, for random jitter as for sinusoidal jitter, by limitations in the jitter amplitude possible with the apparatus used for the tests. However, as already noted in Section 4.2, the results for random jitter with the form of spectrum used, suggests that the effects of high-frequency components predominate. If it is assumed that this applies also when the breakpoint in the jitter spectrum is set at still lower frequencies, then the tolerance curve, Fig. 8(*a*), may be extended downwards to give the tentative curve, Fig. 8(*d*).

An ultimate tolerance limit for sinusoidal jitter at very low frequencies was set at 3·5 ms r.m.s. (i.e. 5 ms peak) from considerations of acceptable synchronism of sound and vision signals transmitted separately. From the same considerations, the limit for random jitter of very low frequencies should also be set at 5 ms, peak.

6. Conclusions

The effects of timing errors in a high-quality monophonic p.c.m. sound-system have been simulated in the laboratory and assessed subjectively; tolerance limits have been suggested for both sinusoidal and random jitter, based partly on the present work, partly on earlier work and on accepted tolerances for wow and flutter in sound recordings. The tolerances suggested thus apply for a complete system and it may be appropriate to set closer limits if a number of systems, each of which may introduce jitter, are connected in cascade.

Over much of the frequency range tolerance of sinusoidal jitter increases as the jitter frequency is reduced. Tolerance limits for sinusoidal jitter for which impairment on critical programme should be perceptible to less than 5% of listeners, are suggested as follows:— a maximum jitter amplitude of 35 ns r.m.s. for jitter frequencies above 2 kHz, a limit of 3·5 ms r.m.s. at very low frequencies, below 0·02 Hz, (set by the need to maintain adequate sound and vision synchronism in a television programme of which the two signals are sent separately), and a tolerance inversely proportional to jitter frequency in the intermediate frequency range from 0·02 Hz to 2 kHz.

For jitter having a random, white noise, spectrum extending from 30 Hz to 16 kHz, it is estimated that impairment on critical programme would be perceptible to less than 5% of listeners provided the jitter amplitude is no more than 50 ns r.m.s. Tolerance of random jitter tends to increase as the high-frequency content of the jitter spectrum is reduced. The results obtained suggest that, for the form of noise spectrum used in the investigation — 30 Hz to 16 kHz band of random white noise, subjected to 6 dB/octave high frequency cut-off introduced at a range of break frequencies — a $\sqrt{2}$ increase in jitter amplitude can be allowed for every halving of the breakpoint frequency.

The relaxation of tolerance indicated for low jitter frequencies is fortunate, since the process of data regeneration in p.c.m. multiplex systems can give rise to a jitter spectrum in which low frequencies predominate.

The present work was carried out using unprocessed sound signals — no assessment was made of the effects of jitter on sound signals subjected, for example, to high-frequency pre- and de-emphasis or to digital companding; it is known from brief tests that in these circumstances the tolerances would probably be affected only marginally. However, the most stringent tolerances for jitter suggested in this report are more than an order greater than those recommended elsewhere[2] for vision circuits; it seems safe to assume therefore that, where tolerance limits are appropriate for the transmission of vision signals, they should also be adequate for transmission of sound signals.

The investigation described in this report was carried out for monophonic programme only; it seems reasonable to assume, however, that the tolerances suggested would apply also to the two channels of a stereophonic programme, carried in the same p.c.m. multiplex system, provided that differential jitter between the two channels is negligible. The effect of differential jitter on a stereo programme has not been investigated.

Although the work described was carried out with p.c.m. sound systems principally in mind, the results obtained apply equally to any process which introduces timing perturbation of sound programme signals; indeed, use has been made in this report of existing work relating to 'wow' and 'flutter' in sound recording.

7. References

1. BYRNE, C.J., KARAFIN, B.J. *and* ROBINSON, D.B. 1963. Systematic jitter in a chain of digital repeaters. *Bell Syst. tech. J.*, 1963, **62**, 6, pp. 2679 – 2714.
2. DEVEREUX, V.G. 1971. Pulse code modulation of video-signals: subjective tests on acceptable limits for timing jitter in the decoded analogue samples. BBC Research Department Report No. 1971/42.
3. STOTT, A. *and* AXON, P.E. 1955. The subjective discrimination of pitch and amplitude fluctuations in recording systems. *Proc. IEE*, 1955, **102B**, 5, pp. 643 – 656.
4. MOIR, J. 1958. High quality sound reproduction. London, Chapman and Hall Ltd.
5. OLSON, H.F. 1972. Modern sound reproduction. New York, Van Nostrand.
6. CMTT Report 412-1. CCIR XIIth Plenary Assembly, New Delhi, 1970, p. 259.

Appendix

For the investigation into the effects of random jitter, the control signal used comprised a 30 Hz to 16 kHz band of random white noise, passed through a single-stage Resistance-Capacitance (R-C) low-pass filter set at one of a number of switched cut-off frequencies.* The results of the subjective investigation suggest that, for equal impairment, the noise power, at the filter output, may be raised 3 dB if the filter cut-off frequency is halved.

A possible explanation for the above law may be obtained from the following consideration of the noise spectra concerned, where, for simplicity, a wideband source of random white noise is assumed.

For a constant power of the applied wideband noise, the total power of the noise after R-C filtering is proportional to the filter cut-off frequency, f_o. The power per unit frequency, at frequency f, is proportional to

$$\frac{1}{1+\left(\frac{f}{f_o}\right)^2}\text{, tending to}\left(\frac{f_o}{f}\right)^2\text{ when } f \gg f_o.$$

If f_o is halved, the total power of the filtered noise is also halved (reduced by 3 dB) but the power per unit frequency at frequency f (where $f \gg f_o$) is quartered (reduced by 6 dB). However, if now the applied wideband noise is raised in level by 6 dB, the power per unit frequency at frequency f, is restored to its original value, while the total power of the filtered noise is raised to 3 dB above its original value.**

Thus the results of the present investigation suggest that the subjective effects were determined largely by high frequency noise components, which were being kept substantially constant for equal impairment, and that the increasing amplitude of lower-frequency components had relatively little effect.

---

* For this Report the cut-off frequency, f_o, of the low-pass filters used to shape the noise spectrum, is defined as $f_o = 1/(2\pi RC)$, where R and C are the filter elements.

** Acknowledgement is due to T.A. Moore for helpful discussion during this work.

SMW/MS

Printed by BBC Research Department, Kingswood Warren, Tadworth, Surrey