No. 1 1972

Technical Review

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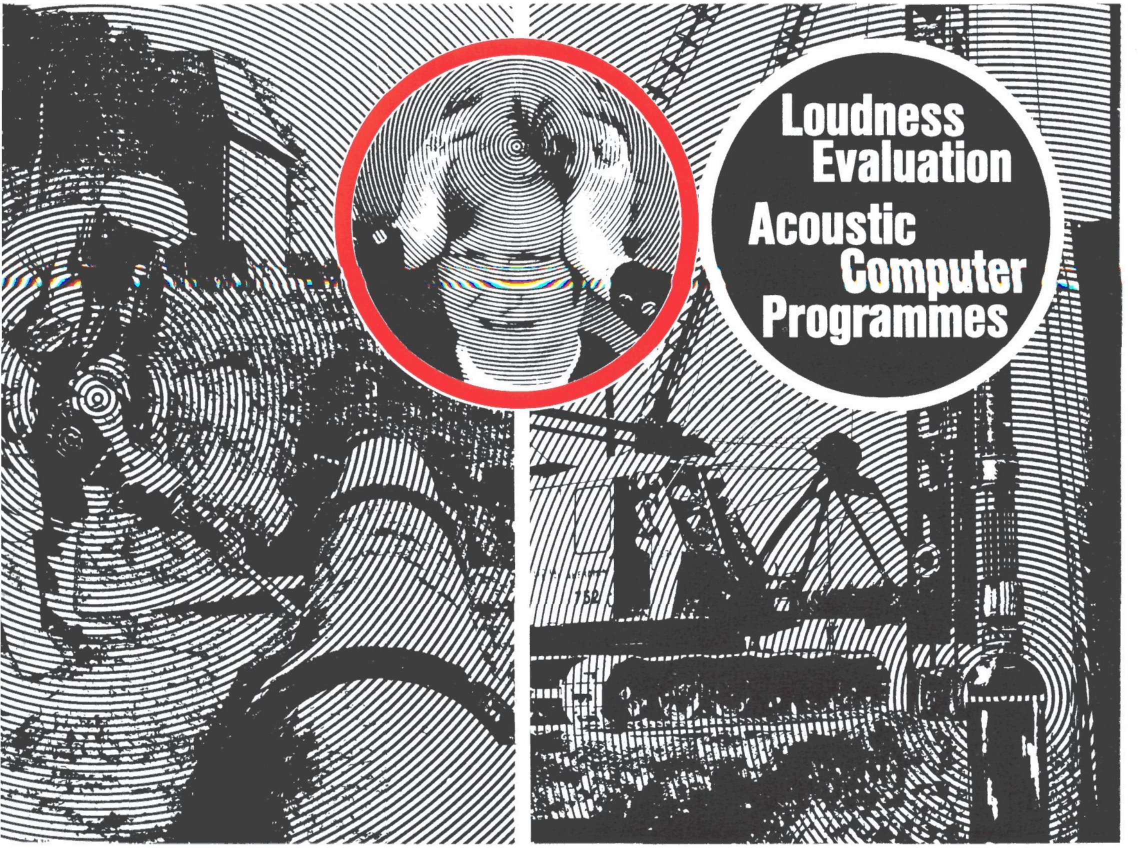

BRÜEL & KJÆR

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TECHNICAL REVIEW

No. 1 — 1972

Contents

Loudness Evaluation of Acoustic Impulses

by

P. Hedegaard

ABSTRACT

A method for the measurement and calculation of loudness is described and discussed. The method is based on measurements with the Impulse Precision Sound Level Meter, Type 2204 in connection with a 1/3 octave filter set. The procedure is compared with a procedure for evaluation of the loudness of sonic bangs by D.R. Johnson and D.W. Robinson of the National Physical Laboratory in Teddington, England.

SOMMAIRE

Cet article a pour objet l'étude et la discussion d'une méthode de mesure et de calcul de sonie. Celle ci est basée sur des résultats obtenus avec un Sonomètre de Précision pour impulsions 2204 associé à un jeu de filtres 1/3 d'octave.

Cette technique est comparée avec celle utilisée pour les mesures de sonie des bangs soniques et étudiée par D.R. Johnson et D.W. Robinson du Laboratoire Nationale de Physique de Teddington (Grande Bretagne).

ZUSAMMENFASSUNG

Eine Methode zur Messung und Berechnung der Lautheit von Schallimpulsen wird beschrieben und diskutiert. Sie basiert auf Messungen mit dem "Impulsschallpegelmesser Typ 2204" in Verbindung mit Terzfiltern. Als Vergleich dient eine von D.R. Johnson und D.W. Robinson (National Physical Laboratory in Teddington, England) angegebene Prozedur zur Ermittlung der Lautheit von Überschallknallen.

Introduction

Calculation of the loudness of impulsive sounds may be carried out using a method described by D.R. Johnson and D.W. Robinson. The method is based on a known Fourier density spectrum which can be calculated from the impulse pressure-time history or estimated by sequential filtering of a tape recording. The Fourier spectrum is transformed into a 1/3 octave "energy" spectrum (sound pressure squared x time) which is then converted into an equivalent 1/3 octave band pressure level system. The conversion is made assuming an integration time of the ear of 70 msecs. This method is rather complicated and can involve extensive, and therefore expensive, equipment. Therefore it can be an advantage to make the measurement in 1/3 octaves and use the standardized*)RMS Rectifier in an Impulse Precision Sound Level Meter for the integration process. A comparison is made of the results obtained by the two methods using several different shaped pulses.

*) DIN 45633 part 2.
Draft for supplement to IEC publication 179, Precision Sound Level Meter.

Choice of Impulse Shapes

Five different shapes of impulses were used for the investigation. The impulses were chosen for their different spectra in order to make a comprehensive examination of the calculation methods. The impulses came directly from impulse generators and were fed into the input of the measuring system. The impulse shapes are shown in Fig. 1.

Note Equivalent sound pressure levels are given in dB re 2×10^{-5} N/m^2 (2×10^{-4} μbar).

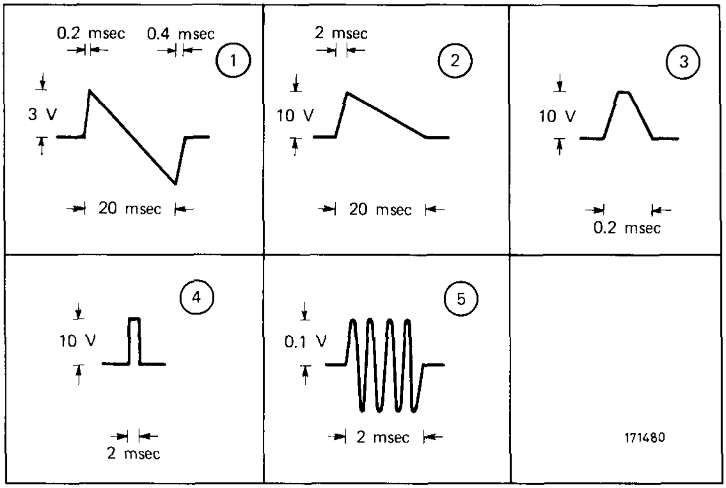

Fig. 1. Impulse shapes used for the investigation

Measurement of Density Spectra

The spectra were estimated by sequential filtering with a narrow band (constant bandwidth) filter. This method is described by J.T. Broch and H.P. Olesen, see references. The measuring set-up is shown in Fig.2. The frequency range was scanned and the peak value of the filtered signal recorded on a level recorder.

The recordings are calibrated by means of the formulae

$$F(f_o) = \frac{F_{\Delta f}(t)\ \text{max}}{2\Delta f}$$

where: $F_{\Delta f}(t)$ max is the peak value measured,
Δf is the filter bandwidth,
and $F(f_o)$ is the amplitude density at the frequency f_o.

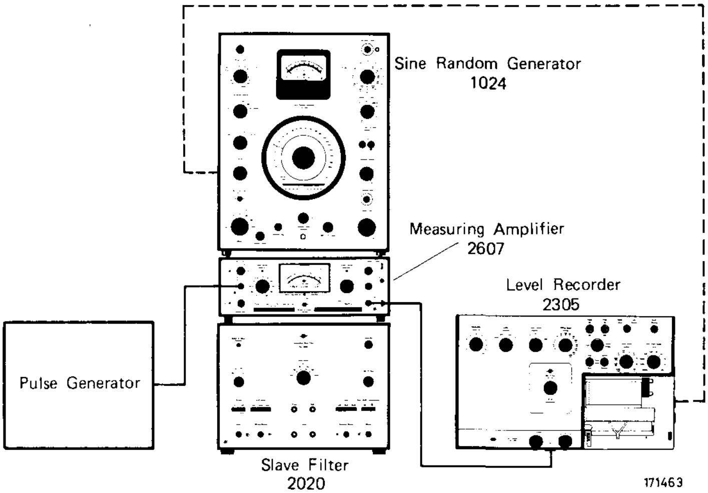

Fig.2. Set up for measuring Fourier density spectra

The density spectra measured for the different impulse shapes are shown in Fig.3.

Calculation of 1/3 octave "energy" spectra and equivalent band pressure level

The "energy" spectrum is calculated from the amplitude density spectrum by means of the formula,

$$E_{1/3} = 2 \int_{f_o - \Delta f/2}^{f_o + \Delta f/2} F^2(f)\, df$$

where: $F(f)$ is the amplitude density spectrum value at the frequency f,
f_o is the centre frequency of the 1/3 octave band,
and Δf is the 1/3 octave absolute bandwidth in Hz.

The equivalent band pressure level is calculated from the "energy" spectrum by means of the formula

$$P_{1/3} = \sqrt{\frac{E_{1/3}}{0.07}}$$

This assumes an integration time of 0.07 sec. (70 msec.).

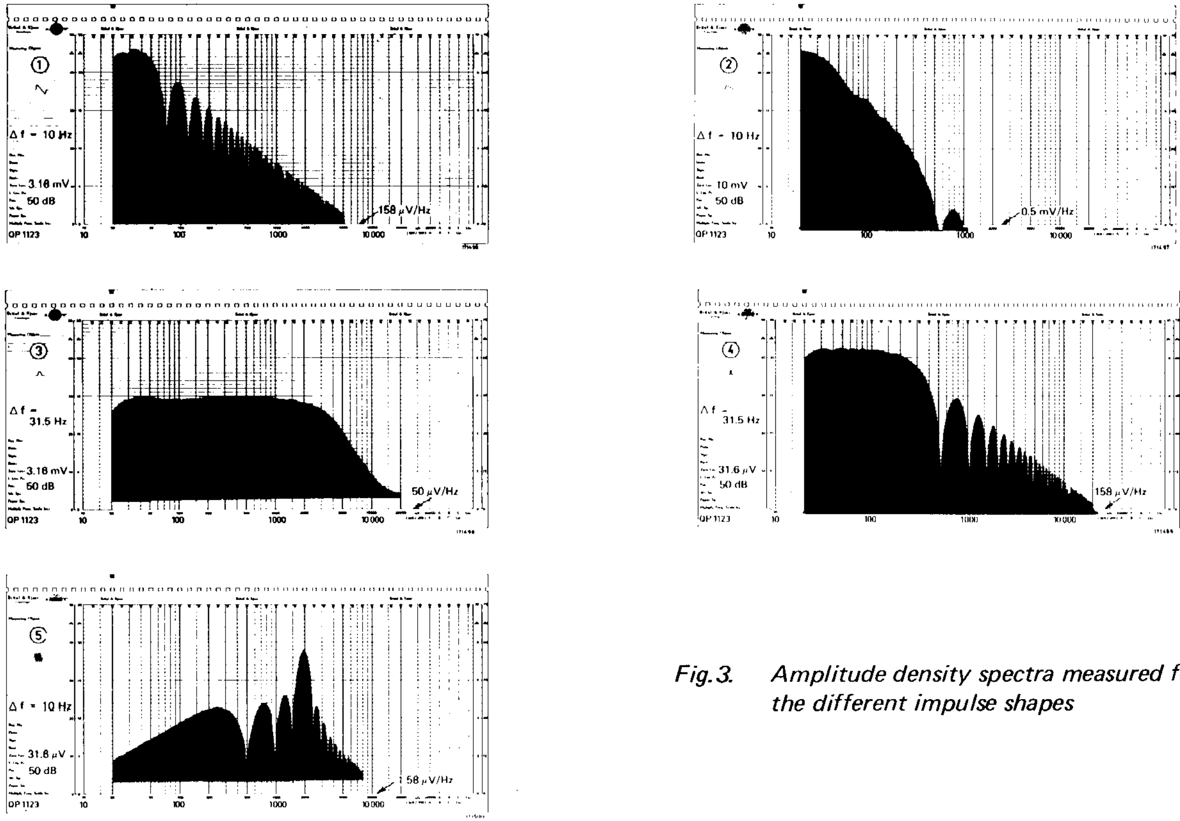

Fig. 3. Amplitude density spectra measured for the different impulse shapes

Direct measurement of Band Pressure Levels

In the previous section, a linear integration is assumed, and the calculation is based on an integration time of 70 msec. However, an integration time constant of 35 msec. is recommended, for impulse SL-meters (IEC proposal). The principle of the rectifier system is shown in Fig.4.

Direct measurements of the band pressure levels were made using the set up shown in Fig.5. A "Hold" circuit on the impulse sound level meter works in a similar way to that shown in Fig.4, however, the decay time constant of the storage device is made very large, which enables the meter deflection to be held, and therefore, easy to read. The impulse was repeated for each filter used.

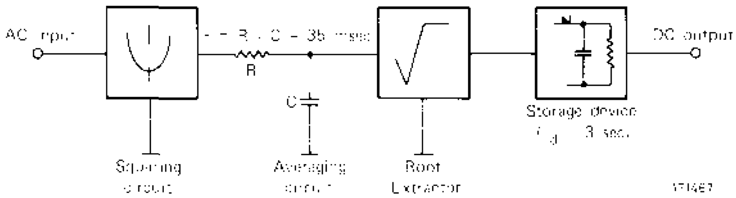

Fig.4. *Principle of the R.M.S. circuit standardized for impulse sound level meter*

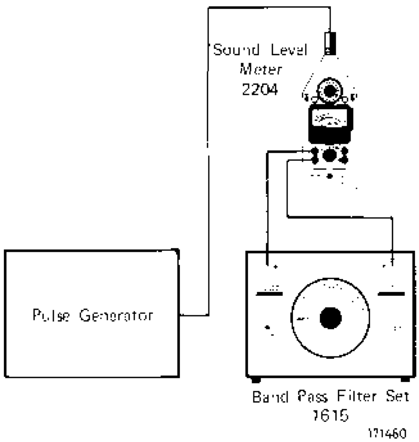

Fig.5. *Set up for measuring impulse band pressure levels*

Measuring Results and Comparisons

The 1/3 octave band pressure level spectra for the various impulse shapes used are shown in Figs.6 to 15. The spectra are obtained in two ways for all the impulse shapes.

i) measurement directly from the technique previously described.
ii) calculation from the respective density spectra.

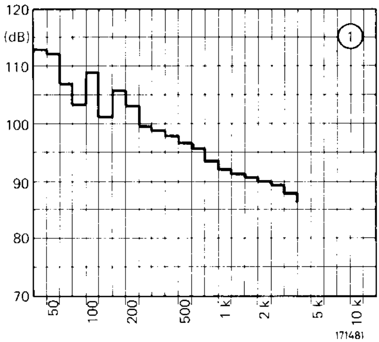

Fig.6. 1/3-octave spectrum for sonic bang (calculated from density spectrum)

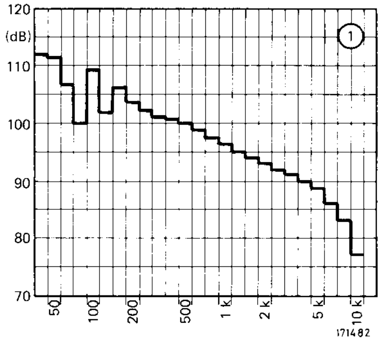

Fig.7. 1/3-octave spectrum for sonic bang (direct measurement)

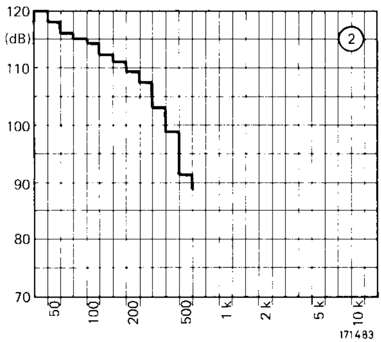

Fig.8. 1/3-octave spectrum for triangular impulse (calculated from density spectrum)

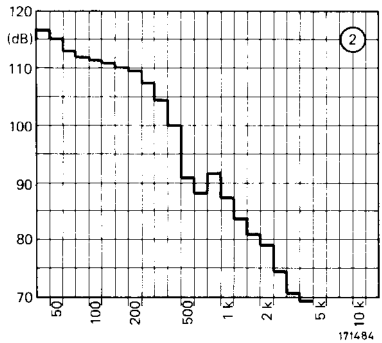

Fig.9. 1/3-octave spectrum for triangular impulse (direct measurement)

The dB values are referred to 1 μV and are equivalent to Sound Pressure Level (SPL) assuming a microphone with a sensitivity of 50 mV per N/m^2 (5 mV/μbar) is used. This corresponds to the 1 inch B & K microphone, Type 4145.

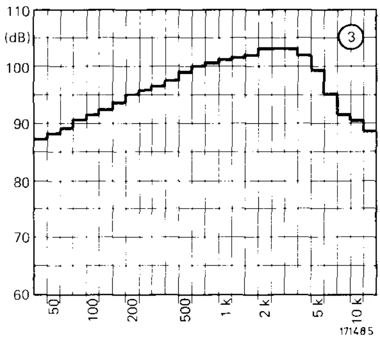

Fig. 10. 1/3-octave spectrum for "gun-shot" (calculated from density spectrum)

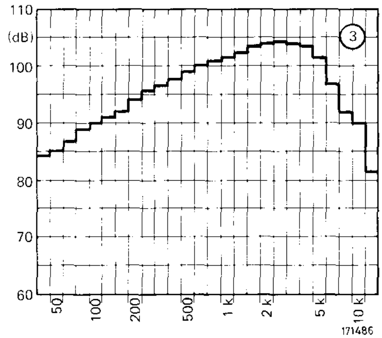

Fig. 11. 1/3-octave spectrum for "gun-shot" (direct measurement)

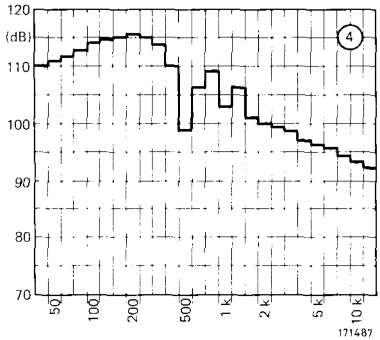

Fig. 12. 1/3-octave spectrum for square-wave impulse (calculated from density spectrum)

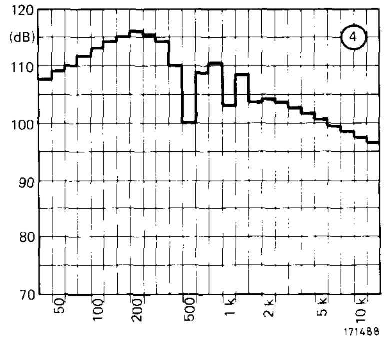

Fig. 13. 1/3-octave spectrum for square-wave impulse (direct measurement)

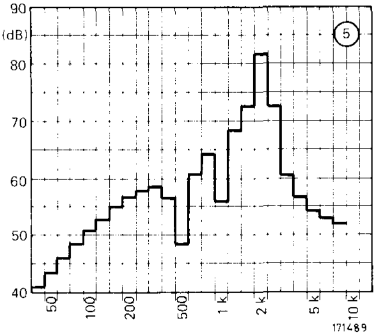

Fig.14. 1/3-octave spectrum for tone-burst (calculated from density spectrum)

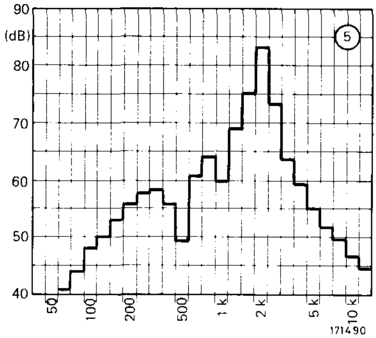

Fig.15. 1/3-octave spectrum for tone-burst (measured directly)

The band pressure levels are converted to loudness indices by means of the chart shown in Fig.16*), and the total loudness of the impulses calculated by the formula:

$$S_t = S_m + 0.15\,(\Sigma S - S_m)$$

where S_m is the largest of the band loudness indices
ΣS is the sum of the loudness indices of all bands
and S_t is the total loudness in sones.

The total loudness value is converted to a total loudness level value in phons by applying the nomograph on the right of Fig.16.

The table, Fig.17, shows the loudness level values for the different impulses used. The loudness level values are calculated from both the measuring methods described. A comparison was also made between measurements using the standard weighting curves ("D", "A", "B", "C").

Discussion

It can be seen from the 1/3 octave spectra, see Figs.6 to 15, that there is, in some cases, a certain lack of information in the high frequency region. This

*) The chart shown in Fig.16 has been recommended by the ISO in their recommendation R 532 "Procedure for Calculating loudness level". It originates from research carried out by the American scientist S.S. Stevens, who denoted it "Mark IV". A modification of this chart, "Mark VII" has, however, been proposed by Prof. Stevens.

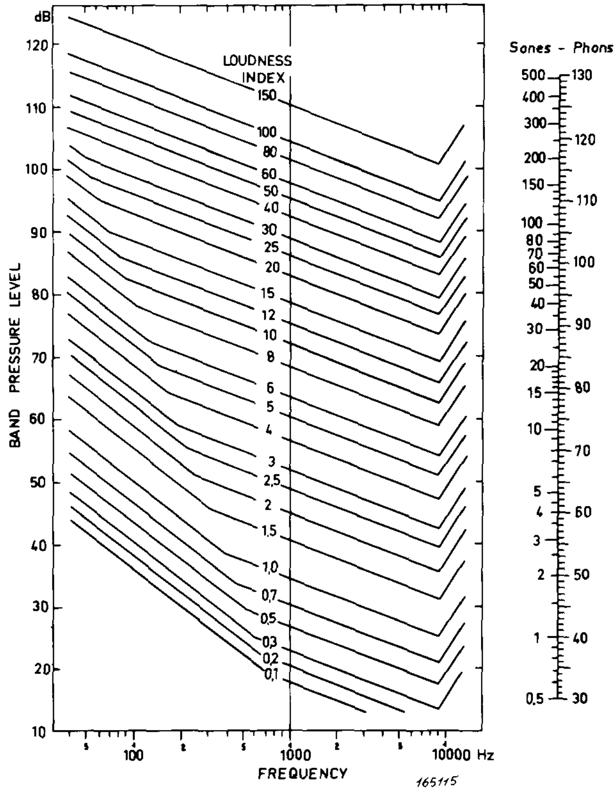

Fig. 16. Contours of equal loudness index

		①	②	③	④	⑤
LOUDNESS LEVEL (calculated from density spectrum)	Phon	116.5	120	125	129.5	92.5
LOUDNESS LEVEL (calculated from the directly measured 1/3 octave impulse spectrum)	Phon	118	117	126.5	132	93
"D"	dB (I)	113.5	116.5	121.5	124	92
"A"	dB (I)	106.5	109	114	119	85.5
"B"	dB (I)	113	118	113.5	123	84.5
"C"	dB (I)	120	125.5	113.5	125.5	84.5
"Lin"	dB (I)	120.5	125.5	114	126	84.5

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Fig. 17. Comparison of calculated and measured results

can be seen especially for the calculated spectra. The missing information is due to the limited signal-to-noise ratio for the density spectra measurements. Therefore, for a fair comparison, the loudness calculation is only carried out for the part of the spectra which occurs in both the calculated and measured spectra. The limitation of signal-to-noise ratio on account of the inherent noise can be seen from the difference in curves of Figs.10 and 11 and Figs.14 and 15 at high frequencies. In spite of these limitations, the deviations between the results of these two methods are small, (see Table Fig.17).

Fig.18 shows the deviation between the calculated and measured band pressure levels for the 1/3 octave spectra shown in Figs.6 to 15. It can be seen that the deviation is consistent from the "envelope" of the deviation points. (See Fig.19). The deviation, approximately + 3 dB, at the high frequency end, may be explained by the shorter time constant used, (i.e. 35 msec. instead of 70 msec.). The deviation at the low frequency end may be explained by the fact that the relatively short time constant will give rise to an integration of a part of the signal, and not the complete signal. This can be compared to the calculation principle where the whole signal is integrated in the density spectrum and the converted (band pressure squared x time) spectrum is corrected for a 70 msec time constant. Assuming the Johnson-Robinson method to be correct, corrections for the directly measured spectra taken from the dotted curve in Fig.19 will give rise to even smaller deviations for the calculated phon values.

Conclusion

The loudness measurement method based on directly measured impulse band pressure levels give results which agree fairly well with the Johnson-Robinson method. The Johnson-Robinson method is especially suitable for digital processing where large amounts of data can be handled. Otherwise the method seems to be rather time consuming, and if the detailed information from the Fourier analyses is not used even slightly better results do not seem to justify the more expensive measuring system. Furthermore the density measurements demand a large dynamic range which is not always easy to obtain in narrow band analyses of pulses over the whole audible frequency range and the conversion from the density spectrum to the equivalent 1/3-octave band pressure spectrum is in practice not very accurate.

The method based on band pressure levels which are directly measured is easier to handle and permits larger dynamic range to be obtained on account of the 1/3 octave bandwidths. On the other band, however, inaccuracies will be introduced at low frequencies on account of the slow response of the 1/3

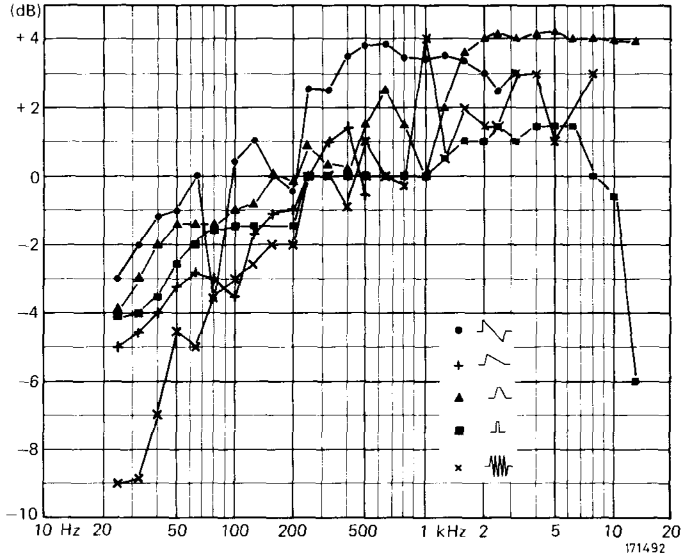

Fig. 18. Deviations between calculated and directly measured 1/3 octave band pressure levels

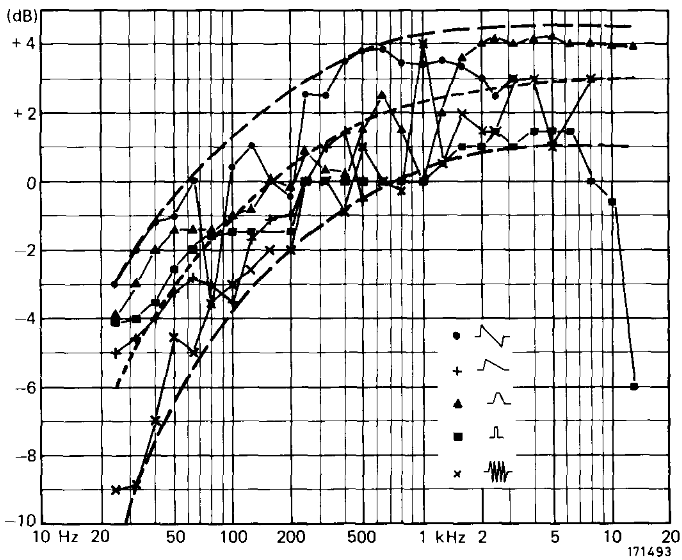

Fig. 19. "Envelope" and estimated correction curve. (For correction of direct measurements, shift sign on dB scale)

octave filters compared with the integration time constant used. There is, of course, great practical value in having a single rectifying system and single time constant, which can be used for both wide band, and 1/3 octave band measurements. If this is acceptable, it may introduce a measuring technique where the same principles, loudness contours and calculation procedure can be used for both impulsive and steady state signal processing.

Furthermore, it is not necessary for the operator to apply corrections in certain circumstances e.g. in a double sonic bang, as the system will measure the highest level if the bangs have a time interval greater than the integration time constant used, and integrate if the time interval is less.

Appendix I

FEATURES OF THE RECTIFIER SYSTEM IN THE "IMPULSE PRECISION SOUND LEVEL METER TYPE 2204"

Crest Factor

Impulse phenomena filtered in 1/3 octave bands and detected in RMS values will only give a correct picture of the energy if the integration time is long compared with the length of the impulse filter response. At the high frequency end of the spectrum the peak value of the signal measured at the filter output may be very large compared to the RMS or averaged value. This means that the RMS detector must be capable of dealing with large crest factors if large errors in this region cannot be tolerated. The RMS detector in the "impulse Sound Level Meter" Type 2204 is capable of dealing with crest factors up to 40, which is sufficient for a vast majority of applications. Overload lamps inform the user of measuring conditions which are outside the specified range. Fig.20 shows the 1/3 octave spectrum of a step function obtained using the system described.

Impulse Hold

The standardized rectifier system shown in Fig.4 contains a storage device with a decay time constant of 3 secs. This time constant has no effect on the measuring results when measuring single impulse phenomena. The "Hold" mode of the system is identical to the standardized "impulse" mode except for a very long time constant of the storage device. This facilitates measurements on single impulses (filtered or unfiltered), as the meter will record the highest value until it is reset by a push-button. For analysis of measurements stored on tape, the tape recorder is simply rewound and the "Hold" position reset for each (filtered) measurement.

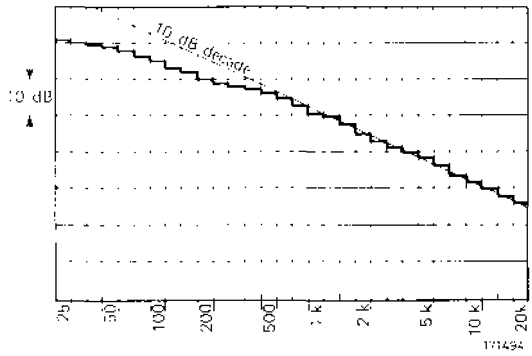

Fig.20. 1/3 octave spectrum of step function. (Integration time constant = 35 msec.)

Appendix II

Alternative measuring systems

The measuring set-up shown in Fig.5 can be substituted by other B & K set-ups. Figs.21 and 22 show examples.

The rectifier in the "Measuring Amplifier" Type 2606, and in the "Frequency Spectrometer" Type 2113, is identical to the rectifier used in the "Impulse Precision Sound Level Meter" Type 2204.

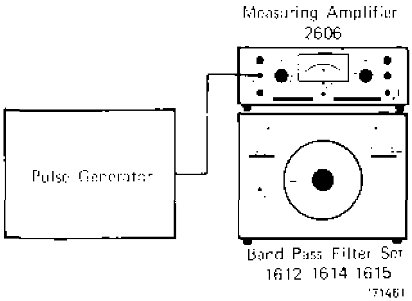

Fig.21. Alternative for set-up shown on Fig.5

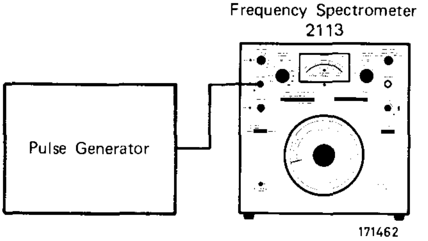

Fig.22. Alternative for set-up shown on Fig.5

References

D.R. JOHNSON and D.W. ROBINSON:	The subjective evaluation of sonic bangs. Acoustica Vol. 18 no. 5 (1967).
D.R. JOHNSON and D.W. ROBINSON:	Procedure for calculating the loudness of sonic bangs. Acoustica Vol. 21 no. 6 (1965).
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C.G. WAHRMANN:	Impulse Noise Measurements B & K Technical Review 1969 no. 1.

Computer Programming Requirements for Acoustic Measurements*

by

H. Melchior

ABSTRACT
The paper contains a survey of problems encountered when programming small computers for instrumentation purposes. Examples are given of special acoustical routines in Assembler language. Advantages and limitations of using higher level programming languages, as for example BASIC, are also discussed.

SOMMAIRE
Cet exposé traite des problèmes rencontrés dans la programmation des petits calculateurs. Nous donnons des exemples de programmes classiques dans le domaine de l'acoustique rédigés en "Assembleur". Les avantages et les limites dus à l'emploi de langages plus élaborés, tel le BASIC, sont en outre exposés.

ZUSAMMENFASSUNG
Die Ausführungen des Verfassers geben einen Überblick über die Probleme die bei der Programmierung von Klein-Computern bei Meßaufgaben auftreten. Beispiele für spezielle akustische Programme in Assembler-Sprache werden erläutert, und die Vorteile und Begrenzung in der Anwendung höherer Programmierungssprachen, wie z.B. BASIC, werden diskutiert.

Introduction

Programming of small computers for instrumentation purposes is not essentially different from programming large computers, but the limited size of the core memory forces the user to minimize the size of his programs. When using a large computer in a data processing centre, the user will have a large core memory at his disposal, and he will have access to extensive peripheral storage devices, so that he will feel that the computer is virtually unlimited in size.

A mini-computer usually has a core memory of 4 k or 8 k computer words. (1 k = 1024 words). One computer word consists of 16 bits in most mini-computers. Program as well as data are stored in the core memory, and the central processing unit performs the instructions of the program and controls the sequence in which the instructions are executed.

*) Paper presented at the 7th International Congress on Acoustics, Budapest, Hungary, 18–26. August 1971.

Each single instruction must somehow be placed in the computer memory. For this purpose, higher-level programming languages such as FORTRAN, ALGOL and BASIC exist. FORTRAN and ALGOL are very general languages, made for handling mathematical problems on large computers. Most mini-computer manufacturers supply a FORTRAN compiler with the computer, which can process a somewhat restricted version of the usual FORTRAN IV. Because of its very general nature, the subprograms etc. for a given FORTRAN program will take up a lot of the space available in the minicomputer, leaving very little room for the FORTRAN program itself. For this reason, only high-level languages specially suited for mini-computers e.g. BASIC, are of practical value to the user.

BASIC programs

A BASIC program for computing and printing sin (x) from 0^{o} to 90^{o} in steps of 10^{o} will be as follows:

```
10   FOR N=0 TO 9
20   LET X=SIN(N*10/180*3.1416)
30   PRINT 10*N,X
40   NEXT N
50   END
```

Calculations in BASIC can be done by the simple set-up shown in Fig.1, provided the computer has an 8 k memory. The tape reader shown is convenient for fast read-in of programs and data, but is not necessary for the operation. The user types his program line by line on the teletype keyboard. If there is an error of syntax in a statement, the computer will print an error message on the teletype, and the user must retype the statement. This is called conversational programming. When the program is finished, the user types RUN, and the calculation is performed including the typing of the results on the teletype.

For acoustic instrumentation purposes, the BASIC compiler can be modified, so that a program can read in data from external measuring equipment. As an example, Fig.2 shows the Brüel & Kjær Real Time Third Octave Analyzer connected to a computer through an electronic interface. The special order

```
CALL $RTA,N,A(1)
```

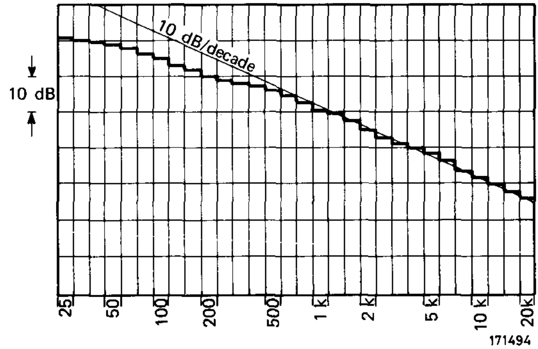

Fig.20. 1/3 octave spectrum of step function. (Integration time constant = 35 msec.)

Appendix II

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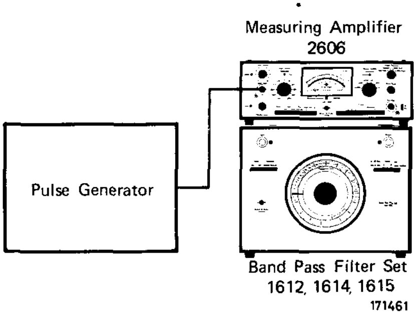

Fig.21. Alternative for set-up shown on Fig.5

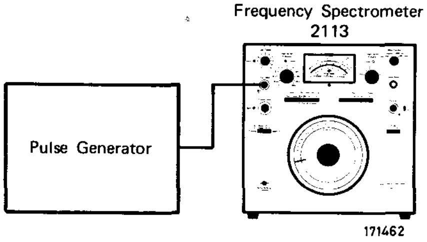

Fig.22. Alternative for set-up shown on Fig.5

References

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C.G. WAHRMANN: Impulse Noise Measurements B & K Technical Review 1969 no. 1.

Computer Interface and Software for On-Line Evaluation of Noise Data*)

by

F. Skøde

ABSTRACT

A mini-computer with the aid of some analog instruments renders itself extremely versatile for complicated acoustic measurements and for stationary noise control. Special external devices which make the system even more flexible are described. Some of the fields of application which are thus possible to undertake are discussed, with a particular example of reverberation time measurements illustrating the procedure.

SOMMAIRE

Un ordinateur miniature associé à certains instruments analogiques devient extremement versatile pour les mesures acoustiques complexes et pour le contrôle de bruits stationaires. Des équipements spéciaux donnant au systeme une flexibilité encore plus grande sont décrits. Quelques domaines d'applications devenues possibles sont diskutés, un exemple particulier relatif aux mesures de temps de réverbération illustrant la procédure.

ZUSAMMENFASSUNG

Ein Mini-Computer stellt im Zusammenhang mit einigen Analog-Geräten eine äußerst vielseitige Ausrüstung für komplizierte akustische Messungen oder für die stationäre Geräuschüberwachung dar. Spezielle externe Geräte, die eine solche Ausrüstung sogar noch flexibler werden lassen, werden beschrieben, sowie einige der Anwendungsgebiete für derartige Ausrüstungen diskutiert und die Meß- und Auswertverfahren durch ein einzelnes Beispiel für Nachhallzeitmessungen veranschaulicht.

Introduction

The present day increasing demands for reduction in manual work has resulted in the widespread use of data machines especially in the field of commerce. Although, in the technical field their use is often limited to routine calculations and process monitoring and control, the digital computer can be shown to be a very flexible tool yielding feasibility in evaluation of complicated data or time consuming measurements.

Computers with large memory capacity have the obvious advantage of being programmed in high order languages, however, they are stationary and are often used on a time-sharing basis. For real time measurements, either the

*) Paper presented at the 7th International Congress on Acoustics, Budapest, Hungary 18-26 August 1971.

measuring system would need to be close to the computer or the measured data would have to be transmitted to the computer by for example the public telephone network. On the other hand, mini-computers, about which this article is concerned, have the advantages of mobility and relatively simple interfacing with external devices. They can therefore very conveniently be utilized for data sampling and data reduction which can later be forwarded to computers of larger memory capacity for further computations. In many cases, however, a mini-computer alone (with suitable external devices) may be capable of solving a measurement and evaluation problem. In the following, special acoustic problems which can be solved by the aid of mini-computers are discussed.

A system containing a real time analyzer and a mini-computer is able to read-in a complete frequency spectrum in about 3 msec and then be able to perform calculations or store the spectrum in the memory for later computations. The measured data may then be used for comparison with a reference spectrum, for example, of noise emitted by a machine, to ensure that it does not exceed certain preset limits. Alternatively, the system could be made to read spectra at certain intervals over a period of time and then calculate the amplitude distribution of each frequency band over the time period. More complicated calculations of assigning some weighted value (which could not be carried out by a single filter of a real time analyzer) can also be accomplished e.g. Steven's calculation (ISO 532 method A), PNDB (ISO 1761, FAR 36), Zwicker (ISO 532 method B). For evaluation of EPNDB (effective perceived noise level) the PNDB value is calculated every

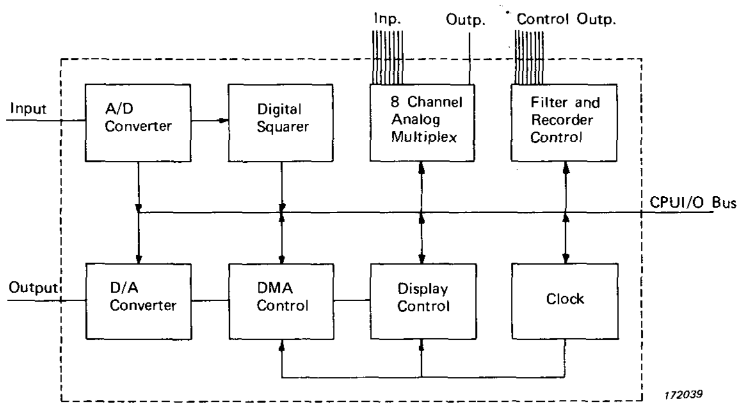


Fig. 1. Experimental unit with external devices

half a second and stored in the computer memory; when the fly-over is detected by the system, the stored values are integrated according to the recommendation and the result printed out on the teletypewriter.

External Devices

To aid the computer in satisfying the various requirements for acoustic problems, an experimental unit (see Fig. 1) containing eight external devices for the mini-computer was developed. Each external device is interfaced with the computer and can be controlled by the computer or externally.

A/D Converter

The analogue digital converter utilizes the method of successive approximation converting a DC signal to a nine-bit digital signal in 7 μsec including the functions sample and hold. This permits a maximum sample frequency of 130 kHz.

Squaring Unit

Here the nine-bit digital signal from the A/D converter is transformed into its sixteen-bit square valued in 0.2 μsec. The squaring procedure can, of course, be carried out by the computer itself, at the expense of 18 μsec of computer time. The maximum permissible sample frequency would however then be reduced considerably for calculations of for example true RMS or reverberation measurements.

Multiplexer

The analogue multiplexer consists of eight field effect transistors which can switch between analogue signals lying between $\pm$ 10 V in 2 μsec. The computer controls the scanning rate as well as the order of the signals to be connected to the common output.

Filter and Recorder Control

The filter control selects the frequency setting of eight individual third octave filter sets. The recorder control automatically starts and stops the level recorder as well as controls the lifting of the recorder pen.

D/A Converter

The digital analogue converter converts a ten-bit signal from the computer to an analogue signal in the range $\pm$ 10 V in 2 μsec. It can be activated either by the computer or by the Direct Memory Access Control.

Direct Memory Access Control (DMA)

The Direct Memory Access allows an external device to interchange data directly in the memory of the computer with no more interference than a

3.5 µsec stoppage for the running program. With the aid of the DMA control, the memory address formed by the display control is strobed into the computer after which the contents of the memory are strobed into the D/A converter.

Display Control
The display control can be used to form addresses for the DMA unit and to synchronise the data transfer with an external display unit which may be an oscilloscope, or a level recorder or the display unit of a real time analyzer.

Digital Clock
The digital clock contains a 4.096 MHz crystal-controlled oscillator and a number of frequency dividers which are set by a single data transfer from the computer, facilitating a choice of period times ranging from 0.25 µsec to 16 sec. The accuracy achievable of any period in this range is ± 0.125 µsec or ± 0.05% whichever is greater.

Fields of Application
The simplest application of the system is as a waveform generator. Any complex waveform which can be made up of a number of amplitudes as a function of time may be generated at the output of the D/A converter. The amplitude limits are ± 10 V while the resolution is 10 mV. Utilizing the DMA unit and the internal clock it is possible to transmit between 6.2×10^{-3} and 2×10^{5} values per second. The signal generated may also be an analog input signal which has been sampled by the A/D converter and stored in the computer. In this case the signal may be repeated as many times as required and at whatever rate desired.

Using the A/D converter and the squaring device it is equally easy to perform RMS calculations. Limitations for crest factor and dynamic range are set by the resolution of the A/D converter, while sampling frequency is limited by the evaluation time (for determining the mean value and its square root) set by the time constant chosen. The resulting RMS value can either be displayed as an analogue signal via the D/A converter or it can be printed on a teletypewriter.

Digital filtering techniques can be implemented since all the necessary hardware is contained in the system, however, the combination of filter complexity and the available frequency range is limited by the evaluation speed of the computer. The available dynamic range is determined by the resolution of the A/D converter. Nevertheless considerable investigations can be undertaken to establish what additions the required filter would need. The final filter implementation may have to be realised either fully by hardware or partly by hardware and partly by software.

Another field of application where the system is well suitable is the statistical distribution of signals. The percentage of the total time the signal lies in a particular amplitude window can be plotted against the amplitude on the display unit of a real time analyzer, for any third octave frequency band. Similarly the percentage of the total time the signal lies above a particular amplitude can also be displayed.

A typical application of the system with the aid of some additional analogue instruments is the measurement of reverberation time by Schroeder's method. For an upper frequency limit of 10 kHz it is possible to use three microphones placed at different locations in the room to be measured. The output from each microphone is filtered in third octave bands while the output of the filters are connected to the multiplexer. See Fig.2.

The frequency of the filters are set either manually or chosen by the software. A suitable pulse is sent from the computer via the D/A converter and a power amplifier to the loudspeaker. As soon as the pulse is over, the A/D converter starts sampling the signals from the three microphones, the values of which are squared and passed on to the computer. The computer then evaluates the mean of all the squared samples lying in intervals of, for example, 50 msec. After carrying this procedure out for a sufficient length of time integration is carried out by adding the last mean value to the previous one and replacing the previous one by the sum of the two, until all the samples are replaced by the sum of the sample itself and all the follow-

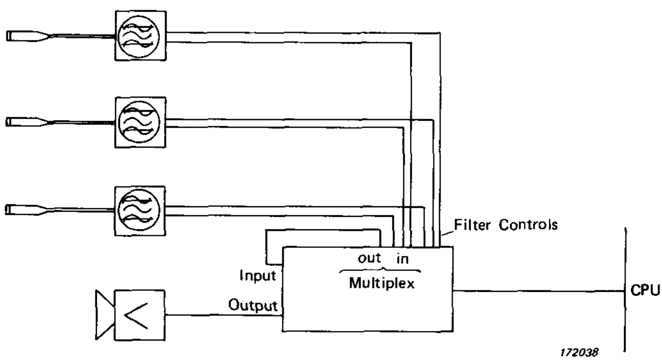


Fig.2. Reverberation measurement system

ing samples. Taking 10 times the logarithm of these values, the decibel values (50 msec apart) are obtained on the reverberation curve. The filters are then stepped to the next frequency and the procedure repeated until the curves for each frequency band are stored in the computer. The results may be displayed on an oscilloscope or recorded on a level recorder, or the reverberation time may be calculated and printed on the teletypewriter.

Conclusion

The above mentioned applications are just a few examples, as most analogue and digital instruments may be simulated by a computer and some software. Such systems are not only relevant in laboratories and educational centres, but also where a lot of routine work may be reduced by facilitating comparison of large amounts of data, for drawing final conclusions easily.

Reference:

S. BOEL PEDERSEN: Reverberation Time Measurements using Schroeder's Method and a Mini-Computer
Lecture given in Danish at Nordic Acoustic Meeting, Copenhagen 1970.

Brief Communications

The intention of this section in the B & K Technical Reviews is to cover more practical aspects of the use of Brüel & Kjær instruments. It is meant to be an "open forum" for communication between the readers of the Review and our development and application laboratories. We therefore invite you to contribute to this communication whenever you have solved a measurement problem that you think may be of general interest to users of B & K equipment. The only restriction to contributions is that they should be as short as possible and preferably no longer than 3 typewritten pages (A4).

Evaluation of Noise Measurements in Algol-60

by

Dr. Ing. K. Szymanski

Introduction

Noise measurements made in the field with a sound level meter often produce a large quantity of numerical results. Levels are read in dB for several working conditions, for nine octave filters ranging from 31 Hz to 8 kHz, and for A, B, C and linear ranges. It is sometimes necessary to repeat the measurements three or more times and obtain a mean value for the results. In addition, the results must be evaluated, for example, compared with the N85 level and tabulated as above or below by some number of dB.

If this analysis is done often, it is worthwhile to use a computer. Most computers can handle the Algol-60 language and therefore the following program is written in this language.

All that needs to be done is to:

1. *Give the programme to the computer*
2. *Give the data to the computer*
3. *Take from the computer the completed report of measurement, including the noise evaluation.*

Data for computer
The following programme will take the test data such as

1. *report number*
2. *machine type,*
3. *number of order,*
4. *family names of measuring group members,*
5. *date/day, month, year,*
6. *number of conditions under which the measurements are taken,*
7. *description of each condition*

Further comes the numerical data; either one number or three numbers to calculate mean value. In the first case a button, e.g. number 18, on the computer control desk must be switched. For two different conditions and three measurements of each value [(9 + 4) x 3] x 2 = 78 numbers represent the measurement data. There are: 9 octave bands, 4 ranges (A, B, C, Lin) 3 measurements and 2 conditions.

Example of complete data for computer

tm – 4 – 1
auto-crane
acra-crane factory 8972/71
G. Brown, T. Smith
January the 18th, 1971.

2.
raising load
lowering load
112, 110, 111,   110, 111, 111,   102, 104, 104
92,   97,   85,...113, 114, 114,   117, 118, 118.

Computer programme in Algol-60*)
See Appendix A.

Report of measurement
The report of measurement obtained from the computer gives, in a printed form in arrays, all the data which has been put into the computer. Further, the computer prints an array of differences between the mean values of the measurements, and the values of N85, for eight octave-bands (63 Hz to 8 kHz) and for each condition. The report ends with a conclusion that the highest measured level is higher or lower with respect to the N85-level and

*) There are small deviations of the programme necessary in some computers, those changes are not difficult to introduce. E.g. ge means greater than or equal to.

by how many decibels. If the level is higher than N85, a special array is printed which shows only the highest oversteps within a range of 3 dB from the top value. This array is printed for the octave bands and for each condition.

An example of report obtained from computer is shown in Appendix B.

Appendix A

Computer programme in Algol-60

```
begin integer i, j, k, m;
comment: NOISE EVALUATION PROGRAMME;
boolean array N1, N2, N3, N4, N5[1:10], W1, W2, W3, W4, W5,
              W6[1:21];
array N[2:9];
format ('ddd'); affix ('   ');
j:=2 ; for N[j]:= 102.6, 95.9, 91, 87.6, 85, 82.8, 81, 79.5
do j:=j+1 ;
read (N1,N2,N3,N4,N5,m);
space (10);
print ('NOISE MEASUREMENT REPORT', N1); line (2);
print ('TYPE OF MACHINE ', N2); line (1);
print ('ORDER NUMBER :', N3); line (1);
print ('MEASUREMENTS DONE BY :', N4); line (1); space (4); print (
'IN THE STATE INSTITUTE OF BLDG MECHANIZATION IN WARSAW'); line (1);
print ('DATE OF MEASUREMENTS :', N5); line (2);
print ('NOISE MEASUREMENTS ARE CARRIED OUT IN THE FOLLOWING', m,
'CONDITIONS :'); line (1);
if m ge 1 then begin read(W1); print(W1); line(1) end;
if m ge 2 then begin read(W2); print(W2); line(1) end;
if m ge 3 then begin read(W3); print(W3); line(1) end;
if m ge 4 then begin read(W4); print(W4); line(1) end;
if m ge 5 then begin read(W5); print(W5); line(1) end;
if m ge 6 then begin read(W6); print(W6); line(1) end;
begin array W[1:m, 1:13, 1:3], S[1:m, 1:13], P[1:m, 2:9];
comment IF 13xm DATA FROM MEASUREMENTS THEN SWITCH BUTTON 18,
             ELSE 39xm DATA; line (2); space (15);
print ('MEASUREMENT DATA'); line (1);
print ('
31HZ 63HZ 125HZ 250HZ 500HZ 1KHZ 2KHZ 4KHZ 8KHZ A B C LIN );
line (1);
if button (18) then go to ET1;
read (W)
for i:=1 step 1 until m do
begin line (2); print ('CONDITION ', i); line (1);
for k:=1 step 1 until 3 do
begin line (1); for j:=1 step 1 until 13 do print (W[i, j, k]);
end;
end;
```

```
line (2); comment MEAN VALUE CALCULATION;
for i:=1 step 1 until m do
for j:=1 step 1 until 13 do
S[i, j]:=(W[i, j, 1]+W[i, j, 2]+W[i, j, 3])/3;
if not button (18) then go to ET2;
ET1: read (S);
for i:=1 step 1 until m do
begin line (1); for j:=1 step 1 until 13 do print (S[i, j]); end;
ET2: line (1); space (10);
print ( ARRAY OF DIFFERENCES WITH RESPECT TO N85 ); format ('–dd,d'); line (1);
print ('
63HZ  125HZ  250HZ  500HZ  1KHZ  2KHZ  4KHZ  8KHZ');
for i:=1 step 1 until m do
begin line (1);
for j:=2 step 1 until 9 do begin
P[i, j]:=S[i, j]–N[j]; print(P[i, j]); end;
end; line (3); space (15);
print ('CONCLUSION'); line (3);
if max (P) gt 0 then
begin print ('ADMISSIBLE NOISE LEVEL EXCEEDED BY
                    '); print (abs(max(P))); print ('DECIBELS'); line (2);
print ('ARRAY OF CHIEF OVERSTEPS OF N85');
line (2); format ('–dddd');
print ('
CONDITION  63HZ  125HZ  250HZ  500HZ  1KHZ  2KHZ  4KHZ  8KHZ ');
line (1);
for i:=1 step 1 until m do
begin line (1); print (i); for j:=2 step 1 until 9 do begin
if P[i, j] lt ((max(P))–3) or P[i, j] le 0 then P[i, j]:= 0;
print (P[i, j]); end;
end;
end else
begin print ('THE HIGHEST NOISE LEVEL IS LOWER BY
                    '); format ('–dd,d');
print (abs(max(P))); print ('DECIBELS THAN THE ADMISSIBLE LEVEL'); line (2);
end;
line (2); space (25); print ('THE END');
line (20);
end;
end;
```

Appendix B

Example of report obtained from computer

noise measurement report tm–4–1
type of machine auto crane ac–303
order number acra crane factory 8972/71
measurements done by g. brown, t. smith
in the state institute of bldg mechanization in warsaw
date of measurements january the 18th 1971

noise measurements are carried out in following 2 conditions
raising load
lowering load

measurement data

31hz	63hz	125hz	250hz	500hz	1khz	2khz	4khz	8khz	a	b	c	lin
condition		1										
112	110	102	92	87	83	84	84	78	94	105	112	114
110	111	104	97	90	86	82	77	74	94	105	112	114
111	111	104	95	89	86	82	78	72	93	104	112	113
condition		2										
117	105	100	100	93	90	86	83	80	96	104	113	117
117	105	101	99	94	89	86	84	84	96	103	114	118
118	108	105	98	98	90	86	84	80	96	104	114	118

array of differences with respect to n85

63hz	125hz	250hz	500hz	1khz	2khz	4khz	8khz
8.1	7.4	3.7	1.1	.0	–.1	–1.3	–4.8
3.4	6.1	8.0	7.4	4.7	3.2	2.7	1.8

conclusion

admissible noise level exceeded by
8.1 decibels

array of chief oversteps of n85

conditions	63hz	125hz	250hz	500hz	1khz	2khz	4khz	8khz
1	8	7	0	0	0	0	0	0
2	0	6	8	7	0	0	0	0

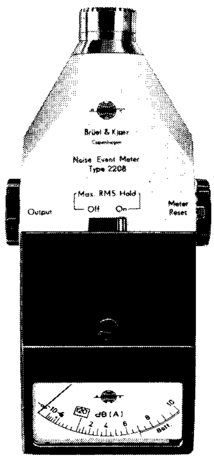
the end

News from the Factory

Noise Event Meter Type 2208

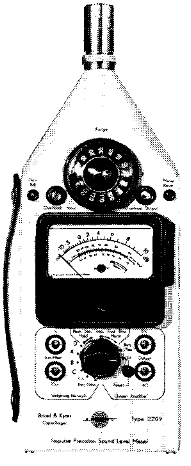
The Noise Event Meter is essentially a normal sound level meter equipped with the special feature of being able to hold the maximum meter deflection occurring during a measurement. In addition, the meter setting can be stored by a simple switch operation. These features are especially valuable for measurement of traffic noise or whenever the maximum noise level of a passing event must be measured.

In the standard version it is equipped with a piezoelectric microphone to meet all standards for ordinary (A weighted) sound level meters. However, in conjunction with a B & K 1/2" condenser microphone and adaptor it performs in accordance with the full precision standards (IEC Recommendation Publication 179).



Impulse Precision Sound Level Meter Type 2209

The new Impulse Precision Sound Level Meter Type 2209 fulfils all the existing standards for impulse and precision sound level meters, providing the sound and vibration specialist with everything desirable in a compact portable instrument. "Peak Hold" with 20 μsec rise time as well as "Maximum RMS Hold" facilities, with crest-factor capability up to 40, are incorporated, together with the A, B, C, D weighting and linear networks. The meter response may be adjusted to Slow, Fast, Impulse, Impulse Hold or Peak Hold as desired. The low inherent noise level permits measurement of sound levels as low as 15 dB while the frequency range 2 Hz to 70 kHz extends far beyond the audible range. Connection of B & K Filter Set Type 1613 makes the instrument a 1/1 octave audio frequency analyzer while connection of an accelerometer with an integrator readily converts the instrument to a portable vibration meter. 20 interchangeable meter scales (supplied with the instrument) facilitate direct reading of sound or vibration levels depending on the transducer used.



Digital Event Recorder Type 7502
To provide a better means for shock analysis and for some types of low frequency measurements than achievable by electro-mechanical recording devices such as tape recorders Brüel & Kjær have developed a purely electronic Digital Event Recorder, Type 7502 for which the input and output signals can either be analog or digital. Basically the instrument contains a sampling circuit, an analog to digital converter a memory and a digital to analog converter with associated command and control circuitry.

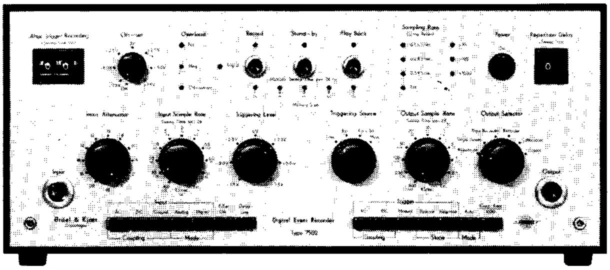
In the recording mode of the instrument the input signal is continously sampled and converted to digital numbers which are transferred through the memory by a clock signal. The signal again appear at the output with a delay determined by the number of memory positions and the clock frequency.

A part of the signal can be selected and stored for analysis purposes by triggering the instrument either manually or automatically by the signal itself (e.g. a shock) or by an external signal. By the setting of a counter the recorded part (one sweep length) can be selected to start from one sweep length before to nine sweep lengths after the triggering time. This feature facilities the recording of for example the first part of a pulse or the delayed response of a system to a given excitation.

The recorded signal can now be replayed by recycling continuously the stored information through the memory and, as the input sampling rate can be varied between 100 and 100,000 samples/second (even slower by an external generator) and the output rate at playback can be varied between 0.5 and 500,000 samples/second, it can be seen that a large range of frequency transformation ratios is available.

In the "Auto" mode the instrument automatically switches between the recording and the playback mode to facilitate permanent recording on a tape recorder or a tape punch of signals which exceed the triggering level of the instrument.

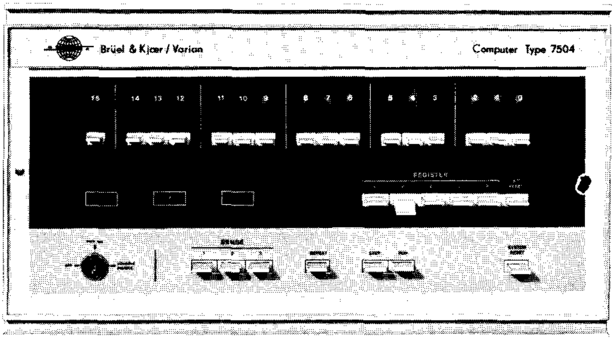
The standard version has 4096 (4K) 8 bit memory positions but the memory can be reduced to 2K or increased to 10K in 2K steps.



Computer Type 7504

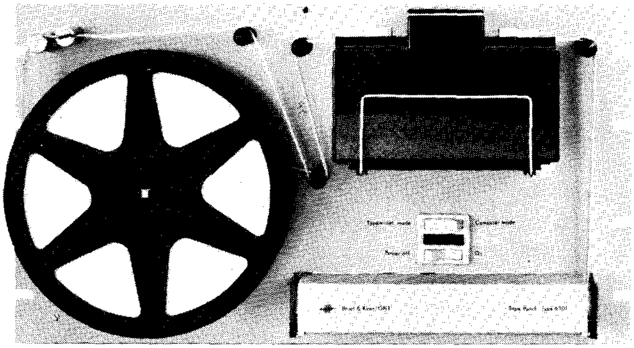
The Brüel & Kjær, Varian Computer Type 7504 has been interfaced to be especially suitable for use with acoustic and vibration measuring instruments, such as the Real Time Analyzer Type 3347. Although all the standard software developed for the Varian Computer 620/L for numerical calculations are available, special programs have been developed to carry out on-line measurements of acoustic and vibration problems.

The use of TTL and DTL digital integrated circuits not only account for high performance and reliability, but also makes the instrument a light and compact instrument (occupies only 10.5 inches = 26.7 cm in a standard 19" rack). The memory capacity is 4K, 16 bit words expandable in 4K steps to 8K in the computer and to a maximum of 32K with one additional 10.5 "chassis". The computer contains 9 hardware registers and has a full cycle execution time of 1.8 μsec i.e. over 500,000 cycles/sec. Also the interface for all the peripheral devices mentioned below are built into the computer, while basic and fortran compilers for 8K version are available making the computer extremely versatile both for laboratory experiments and in field applications, some of which are described in the articles by F. Skøde and H. Melchior in this issue of the Technical Review.



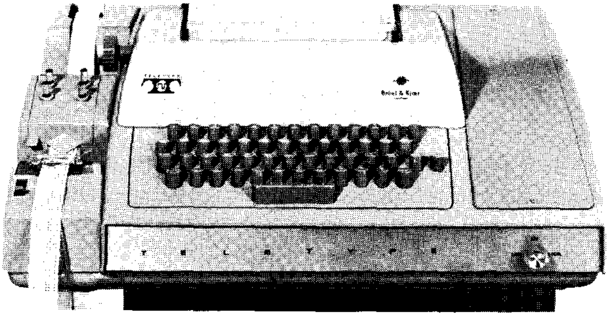
Tape Punch Type 6301

The Tape Punch Type 6301 is a version of the GNT Tape Punch model 34 supplied with an interface for operating together with Computer Type 7504, Real Time Analyzer Type 3347 or a Digital Encoder Type 4421. Since it punches up to 75 characters per second it is well suited for use in set-ups where a high punch speed is required, either when large amounts of data are to be read out or when time of operation on-line has to be cut down. It punches standard 1" paper tape with 8 information channels according to one of the three standard codes namely ASCII. Fridens Flexowriter, or IBM TTC/8. Besides, it offers two formats, computer format or typewriter format which can be chosen by means of a switch.



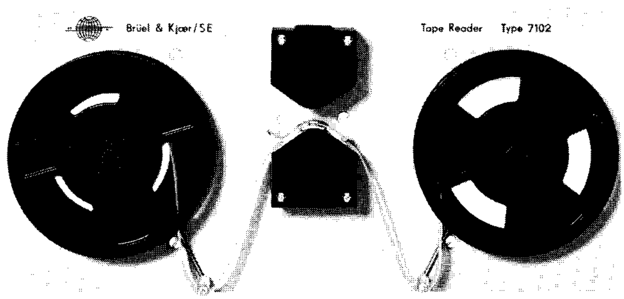
Teletypewriter Type 6401

For communication between the computer and programmer "Teletype" Type ASR 33 has been modified to operate with the computer Type 7504. The teletypewriter which is of the automatic send-receive type is equipped with a paper tape punch and a 1" tape reader. While the tape punch has facilities for deleting errors, the reader is equipped with an "end of tape" mechanism that automatically stops the reader when the end of the tape is reached. The "Local" mode of the teletypewriter is used for composing, punching and correcting programs while "On-Line" mode is used for reading in programs to the computer and reading out the results. The teletypewriter which uses standard ASCII code and handles up to 10 characters per second can transmit information to the computer either manually by the keyboard or automatically through the reader. However, to provide errorless tape for fast transmission to the computer, data can be punched off-line.



Tape Reader Type 7102

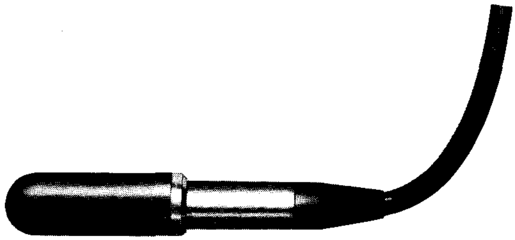
The Brüel & Kjær/SE Tape Reader Type 7102 which is well suited for fast reading-in of programs and data to the Computer Type 7504 has a reading rate from 0 – 125 characters per second in both directions. The instrument reads standard 1" tapes (opaque and translucent of maximum 40% light transmission) optically, by means of photodiodes achieving high speed and minimum wear of tape. Since the computer contains the necessary interface, the tape reader can be connected directly to the computer. Besides the simple tape loading and high speed spooling features, the tape reader has TTL and DTL compatible inputs and outputs.



Standard Measuring Hydrophone Type 8100

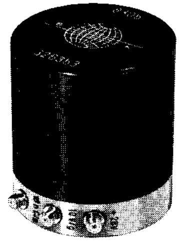
The Standard Measuring Hydrophone Type 8100 is a wide range underwater transducer for making absolute sound measurements over the frequency range 0.1 Hz to 200 kHz with a receiving sensitivity of −110 dB relative 1 Volt/μbar. The hydrophone assembly employs lead zirconate titanate as the active sensing element. Special measures have been taken to obtain good electrical shielding enabling the calibration and use of these hydrophones in air. An extensive calibration and temperature stabilizing procedure has been undertaken in order to ensure completely predictable performance and stable operation, both in the laboratory and under field conditions. The moulded neoprene rubber boot is permanently bonded to a monel support which is electrically and vibrationally isolated from the sensing element. The hydrophone is equipped with a 6 m waterblocked low noise cable and waterproof extension connector. A 1.2 m cable is included with waterproof connector one end and a B & K standard plug on the other. A series of low noise waterproof extension cables, AO 0104, AO 0105 and AO 0106 of 10, 30 and 100 m lengths, respectively, which meets MIL-C-915 are also available.

The broad frequency response and relatively high sensitivity offer the user a wide range of applications. Although Type 8100 is primarily intended for the calibration of other hydrophones, it can also be used to project recorded signals, examine output patterns of sonars, measure cavitation noise, measure sea state spectra, and many other acoustic phenomena occurring in gaseous and liquid media. To overcome the problems of water tank calibration a gated calibration system has been developed which may be of interest for application in closed compartments etc.



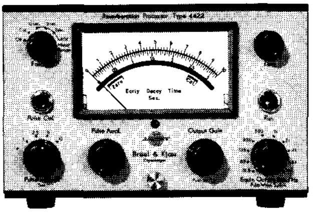
Accelerometer Type 8306

The requirements placed on measurements of vibrations of large structures such as buildings, bridges ships, etc. as well as seismic investigations are rather demanding. This is chiefly because the vibration levels concerned are very low and so are the frequencies. To be able to measure such vibrations accurately, Brüel & Kjær have developed a carefully designed Accelerometer Type 8306 with a built-in preamplifier (impedance converter of unity gain 0 dB) to measure down to levels of 2×10^{-6} g in the frequency range 0.2 Hz – 1000 Hz. Since the accelerometer is of the Uni-Gain Type, the voltage and charge sensitivities of 10 Volt/g and 10,000 pC/g respectively have been adjusted to 2% of these values. The preamplifier also incorporates an active low pass filter (18 dB per octave attenuation above 1000 Hz) making the accelerometer insensitive to its own resonance and external high frequency vibrations.



Reverberation Processor Type 4422

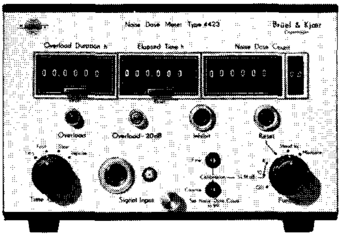
The traditional method of evaluating the acoustic quality of a room by a evaluating the reverberation occurring after the noise of a sound source is interrupted, suffers from the statistical uncertainty of the excitation noise. To overcome this limitation Brüel & Kjær have developed a Reverberation Processor Type 4422 which utilizes the method of "Integrated Tone Burst" suggested by M.R. Schroeder and modified by H. Kuttruff. Since this method yields very smooth reproducible reverberation decay curves, the initial slope of the curve, which is of special importance when evaluating the acoustic quality of the room, can be determined more accurately in the frequency range 20 Hz to 20 kHz. The reverberation time extrapolated from the initial slope is termed "Early Decay Time" and can be read off directly from the meter of the instrument.



Noise Dose Meter Type 4423

The Noise Dose Meter is used in conjunction with sound measuring equipment to determine the "Equivalent Continuous Sound Level" L_{eq} according to the ISO Recommendations R 1996 and R 1999 and of DIN 45 641 (draft April 1971). The Noise Dose Meter which has a frequency range of 20 Hz to 20 kHz is equally suitable for assessment of risk of hearing impairment and annoyance from industrial or road and air traffic noise. Any of the B & K Sound Level Meters, Measuring Amplifiers, Spectrometers and Analyzers may be used in the measuring arrangement.

Further information about the Noise Dose Meter is given in the paper "Noise Dose Measurements" in the next of the Technical Review.



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(Continued from cover page 2)

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As shown on the back cover page Brüel & Kjær publish a variety of technical literature which can be obtained free of charge. The following literature is presently available:

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- Acoustic Noise Measurements (English),) 2. edition
- Power Spectral Density Measurements and Frequency Analysis (English)
- Standards, formulae and charts (English)
- Lectures and exercies for educational purposes
- Instruction manuals (English, some available in German, French, Russian)
- Catalogs (several languages)
- Product Data Sheets (English, German, French, Russian)

Furthermore, back copies of the Technical Review can be supplied as shown in the list above. Older issues may be obtained provided they are still in stock.

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