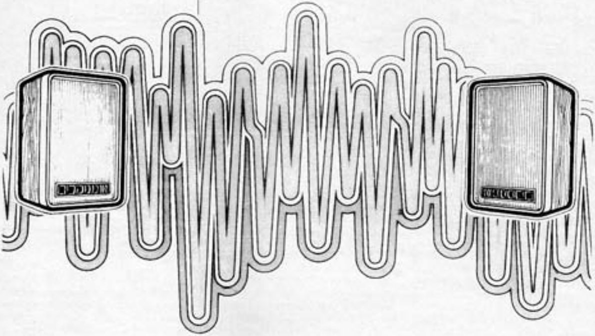
audio–analyser



Without an accurate picture of the frequency response of the sound reproduction system, the use of an equaliser can do more harm than good. For this reason an audio spectrum analyser, which can pinpoint the deficiencies in a particular audio chain and/or listening environment, is a virtually indispensable piece of equipment for the equaliser-user.

Attempting to set up a room acoustically by twiddling the controls on an equaliser and 'playing it by ear' is an almost certain recipe for heated tempers and high blood pressure, such is the difficulty of the task. To obtain any real benefit from an equaliser it is essential that the user knows exactly what changes he wants to implement in the frequency response of the audio system in question. It therefore follows that a reliable audio spectrum analyser is required to provide the acoustic information which is a necessary preliminary to effective equalisation.

An audio analyser system basically consists of three sections: a test-signal source (pink noise generator), a microphone to monitor the output of the audio system under test, and a suitable means of analysing and displaying the energy level of the incoming signal. Broadly speaking, audio analysers fall into one of two types, depending upon whether the analysis is real-time or not.

Real-time analyser

A *real-time analyser* is the most sophisticated, but also the most expensive way of obtaining a detailed picture of the spectrum of an audio signal. The operation of real-time analysers can be explained with reference to the block diagram of figure 1. A broadband test signal is fed to the audio system under test. Normally the test signal consists of pink noise, which has a uniform energy level over the entire spectrum. The output of the audio system is picked up by a measurement microphone and fed to a bank of octave or third-octave filters, which split the input signal into a corresponding number of adjacent frequency bands. The output voltage of each filter is then rectified and displayed. Various types of display are possible – a moving-coil meter, an oscilloscope, or, as in the commercially available spectrum analyser shown in figure 2, a matrix of LEDs. The advantage of a real-time analyser is that it enables the average energy level of the entire spectrum to be determined at a glance. However, in view of the large number of displays and filter sections which are required, real-time analysers are not cheap. The above-mentioned pocket analyser of figure 2, together with a suitable noise generator, costs in the region of £ 600 – and that is only a fraction of what some of its 'larger brothers' can cost!

Since however, the primary application of the analyser is to monitor the response of an audio system to a *constant* test signal (the output of the pink noise generator, which has a uniform spectral intensity) real-time analysis is something of a superfluous luxury. A much cheaper, but none the less satisfactory arrangement is to have a single tuneable filter, which can be swept up and down the frequency spectrum as desired. This is in fact the solution adopted in the Elektor audio analyser.

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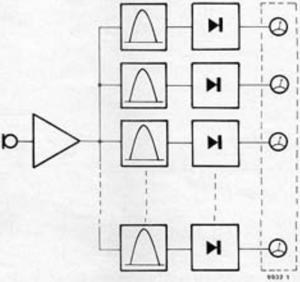

Figure 1. Block diagram of a real-time spectrum analyser.

The Elektor audio analyser

The block diagram of the Elektor, non real-time analyser is shown in figure 3. As can be seen, the basic principle of spectrum analysis remains the same, the only difference being that a single filter and display are employed, resulting in a considerable saving in cost. As far as the placing of the filter is concerned, three possible configurations come into consideration. In figure 3a the variable filter is situated between the pink noise generator and the input to the audio system, whilst in 3b it is fed from the output of the microphone. In figure 3c two filters are employed in an effort to obtain the best of both worlds. Although in theory there should be no difference between these three arrangements, things are not so simple in practice. With the configuration shown in figure 3a, all manner of interference and stray noise can reach the microphone and adversely effect the measurement. With the arrangement of figure 3b, this problem is effectively obviated, since only interference which lies within the passband of the filter can reach the microphone. A disadvantage of this set-up, however, is that only a very small portion of the pink noise spectrum is used, whilst the audio system in question is of course required to reproduce signals over the entire range of audio frequencies. The arrangement of figure 3c thus represents the ideal solution, however in view of the increased cost and complexity of two tracking variable filters, it was decided that, for this type of application, one of the simpler circuits (figures 3a and b) would prove sufficient.

2

Figure 2. Photograph of a commercially available hand-held real-time analyser, incorporating a LED-matrix display.

The basic requirements for an analyser of the above type are therefore:

- a pink-noise generator
- a bandpass filter with stepwise or continuously variable centre frequency
- a suitable microphone with preamplifier
- a rectifier circuit
- a display circuit

As far as the choice of microphone is concerned, it is clear that, unless it itself has a fairly flat response, one cannot hope to obtain an accurate picture of the response of the audio system/listening room under test. For this reason it is important to invest in a reasonably good quality microphone capsule and preamp.

As a display circuit, a multimeter is as good as any, and has the advantage of being cheap and commonly available.

The remaining circuits, which form the heart of the analyser – and the substance of the rest of this article – are shown in figures 4a, 4b and 4c.

Noise generator

As can be seen from the circuit diagram of the noise generator shown in figure 4a, it in fact consists of a pseudo-random binary sequence generator, which has a longer than normal cycle time. This ensures that the noise has a high spectral density and that it is not characterised by the annoying 'breathing' effect obtained with short cycle times. The length of the shift register (IC1 . . . IC4) is 31 bits, and since the frequency of the clock generator (N5 . . . N7, C1, C2, R3, R4) is roughly 500 kHz, the full cycle time is approximately an hour and a quarter!

EXOR-feedback is provided by N1 . . . N4. The circuit however has no anti-latch up gating. Instead there are two pushbutton switches; the START button ensures a logic 1 at the data input Q_0 of the shift register (pin 7 of IC1), thereby starting the clock cycle.

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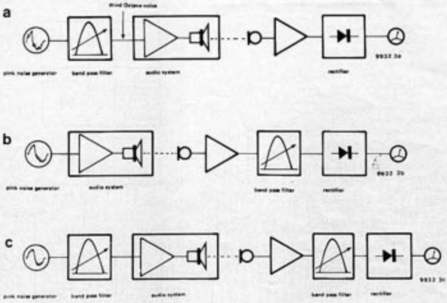


Figure 3. Three possible designs for a non real-time analyser.

4a

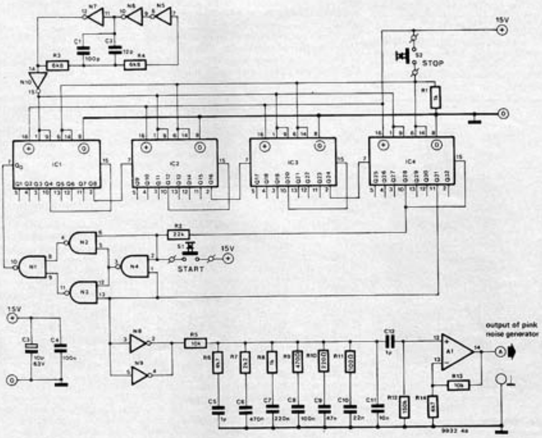
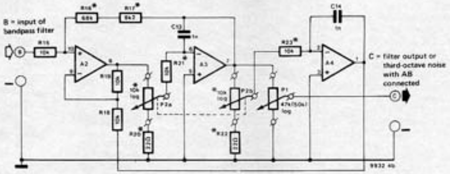


Figure 4a. The pink noise generator.

4b



* see table

Figure 4b. The bandpass filter.

The cycle is inhibited by pressing the STOP button, S2. In this way it is possible to (temporarily) disconnect the noise source without switching off the supply voltage – a useful if not downright indispensable feature. The (pseudo-) white noise output of the shift register is fed to the pink-noise filter formed by R5 . . . R11, C5 . . . C11, before being amplified in the circuit round A1.

Bandpass filter

This section of the circuit (shown in figure 4b) is virtually identical to the third octave filter described in the article on the CMOS noise generator. The output level of the filter can be varied by means of potentiometer P1, whilst the centre frequency can be varied between approximately 40 Hz and 16 kHz by means of the stereo potentiometer P2a/P2b. If stepwise control of the centre frequency of the filter is desired, P2a/P2b can be replaced by a pair of attenuator networks and a twin-ganged switch. The necessary modifications are detailed in figure 5. Resistors R20 and R22 are replaced by a wire link, the values of R21 and R23 are altered, and R40 and R41 are added. Table 1 lists the various resistance values required to give the ISO standard centre frequencies. When calibrating a parametric equaliser, a filter bandwidth of less than 1/3 of an octave is required. By altering the value of R16 to 220 Ω and replacing R17 by a wire link a bandwidth of approximately 1/12 of an octave can be obtained.

Rectifier circuit

It is of utmost importance that the amplitude of the test signal be measured accurately. If a pink noise test signal is used in conjunction with filters which

4c

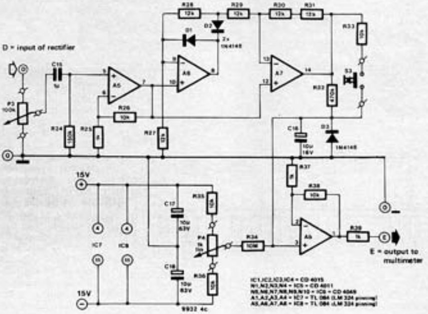


Figure 4c. The rectifier circuit.

have a constant octave or 1/3 octave bandwidth (i.e. filters with a constant Q) one should really measure the RMS value of the noise – not an easy matter. Fortunately, however, a reasonably simple alternative exists – namely to measure the average of the modulus value, i.e. the average of the full-wave rectified noise signal. This is obtained by feeding the output of the peak rectifier to a lowpass filter.
The rectifier circuit is built round IC8. The input level control is followed by an amplifier, A5. The actual (full-wave) rectification is performed by A6, A7, R27 . . . 31, D1 and D2. The output of A7, which always presents a low impedance, is connected via R32 to C16. Because this capacitor has the same charge and discharge time, the voltage on the capacitor will equal the average value of the full-wave rectified noise voltage. The time that this voltage remains stored on the capacitor is determined by the RC time constant, R32·C16, or, if S3 is depressed, R32/R33·C16. Depressing S3 causes C16 to charge and discharge much more rapidly, so that the capacitor voltage will follow rapid variations in the noise voltage. Thus S3 is intended to provide a rapid overall view of the variations in noise level for different centre frequencies of the filter. For accurate measurements, the longer time constant of R32·C16 should be used. After being amplified in A8, the voltage on C16 is displayed on the multimeter. An offset control is provided (P4, R34 . . . R36) to enable the meter to be calibrated accurately (zero deflection under quiescent conditions).

Construction

A printed circuit board, which is shown in figure 6, has been designed to accommodate the circuit of figures 4a, b and c.

Table

1	2	3	4	5
31,5	1/1	2Ω2 + 2Ω2	18 k	w
31,5	1/3	2Ω2 + 2Ω2	68 k	8k2
40	1/3	5Ω6	68 k	8k2
50	1/3	4Ω7 + 2Ω2	68 k	8k2
63	1/1	4Ω7 + 3Ω9	18 k	w
63	1/3	4Ω7 + 3Ω9	68 k	8k2
80	1/3	10 Ω + 1Ω2	68 k	8k2
100	1/3	10 Ω + 3Ω9	68 k	8k2
125	1/1	12 Ω + 5Ω6	18 k	w
125	1/3	12 Ω + 5Ω6	68 k	8k2
160	1/3	22 Ω	68 k	8k2
200	1/3	27 Ω + 1Ω8	68 k	8k2
250	1/1	33 Ω + 2Ω2	18 k	w
250	1/3	33 Ω + 2Ω2	68 k	8k2
315	1/3	22 Ω + 22 Ω	68 k	8k2
400	1/3	56 Ω	68 k	8k2
500	1/1	68 Ω + 3Ω3	18 k	w
500	1/3	68 Ω + 3Ω3	68 k	8k2
630	1/3	82 Ω + 8Ω2	68 k	8k2
800	1/3	100 Ω + 18 Ω	68 k	8k2
1000	1/1	100 Ω + 47 Ω	18 k	w
1000	1/3	100 Ω + 47 Ω	68 k	8k2
1250	1/3	120 Ω + 68 Ω	68 k	8k2
1600	1/3	220 Ω + 27 Ω	68 k	8k2
2000	1/1	270 Ω + 47 Ω	18 k	w
2000	1/3	270 Ω + 47 Ω	68 k	8k2
2500	1/3	390 Ω + 18 Ω	68 k	8k2
3150	1/3	470 Ω + 68 Ω	68 k	8k2
4000	1/1	680 Ω + 47 Ω	18 k	w
4000	1/3	680 Ω + 47 Ω	68 k	8k2
5000	1/3	820 Ω + 150 Ω	68 k	8k2
6300	1/3	1 k + 390 Ω	68 k	8k2
8000	1/1	1k8 + 330 Ω	18 k	w
8000	1/3	1k8 + 330 Ω	68 k	8k2
10.000	1/3	3k3 + 390 Ω	68 k	8k2
12.500	1/3	5k6 + 1 k	68 k	8k2
16.000	1/1	39 k + 1k2	18 k	w
16.000	1/3	39 k + 1k2	68 k	8k2

Remarks:
column 1: centre frequency in Hz
column 2: bandwidth in octaves
column 3: value of resistor to be connected between the junction of resistors R40 and R21 and ground and between the junction of R41 and R23 and ground, rounded up to values from the E12 series.
column 4: value of R16
column 5: value of R17 (w = wire link)

5

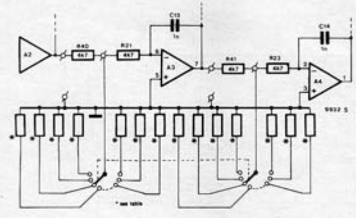


Figure 5. Modifications to the bandpass filter to obtain switched centre frequencies.

The design of the board is such that either of the configurations shown in figures 3a and 3b can be adopted. The construction of the standard version circuit should present no special problems. The wiring for the potentiometers and switches should be kept as short as possible. The connections for these components are arranged at one end of the board. Problems of a practical nature do arise, however, if one desires a number of switched filter frequencies, since one then requires a switch with a corresponding number of ways. Since switches with a large number of ways are both expensive and difficult to obtain, an alternative solution is simply to use the desired number of double-pole single-throw switches. This of course involves operating two switches each time one wants to alter the centre frequency of the filter.
In addition to the switch(es), the choice of fixed filter frequencies involves the following alterations on the board (see

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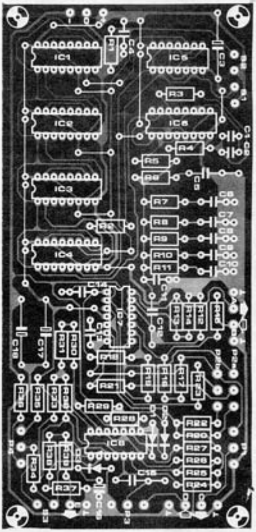

Figure 6. Printed circuit board for the circuit of figure 4.

parts list

Resistors:

R1,R8,R25,R37,R39 = 1 k
R2 = 22 k
R3,R4 = 6k8
R5,R13,R15,R18,R19,R21,R23,
  R26,R33,R35,R36,R38 = 10 k
R6,R14 = 4k7
R7 = 2k2
R9 = 470 Ω
R10 = 220 Ω
R11 = 100 Ω
R12,R24 = 150 k
R16 = 68 k
R17 = 8k2
R20,R22 = 22 Ω
R27 . . . R31 = 12 k
R32 = 470 k
R34 = 10 M
P1 = 47 k (50 k) log potentiometer
P2a/P2b = 10 k log stereo potentiometer
P3 = 100 k log potentiometer
P4 = 1 k linear potentiometer

capacitors:

C1 = 100 p
C2 = 12 p
C3,C17,C18 = 10 μ/63 V
C4,C8 = 100 n
C5,C12,C15 = 1 μ MKM
C6 = 470 n
C7 = 220 n
C9 = 47 n
C10 = 22 n
C11 = 10 n
C13,C14 = 1 n
C16 = 1 Uμ/35 V tantalum

Semiconductors:

IC1,IC2,IC3,IC4 = CD 4015
IC5 = CD 4011
IC6 = CD 4049
IC7,IC8 = TL 084 (Texas Instruments) DIL
D1,D2,D3 = 1N4148

Miscellaneous:

S1,S2,S3 = pushbutton switch, single-pole push-to-make

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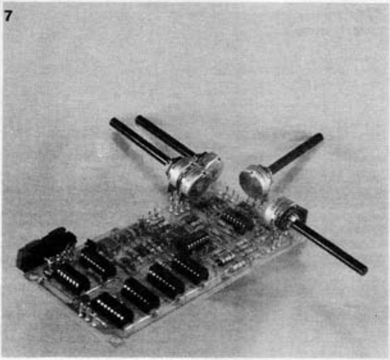

Figure 7. A prototype of the audio analyser.

figure 5): R21 and R23 become 4k7
R20 and R22 are replaced by a wire link
a 4k7 resistor (R40) is soldered between the 'top' two tags of P2a
a 4k7 resistor (R41) is soldered between the 'bottom' two tags of P2b

The resistor pairs forming the switched attenuator network are mounted externally on the switch(es). Suitable values are given in the table.

With the continuously variable filter frequency it is useful to equip P2a/b with a pointer and scale. The scale can of course be calibrated in frequencies, but it is not strictly necessary. What matters is that one has a series of reference points – peak or dip at such and such a filter setting, etc. If, however an absolute frequency scale is desired, this can be obtained by using a tone generator and noting the frequancy when the output voltage at point C is at a maximum, when feeding a pure sinewave into point B.

Using the analyser

The multimeter (10 to 12 V full-scale deflection) which is used to display the amplitude of the noise signal is connected to the output (point E) of the rectifier circuit. In the absence of an AC drive voltage (i.e. point D disconnected or else P3 turned right down) the DC voltage at this point should be set by means of P4 to *exactly* 0 (m)V. The correct setting for P4 is obtained by repeatedly switching down the voltage range of the multimeter and checking the reading by reversing the polarity of the probes. It should be borne in mind that, because of the long time constant of R34 and C16, it will take some time for adjustments to P4 to have any effect. The long discharge time of the storage capacitor in the rectifier circuit together with the natural inertia of the meter ballistics ensure that the needle responds only very slowly to changes in the level of the filter output. Thus when sweeping the filter up and down the audio spectrum, care should be taken to vary the filter frequency *gradually*, lest peaks or dips in the response are camouflaged by the slow response of the circuit.

If the analyser is used to measure a system with a completely flat response, the *mean* meter deflection (i.e. the mean between the maximum positive and negative deflections) should be independent of variations in the filter frequency. An audio system with a completely flat response would be pretty hard to find, however, something which does have a more or less flat response is a wire link! – by joining points A and B and C and D in this way (i.e. connecting the output of the noise generator to the bandpass filter and the output of the filter to the rectifier circuit) it is possible to test the operation of the audio analyser, and in particular, of the pink noise and bandpass filters. Variations of up to ± 2 dB (0.8 . . . 1.25) in the mean meter reading are acceptable. To prevent the rectifier circuit from being overloaded, the mean meter reading can be adjusted to occur at around 3 . . . 4 V.

Finally a word of warning: care should be taken to ensure that the noise signal does not overload one's audio equipment. The risk of this happening is somewhat greater than in the case of a sine or squarewave input signal, since the distortion caused by overloading will be that much less noticeable (but none the less disastrous!). Tweeters in particular are susceptible to damage by being overloaded with high level noise signals.

Constructing the audio analyser is one thing, using it is another. The reader is therefore referred to the article on 'Using an equaliser', which deals with the subject of using the equaliser/analyser combination to measure and then correct a room's response.