inkPhysics

ANSWER BOOK

Data communication

Electronics · Year 13

AQA 3.13.4

17 questions · 51 marks

NAVY EDITION

SECTION A · Warm-up

A1 Baseband transmission sends a signal in its own natural range of frequencies, as a telephone landline sends speech down a private pair of wires (1). Radio needs a carrier because every station shares the same air, so each is given its own carrier frequency and a tuned circuit at the receiver can select one and reject the rest; and because an efficient aerial must be a respectable fraction of a wavelength long, which a kilohertz signal never allows (1). Answering 'so it travels further' alone is the standard error: the two real reasons are sharing the spectrum and aerial length. [2]

A2 Three: the carrier itself at f_c, a lower sideband at $f_c - f_M$ and an upper sideband at $f_c + f_M$ (1) (1). The audio is not sent at its own frequency; it appears as the pair of new frequencies flanking the carrier. Counting the sidebands once instead of twice, and so quoting two frequencies, is the standard sideband slip. [2]

A3 Bandwidth = $2f_M$ (1), where f_M is the highest audio frequency carried (1). The factor of two is there because a sideband sits on each side of the carrier. Quoting f_M alone, and so halving the bandwidth, is the most common lost mark in this topic. [2]

A4 Bandwidth = $2(\Delta f + f_M)$ (1), where Δf is the frequency deviation, the greatest swing of the carrier either side of its unmodulated frequency, and f_M is the highest audio frequency carried (1). Dropping the f_M term and writing $2\Delta f$ is the examiners' favourite error to punish here. [2]

A5 The sampling rate must be **greater** than twice the highest frequency present in the signal (1). Sampled more slowly than that, a high frequency masquerades as a lower one in the reconstructed signal, and the corruption cannot be undone afterwards (1). Writing 'twice the highest frequency' without the inequality, or sampling at twice the **average** frequency, are the standard errors. [2]

A6 Frequency-division multiplexing gives each user a permanent slice of the spectrum and every user transmits continuously on their own carrier, which is how the radio dial works (1). Time-division multiplexing gives each user the whole channel for a brief repeating time slot, interleaving the users in turn, which suits digital signals because samples are short pulses with dead time between them (1). Saying that TDM 'sends signals faster' is the standard error: it shares the same channel in time rather than in frequency. [2]

SECTION B · Standard

B1 Convert the carrier first: 1.53 MHz = 1530 kHz (1). Lower sideband = 1530 - 5.0 = 1525.0 kHz and upper sideband = 1530 + 5.0 = 1535.0 kHz (1). Bandwidth = 2 × 5.0 = 10.0 kHz (1), which is also 1535.0 - 1525.0. Mixing kHz with MHz is the trap: subtract 5.0 from 1.53 without converting and the sidebands land at 1.525 MHz and 6.53 MHz, nowhere near the carrier. Sidebands always sit close to the carrier, so check that before writing the answer down. [3]

B2 Bandwidth = $2(\Delta f + f_M)$ (1) = 2 × (5.0 + 3.5) (1) = 17 kHz (1). Using $2\Delta f$ alone gives 10 kHz, and dropping the factor of two gives 8.5 kHz; both are the standard errors, and both are marked wrong. The audio term belongs inside the bracket. [3]

B3 Twice the highest frequency is 2 × 20 = 40 kHz, and 44.1 kHz exceeds it, leaving margin for imperfect filters (1). Bit rate for one channel = 44 100 × 16 = 705 600 bits per second, so for two channels (1) the total is 1 411 200 bits per second, about 1.41 Mbit s^{-1} (1). Sampling at or below twice the highest frequency, and forgetting the second channel, are the two standard errors. [3]

B4 Number of levels = 2^{12} = 4096 (1). The levels span the range in 4095 steps, so one step = 5.0/4095 (1) = 1.22×10^{-3} V, about 1.22 mV (1). Each extra bit doubles the number of levels and halves the step, which is why quantisation error falls as bits are added. Dividing by 12 instead of by 2^{12} is the standard error. [3]

B5 Total bit rate = $30 \times 64 \times 10^3$ (1) = 1.92×10^6 bits per second, that is 1.92 Mbit s^{-1} (1). Each call is given the whole channel for a brief repeating time slot, its samples interleaved into the gaps between everyone else's (1). Describing the calls as sharing the bandwidth simultaneously is the standard error; that is frequency-division multiplexing, not time-division. [3]

B6 Convert the span first: 2.0 MHz = 2000 kHz (1). Number of stations = 2000/9 = 222.2 (1), so **222** stations fit, the fraction being no use to anybody (1). Dividing 2.0 by 9 without converting to the same unit gives 0.22 stations and is the standard error: keep kHz and MHz apart, and convert before dividing. [3]

B7 $\lambda = c/f = (3.0 \times 10^8)/(3.0 \times 10^3) = 1.0 \times 10^5$ m, that is 100 km, against $(3.0 \times 10^8)/(100 \times 10^6) = 3.0$ m for the carrier (1). An efficient aerial is a respectable fraction of a wavelength, so a baseband aerial is unbuildable while a VHF one is about a metre (1). Using the speed of sound rather than the speed of light is the standard error: these are radio waves, whatever the frequency. [2]

SECTION C · Stretch

C1 Span = 91.0 - 88.0 = 3.0 MHz = 3000 kHz (1). At a deviation of 75 kHz the bandwidth is 2(75 + 15) = 180 kHz (1), so the channel spacing is 180 + 20 = 200 kHz and the band holds 3000/200 = **15** stations (1). At a deviation of 25 kHz the bandwidth is 2(25 + 15) = 80 kHz (1), the spacing is 100 kHz and the band holds **30** stations (1), twice as many. What is lost is noise performance: FM's advantage over AM grows with the deviation, because a wider swing makes the wanted frequency changes large compared with the disturbance any interference can produce (1), so the narrowband stations would sound noisier and would carry less dynamic range even though the audio still reaches 15 kHz (1). Three standard errors haunt this question: leaving the span in MHz and dividing by a spacing in kHz, dropping the f_M term from the FM bandwidth, and forgetting the guard band, which alone changes the answer from 15 to 16. [7]

C2 With 8 bits: 8000 × 8 = 64 000 bits per second (1), the standard worldwide figure for one digital call. With 12 bits: 8000 × 12 = 96 000 bits per second (1). The number of levels rises from 2^8 = 256 to 2^{12} = 4096 (1), a factor of 16, so each step is 16 times smaller and the quantisation error falls by the same factor (1). Note the price: half as much again in bit rate for that improvement. Expecting the error to fall by a factor of 12/8 rather than by 2^4 is the standard error, because bits count doublings, not levels. [4]

C3 The rate is **not** adequate: it must exceed twice the highest frequency, that is more than 14 kHz, and 10 kHz falls short (1). Sampled too slowly, the 7.0 kHz tone masquerades as the difference between the sampling rate and its own frequency, 10 - 7.0 = 3.0 kHz (1) (1), and no processing afterwards can undo it, because the samples are now genuinely indistinguishable from those of a real 3.0 kHz tone. Minimum acceptable rate = 2 × 7.0 = 14 kHz, and in practice a little more (1). Believing a filter at the receiver can rescue the tone is the standard error; the only cure is a filter before sampling, and a fast enough sampler. [4]

C4 Width per channel = 2000 kHz/40 = 50 kHz (1). AM would need 2 × 4.5 = 9.0 kHz, so it fits with room to spare (1). FM would need 2(20 + 4.5) = 49 kHz, which fits, but only just, leaving 1 kHz of guard band (1). Recommend FM if the users need the noise immunity, since the amplitude clipping in an FM receiver removes most interference; recommend AM if more channels may be wanted later, since it uses a fifth of the spectrum per channel and the receivers are simpler and cheaper (1). Two standard errors: forgetting to convert 2.0 MHz to kilohertz before dividing, and dropping the f_M term so that FM appears to need only 40 kHz. [4]