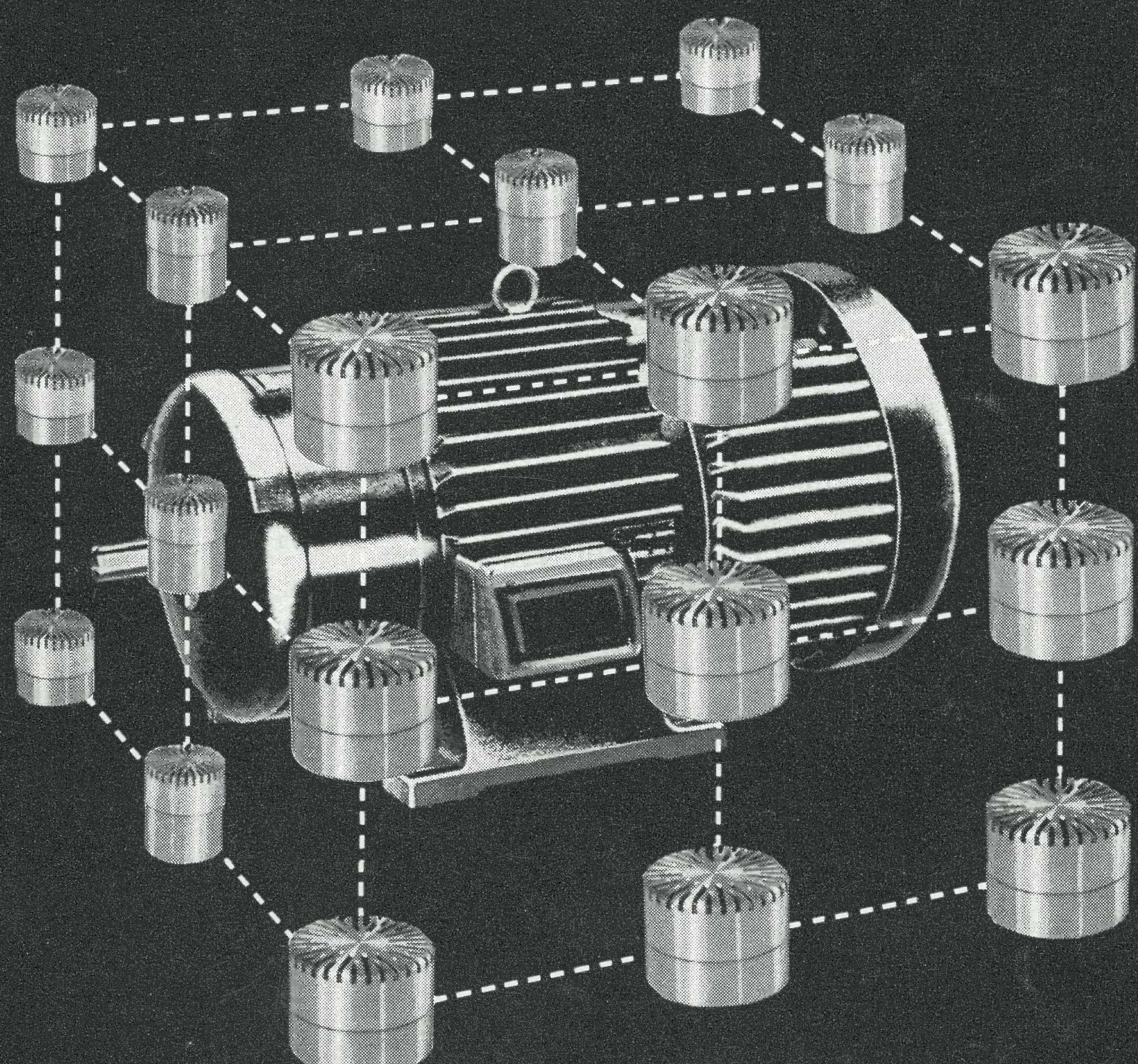
Brüel & Kjær

Technical Review

To Advance Techniques in Acoustical, Electrical, and Mechanical Measurement

Computer Evaluation of Acoustic data



1/3 OCTAVE IMPULSE MEASUREMENTS

No. 1 1970

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TECHNICAL REVIEW

No. 1 - 1970

Contents

Acoustic Data Collection and Evaluation with the Aid of a Small Computer

by

Franz Herz, Hans Stawinski and *Walter Elsasser**)

ABSTRACT

Experimental acoustic investigations with a large number of measuring positions such as are carried out on electrical machines, are very time-consuming because of the time spent collecting and averaging the data. To overcome this disadvantage a digital computing system was developed with a small computer as the central element. The analogue data are converted to digital form and fed to the computer "on line" for evaluation An analysis of the results is printed out at the end of the measurement on an associated Teletypewriter. The kind of evaluation depends on the programme used, which is read into the computer memory from an 8-channel punched tape. With different programmes, widely different problems like airborne noise, vibration measurements etc. can be evaluated in a very short time.

SOMMAIRE

Les expérimentations acoustiques, comportant un grand nombre de points de mesure, telles qu'elles sont pratiquées, par exemple, sur les machines électriques, sont très longues, à cause de la durée de l'acquisition et du traitement des données. Pour compenser a disavantage un systeme de calcul digital a été developpé. Il comporte un petit calculateur comme elément principal. Les données analogiques sont converties sous forme digitale et fournies au calculateur, placé «en série», pour évaluation. Une analyse des résultats est imprimée à la fin de la mesure sur un Téléinscripteur associé. Le type d'évaluation dépend du programme utilisé, ce dernier est lu dans la memoire du calculateur à partir d'une bande perforée à 8 pistes. Ces differents programmes, permettant la résolution de problèmes largement diversifiés, tels que bruits sérians, mesure de vibration etc. , en un temps très court.

ZUSAMMENFASSUNG

Experimentelle akustische Untersuchungen mit zahlreichen Meßpositionen, wie sie beispielsweise an elektrischen Maschinen durchgeführt werden, sind infolge zeitraubender Ablesung und Mittelwertbildung sehr aufwendig. Zur Behebung dieses Nachteils wurde eine digitale Auswerteanlage mit einem Kleincomputer als Hauptelement entwickelt: Die Meßgrößen in analoger Form werden über einen Analog-digital-Wandler dem Rechner »on-line« zugeführt und ausgewertet Nach Beendigung der Messung steht ein vom zugehörigen Blattschreiber ausgedrucktes Klartextprotokoll zur Verfügung.

Die Art der Auswertung ist abhängig vom verwendeten Programm, welches von einem 8-Kanal Lochstreifen über eine Leseeinheit in die Recheneinheit übertragen wird Mit verschiedenen Programmen können damit die unterschiedlichsten Aufgaben wie Luft- und Körperschallmessungen etc. innert kürzester Zeit bewältigt werden.

Introduction

For many industrial products as well as rotating electrical machinery the noise requirements are becoming more and more important. The measurement of machinery noise is today a type-approval test of similar importance to that

*) Brown Boveri, Baden, Switzerland.

of the temperature coefficient or efficiency. The current stage of development of noise measuring equipment allows quick execution of the measurements. The time necessary for evaluation of the results is, on the other hand, always relatively long. For example, if a measurement such as that specified in Ref. [1] Part II is performed (17–23 microphone positions), the ratio of measuring time to evaluation time is about 1 : 4. This is true generally for all acoustic investigations in which time and spatial averaging of frequency analyses is necessary. A significant reduction in data evaluation time can be obtained by the introduction of a data collection system which calculates the required result directly.

The measurements which are undertaken in our laboratory [2] make the following requirements of such a system, as well as the reduction in evaluation time:

a) The data collection and evaluation should take place without the necessity of an external computer, because the results of the measurement can decide under certain circumstances on a continuation of the investigation. At the end of the measurement a complete report with results in clear print should be made available.

b) The set-up should not only allow standard noise measurement procedures to be carried out, also other acoustic investigations, such as noise in solids, insulation measurements, etc., should be possible.

c) The signal from the Analogue Voltage Readout of the Level Recorder varies with time, especially at low frequencies. This signal must be optimally averaged for the sake of accuracy.

A computational system which meets these requirements is described below. The essential part of this system is a small computer which is conceived chiefly as a process computer. Together with an analogue-to-digital converter as a link to the sound measuring instruments on one side, and with a Teletypewriter as input and output terminal on the other side, the computer can be used for acoustic data handling and for solving numerical problems. The possibility of programming allows a large number of different types of measurement to be made optimally. Once a programme has been made, the measurement and evaluation can be carried out by semiskilled personnel. At the present moment there exist programmes for noise measurements after Reference [1] Part II, airborne sound insllation measurements, spatial averaging of narrow band analysis and vibration.

Description of the System

Fig. 1 shows the main parts of the system, where I, II and III are normal B & K instruments. The computational system consists of the following instruments, from Digital Equipment Corporation:

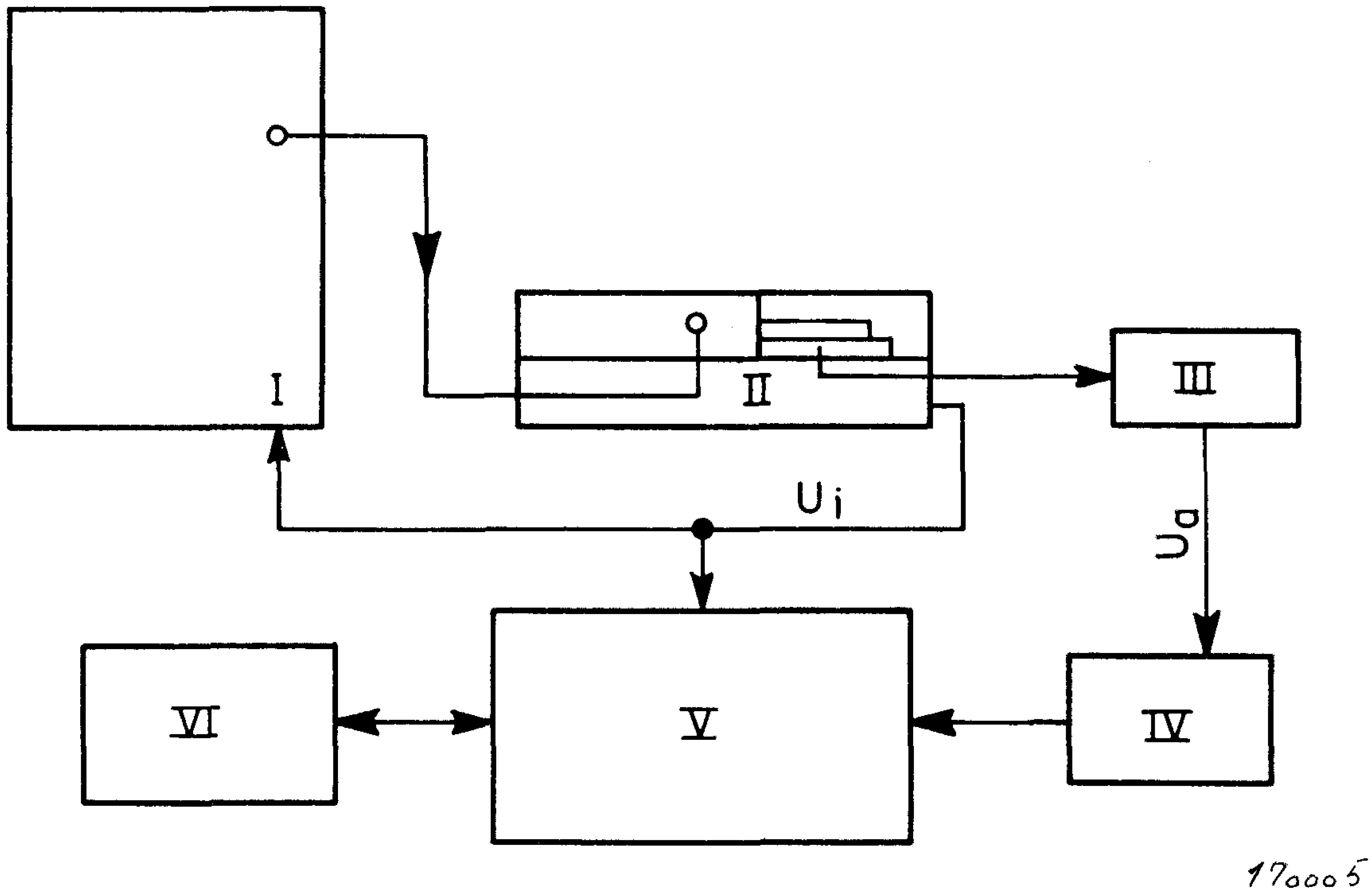


Fig. 1. The build-up of the System.

I 1/3 octave analyzer 2112.
II Level Recorder 2305.
III Analogue Voltage Readout ZR 0021.
IV Analogue-Digital Converter ADO 8A.
V Digital Computer PDP 8/S.
VI Teletypewriter ASR 33.
U_a Analogue voltage.
u_i Switching impulse.

Analogue to Digital Converter Type ADO 8A (IV)

This instrument converts analogue dc input signals into digital information which is fed to the computer.

Operating range:	0 . . . 10 V
Word length:	10 bits
Resolution:	10 mV (0.05 dB when using the 50 dB potentiometer in the Level Recorder)
Conversion rate:	max. 100 kHz
Conversion principle:	Successive approximation

Digital Computer Type PDP 8/S. (V)

This is a serial computer with a core-memory, designed for data handling and for process control. (4)

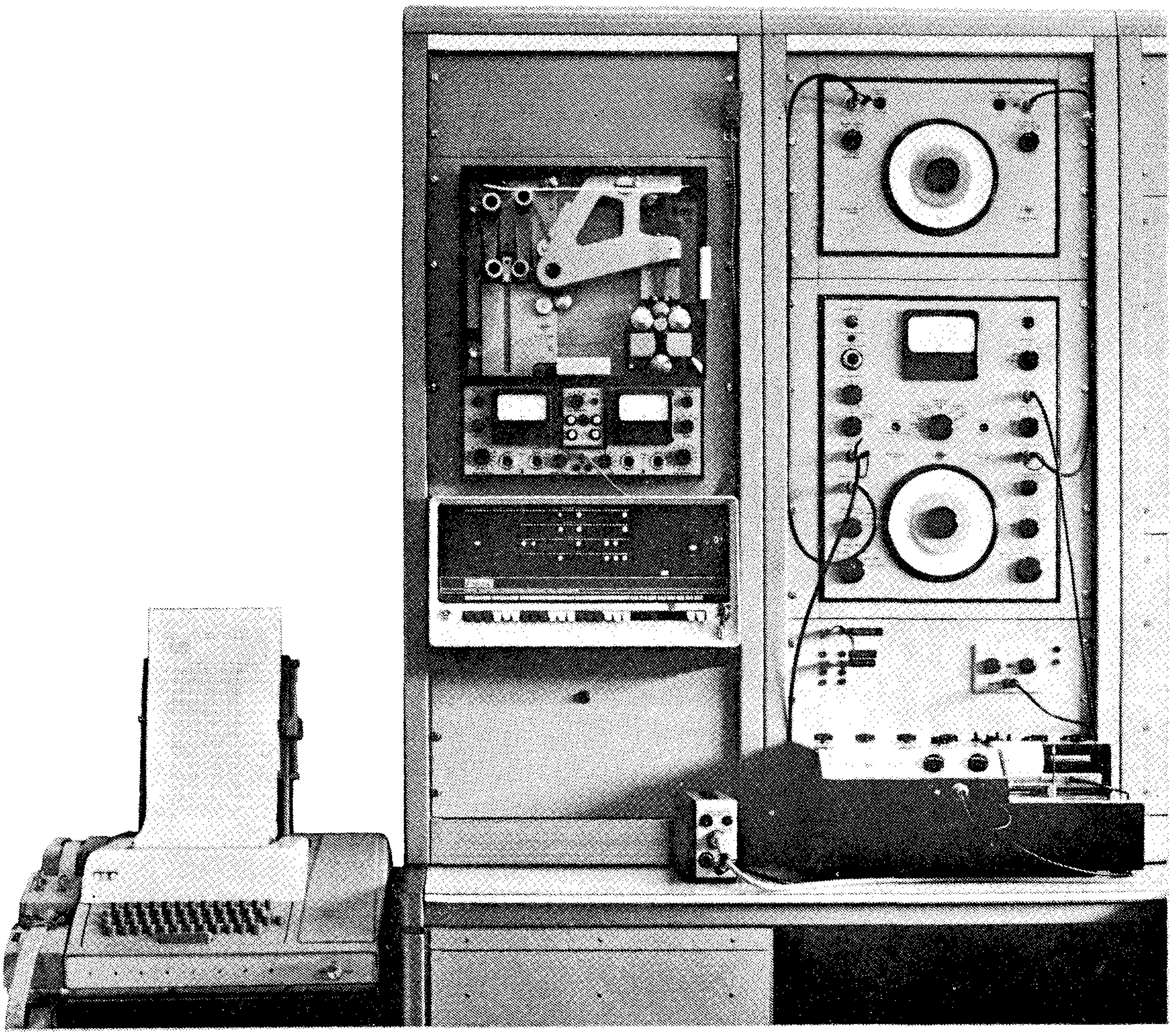
Memory capacity:	4096 words
Word length:	12 bits
Addition time:	36 μs
Cycle time:	8 μs

Up to 64 peripheral devices can be connected to the PDP 8/S.

Teletype Type ASR 33 (VI)

The Teletype ASR 33 is the input and output terminal of the PDP 8/S system. It is furnished with perforated tape punch and reader. (8 channel ASC II code). Speed: 10 charactrs per second.

Fig. 2 shows the complete installation.



170009

Fig. 2. Digital Analysis System.

The photograph shows from right to left:

Noise measurement instrument, Digital computer (below the Tape Recorder), Teletypewriter.

(The Tape Recorder does not belong to the evaluation system).

Principal Operating Functions

The PDP 8/S functions will be described with reference to the measurement of noise, which is the most important application in the author's laboratory. The 1/3-octave filter set (Type 1612 or 2112) has 40 contiguous measurement intervals (Fig. 3) of 1/3 octave. It was decided that the 1/3 octave bands below

50 Hz (Intervals 1–6) and above 20 kHz (Intervals 34–36) were of minor importance for electrical machinery, and only 31 intervals were evaluated in the analyses.

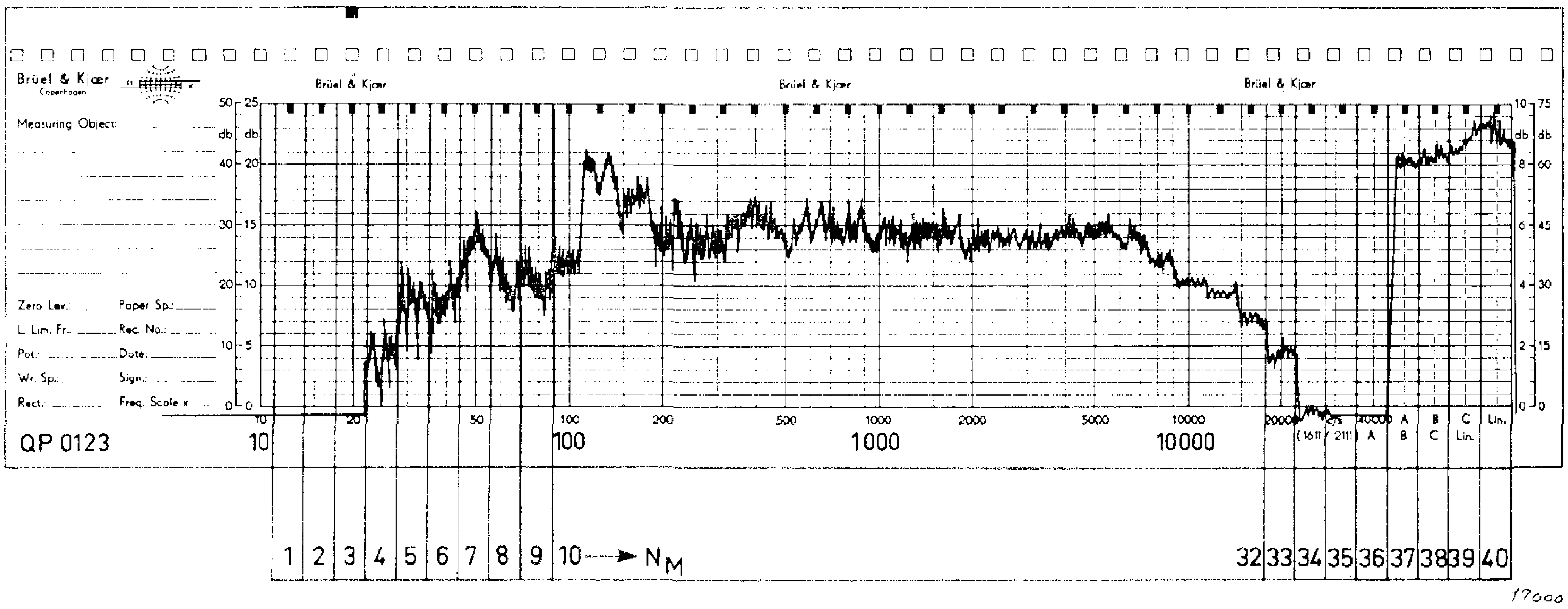


Fig. 3. 1/3 Octave Spectrum.

N_m Interval number.

For each interval (1/3 octave band) the Analogue-voltage Readout of the Level Recorder delivers to the analogue-digital converter, a dc voltage which is proportional to the respective sound pressure level. This analogue voltage is converted into digital information and fed to the computer. For each interval m = 80 conversions are made with a sampling period determined according to the requirements explained in the next section. After running through a 1/3 octave spectrum, digital information corresponding to the sound level in each interval is stored in the computer. The computer then calculates a time average for each interval according to the following:

$$L_i = 10 \log \frac{1}{m} \sum_{j=1}^{m} 10^{0.1 L_{ij}} \quad \text{(dB)} \qquad (1)$$

where L_{ij} = j-th instantaneous value in the interval (dB)
L_i = time average of each interval in measuring position i (dB)
m = 80, the number of instantaneous values sampled for each interval.

The 31 average values L_i are then also stored. The same procedure is repeated for each measuring position. After measuring at the last position, the spatial average L_M for each of the intervals is calculated according to the relation:

$$L_M = 10 \log \frac{1}{n} \sum_{i=1}^{n} 10^{0.1 L_i} \quad \text{(dB)} \qquad (2)$$

where L_i is as above
L_M = spatial average of the 1/3 octave bands (dB)
n = number of measuring positions.

Finally the analysis results in tabulator and graphic form are printed out on the Teletypewriter.
The data transfer is initiated by the switching impulse of the Filter Set as indicated in Fig. 4.
For the present the main programme next remains in a waiting loop, and the switching impulse (see Fig. 5) now sets the Flip-Flop in the 1-state, and the 1-output of the Flip-Flop supplies the signal "Interrupt-Request". This signal initiates a subroutine (e.g. conversion and storage of the voltage level). When the subroutine is finished, it resets the Flip-Flop and a return to the waiting loop of the main programme follows. The system is then ready for another switching impulse which initiates the procedure for the next 1/3 octave band.

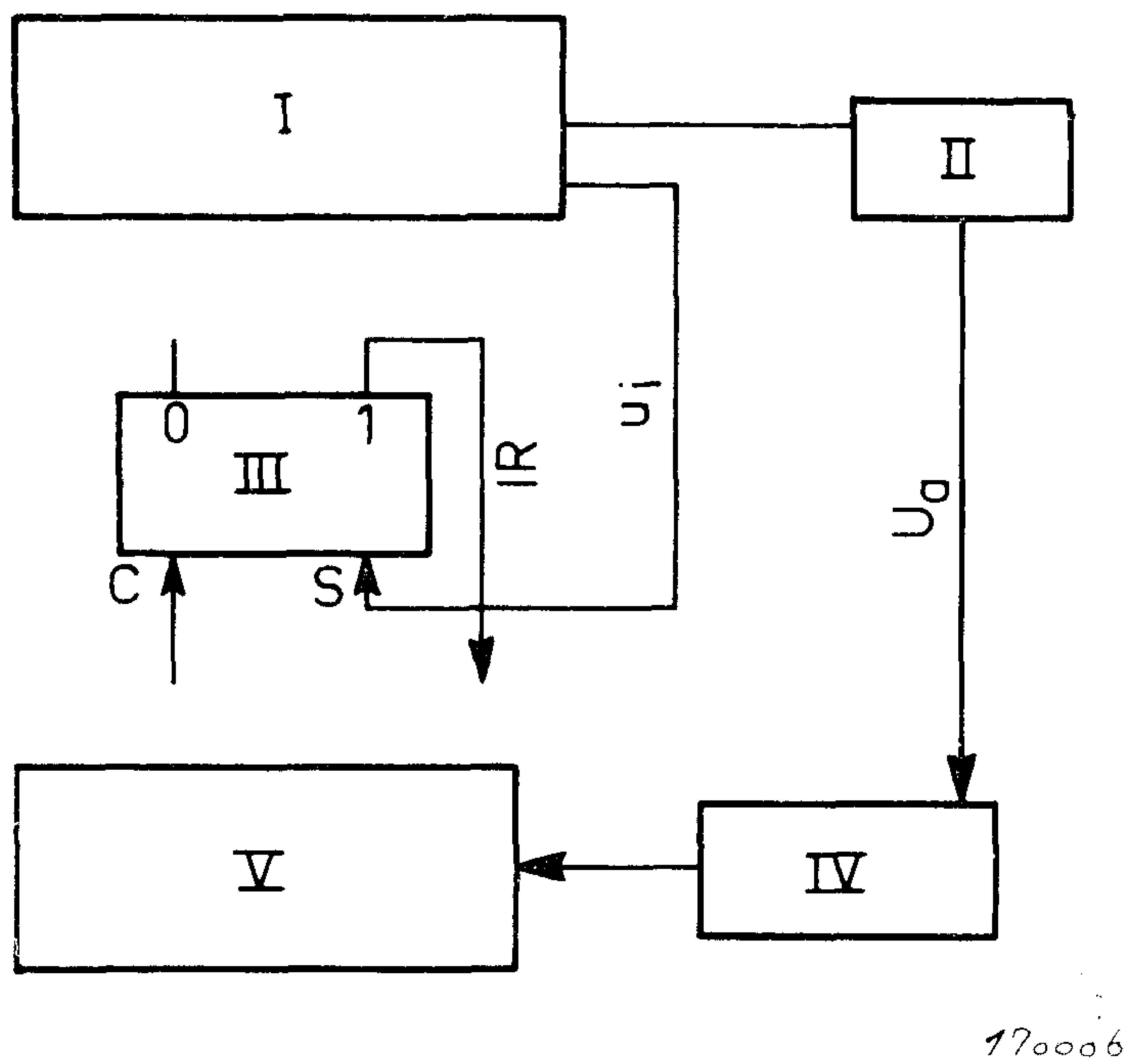


Fig. 4. Control of the data collection.

I	*Level Recorder 2305.*	*V*	*Digital Computer PDP 8/S.*
II	*Analogue Voltage Readout ZR 0021.*	*IR*	*"Interrupt-Request".*
III	*Flip-Flop C = Clear.*	U_a	*Analogue Voltage.*
	S = Set.	u_i	*Switching Impulse.*
IV	*Analogue-Digital Converter ADO 8A.*		

Timing of the Data Transfer

Fig. 5 shows the timing of the data-transfer. This is determined essentially by the writing speed and paper speed of the Level Recorder. A standard writing speed of 250 mm/s was chosen for the analyses programme (lower limiting frequency of Level Recorder: 20 Hz). The start of the 80 samples must be delayed slightly because of the time required by the writing system to reach the new level. The dead time T_t should allow the writing system time enough to adjust to the largest possible level change of 50 dB i.e. 200 ms. The total time T_s of the band is determined by the paper speed, which was fixed at 3 mm/s, or 10 mm/s for special cases. The 80 samples can be taken in the time T_u, a constant sampling frequency giving a constant period T_w. At the end of each 1/3 octave band is a short time T_e during which a part of the computation proceeds.

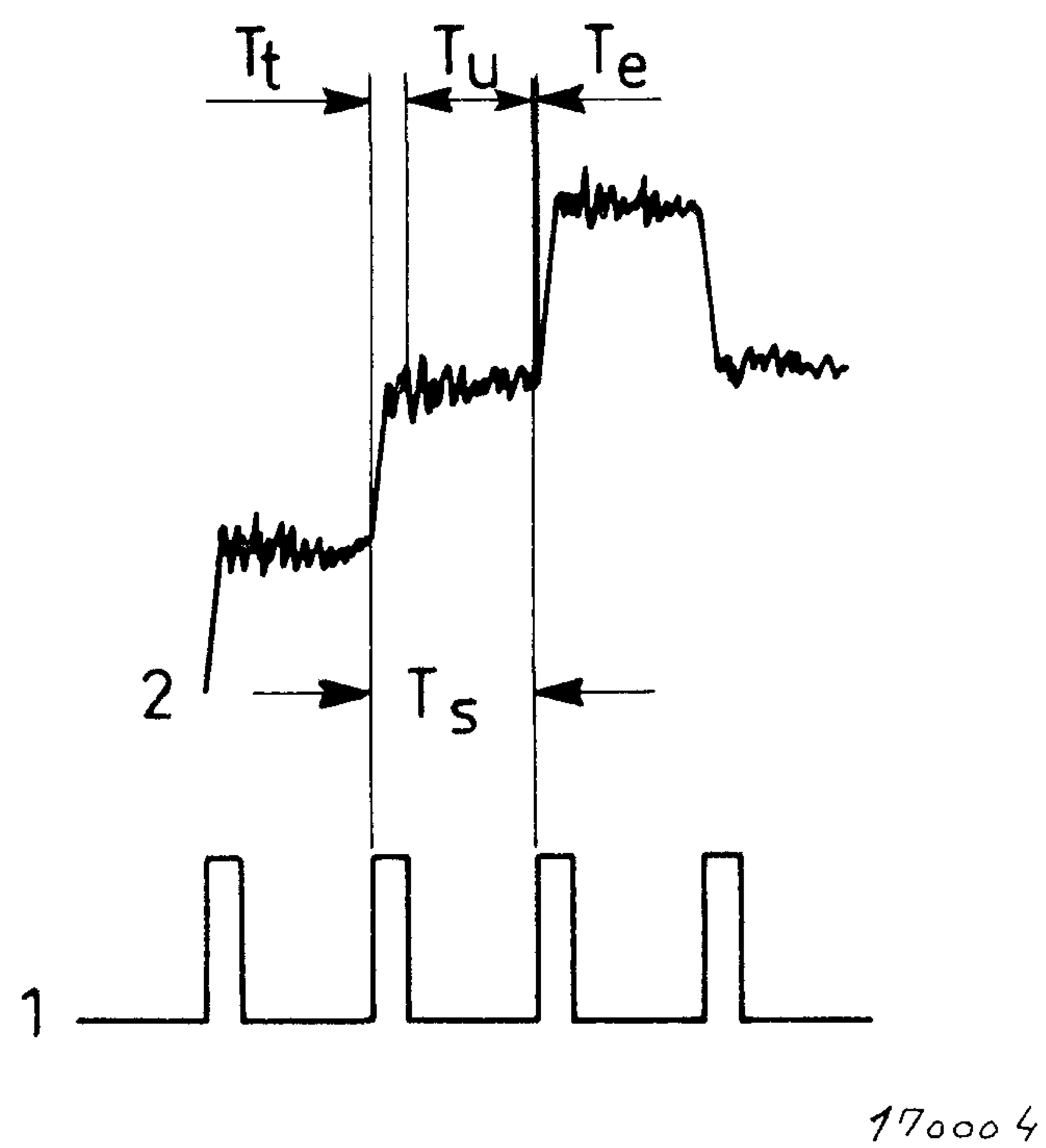


Fig. 5. Timing of the Data Transfer.

1 Switching impulse.
2 Sound pressure level in 1/3 octave bands.
T_s Time of one band (1666/500 ms).
T_t Dead time (200/200 ms).
*T_u Total sampling time (1446/280 ms) $T_u \simeq$ **80** T_w.*
T_w Sampling time (18/3.5 ms).
T_e End time (20/20 ms).

The times in brackets are for 3–10 mm/s paper speed.

Programming

The relatively small memory capacity and the fact that we are using a serial computer with relatively long cycle-time require for certain programming problems special solutions. The storage requirement and computing time could then be considerably limited.

The computer was programmed in machine language. Among other instructions, the machine language also enables programme interruption (INTER-

RUPT) by peripheral equipment, and data-transfer. Additional software (FLOATING-POINT SYSTEM) makes available arithmetical operations and the most important functions as subroutines. The operations proceed in floating-point arithmetic with the operand given as a 23-bit mantissa and 11-bit exponent. The accuracy obtained is 7 decimal places.
For solution of numerical problems, the computer can also be programmed in FORTRAN or in a system-oriented language (FOCAL), but because of the memory capacity the range of such programmes is limited.

Description of Programme

Of the existing programmes, the one "Noise measurements on Electrical Machines" is the most important, and is based on the current recommended test code [1]. In the event of any revisions, the programme can be easily changed. The programme controls the following measuring system functions:

a) Collection "on-line" of the measurement information.
b) Evaluation of the collected data. The following calculations are carried out:
   Time average of sound pressure level in each 1/3 octave band for each measuring position, after eq. (1).
   Spatial average of sound pressure level for each measuring position after eq. (2).
   Sound pressure level in octave bands.
   Sound pressure level in 1/3 octave and octave bands with respect to a reference radius of 3 meter.
   Sound power level in 1/3 octave and octave bands.
c) Printout of full measurement report with results of calculations according to b).

Fig. 6 shows a flow-diagram of the programme. Certain characteristic features of the programme are explained below.

Transfer by Teletypewriter

In order to keep the memory requirements for the printout to a minimum, the ASC II coded Alphanumeric symbols are standardized in 240_8*), which allows storage of 2 symbols in one computer-word. In addition to the machine and test data the machine dimensions are read-in. These are necessary for the calculation of the referred sound pressure level. In our acoustic laboratory a test facility with free-field conditions and another with free-field conditions over a reflecting plane are available [2], and at the beginning of a measurement a choice between them must be made. This s done by putting an index number into the computer. Later in the evaluation, the different spatial characteristics are then taken into consideration.

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*) The octal counting system is used in the computer organization.
The indices $\ldots_8$, $\ldots_{10}$ indicate the octal or decimal system respectively.

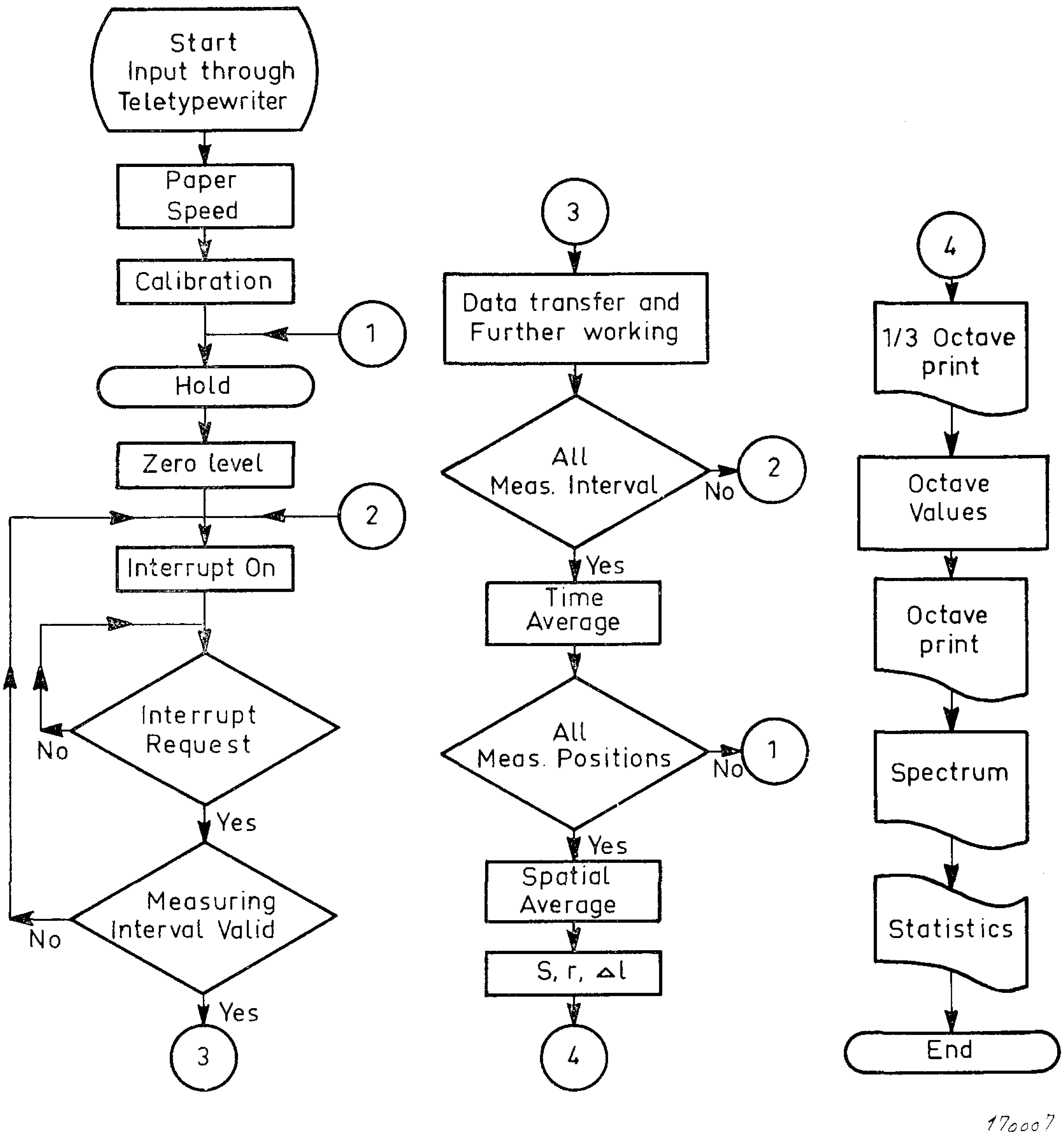

Fig. 6. Flow diagram.

Paper Speed

The duration of the dead and waiting times is controlled by several executions of the counting loops. The chosen paper speed is indicated by a certain combination of the computer switch register. The switch register is interrogated and the number of counting loops necessary for the given paper speed is determined.

Calibration

The programme contains a loop, during which the analogue to digital converter operates continuously. The converted values are transferred to the computer and displayed in binary form on the operator console. The Level

Recorder pen (Potentiometer-Range "Stand by") is placed manually at the 0 and maximum positions respectively and the Analogue-Voltage Read-out (ZR 0021) is adjusted to the appropriate digital values. The calibration of the Analyzer, Level Recorder and microphone was done in the usual way.

Data Transfer and further Processing

The data transfer is described in the section "Principal operating functions". In order to store the 80 values for each of the 31 measurement intervals in each measuring position, $31 \times 80 = 2480$ words are necessary, i.e. more than half the memory. The subsequent calculation of the time average L_i of the sound pressure level with the help of the "FLOATING POINT SYSTEM" would require about 10 minutes, since the calculation of one component $10^{0.1\,L_{ij}}$ requires 250 ms. Fig. 5 shows that this value cannot be computed within the waiting time T_w.

It was necessary to find a way of reducing the storage requirement and the computation time:

From equation (1) the time average L_i is computed as

$$L_i = 10 \log \frac{1}{80} \times S_m \quad \text{(dB)} \tag{3}$$

$$\text{where } S_m = \sum_{j=1}^{80} 10^{0.1\,L_{ij}}$$

The separate values L_{ij} can be written

$$L_{ij} = L'_{ij} + k \times 10 \quad \text{(dB)} \tag{4}$$

where $0 \leqq L'_{ij} < 10$ dB and $k = 0, 1, 2, \ldots.$

Hence $10^{0.1\,L_{ij}} = 10^{0.1\,L_{ij}} \times 10^k$.

When using the 50 dB potentiometer the following proportionality is valid for the analogue to digital converter:

50.00 dB	1777_8	1024_{10}
49.95 dB	1776_8	1023_{10}
⋮	⋮	⋮
10.00 dB	315_8	205_{10}
⋮	⋮	⋮
0.05 dB	1_8	1_{10}
0.00 dB	0_8	0_{10}

205_{10} (315_8) values are therefore possible for $10^{0.1\,L_{ij}}$ for the range $0 \leqq L_{ij} \leqq 10$ dB.

These 315_8 values were read-in with the programme and stored in the computer as a "Table".

The Level L_{ij} from the analogue to digital converter is reduced to the range $0 \ldots 315_8$ to obtain it in the form of equation (4).
This gives the number of decades k and a digital value proportional to the level L'_{ij}, which is used as a memory address of the "Tabulated Values". S_m can now be built without the time-consuming computation of $10^{0.1\,L_{ij}}$, by essentially summing the appropriate "Tabulated Values".
All values of the same decade are added to a partial sum, and these sums are stored in 5 temporary memory locations. After the 80 conversions are finished, the partial sums with their appropriate weights are multiplied and the final sum is built.

Temporary storages	'Decade'	Range	Weight*)
0	0	$0 \leq L_{ij} < 10$ dB	10^{-2}
1	1	$10 \leq L_{ij} < 20$ dB	10^{-1}
2	2	$20 \leq L_{ij} < 30$ dB	10^{0}
3	3	$30 \leq L_{ij} < 40$ dB	10^{1}
4	4	$40 \leq L_{ij} < 50$ dB	10^{2}

The following modified formula is used for computing S_m:

$$S_m = 10^{-2} \sum_{j=1}^{m_0} 10^{0.1L'_{ij}} + 10^{-1} \sum_{j=1}^{m_1} 10^{0.1L'_{ij}} + 10^{0} \sum_{j=1}^{m_2} 10^{0.1L'_{ij}} + 10^{1} \sum_{j=1}^{m_3} 10^{0.1\,L'_{ij}}$$

$$+ 10^{2} \sum_{j=1}^{m_4} 10^{0.1\,L'_{ij}}$$

where $m_0 + m_1 + m_2 + m_3 + m_4 = 80$
and $m_0 =$ No. L_{ij} in decade 0
$m_4 =$ No. L_{ij} in decade 4

The temporary storage must be for each a double word, since the largest positive number which can be put into a 12-bit word is 2047_{10}. Using a double word this number is $2^{23} = 8{,}388{,}608$.
The waiting time T_w between the individual sample conversions is enough to calculate the current sum of $10^{0.1\,L_{ij}}$. To minimize the computing time for the weighting, simplified multiplication and division routines were used. The integer multiplication with 10 can be accomplished by addition and rotation of the accumulator. A single rotation to the left gives a multiplication with 2 and a single rotaton to the right a divsion by 2. This can be written:

$$10\,x = \{[(x \times 2) \times 2] + x\} \times 2$$

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*) The values of $10^{0.1\,L_{ij}}$ of the Table are multiplied by 100 and stored as integer values. The weights are therefore 10^{-2} smaller here.

Division by 10 is done by an approximation, again through addition and rotation.

$$\frac{x}{10} \cong \frac{2^5 x + 2^4 x + 2x + x}{2^9} = \frac{51}{512} x$$

The error is 0.017 dB.

The weighting and building of the final sum proceed in the endtime T_e. The final sum is stored in 2 words, so that for 31 measuring intervals 62 memory words are required. After the final measuring interval of a particular measurement position the time average L_i after equation (3) has still to be computed, and this requires 20 seconds.
It is worthy of note that the computing time has been reduced from about 10 minutes for each measurement position, to 20 seconds, and that the temporary storage requirements are reduced from 2480 to 62 words. The programme modifications (including storage of the 205 values of $10^{0.1\,L_{ij}}$) require a total of 450 words.

Spatial Average
The spatial average L_M is computed after equation (2).

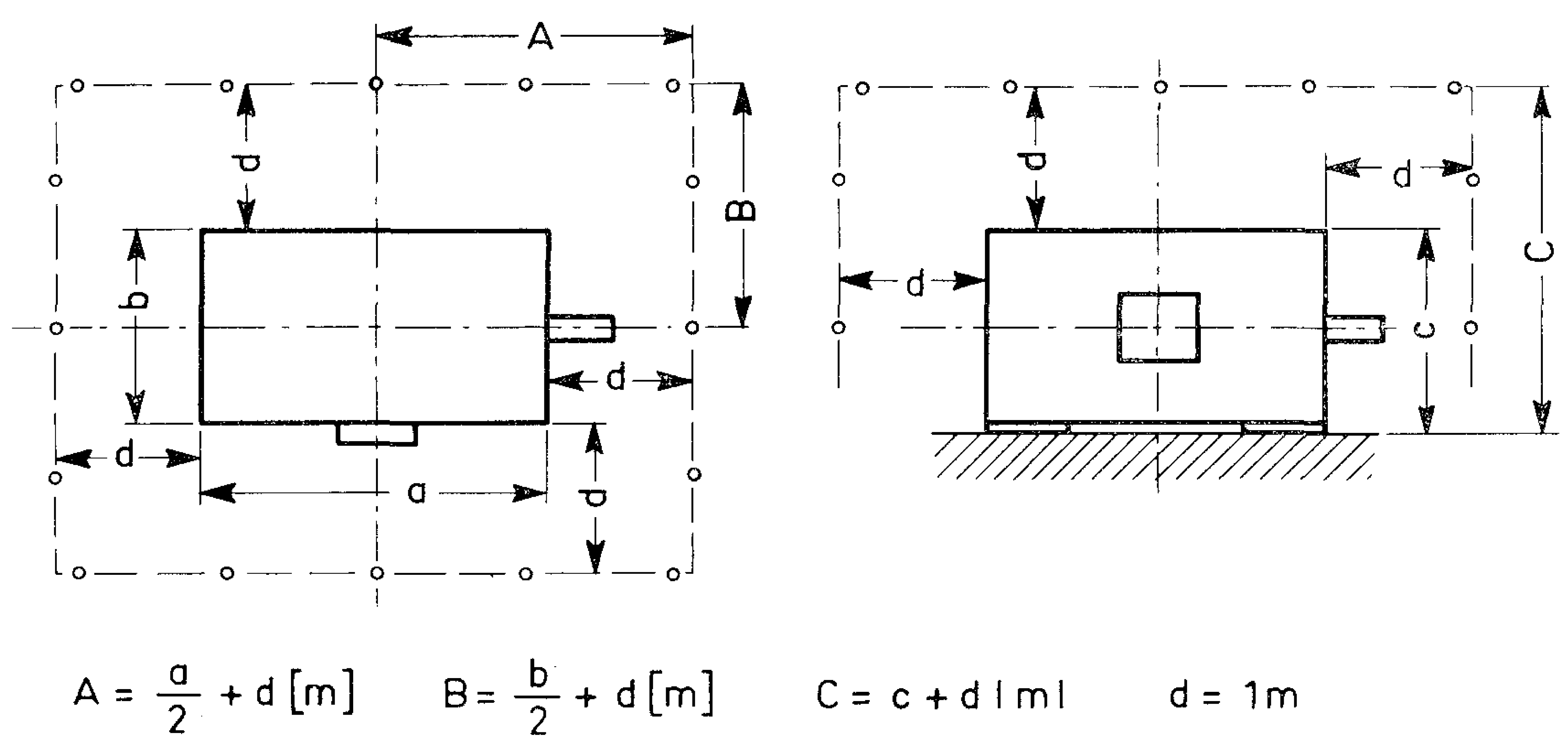

Fig. 7. Position of Measurement points.

a, b, c	*Machine Dimensions.*
A, B, C	*Dimensions of Measurement Surface.*
– – o – –	*Measuring paths and microphone positions.*

S, r, ΔL
The measuring surface (1) can be defined from the machine dimensions (Fig. 7). The equivalent hemispherical surface S is computed from

$$S = 2\pi r^2 = \pi A (B + C) \qquad (m^2)$$

and the equivalent radius r from

$$r = \sqrt{\tfrac{1}{2} A (B + C)} \quad \text{(m)}$$

With the Level difference $\Delta L = 20 \log \dfrac{r}{r_o}$, (dB)

where $r_o = 3$ m (reference radius), the referred sound pressure level L_{Mo} can be computed.

Print Out

The 1/3 octave and octave results are printed out separately. (See Fig. 8). The heading contains the machine-type, manufacture-, sheet-, and investgation numbers, and next the calculated values of A, B, and C (i.e. quantities for determination of the measuring surface), and the equivalent hemispherical radius and surface, r and S respectively. The limitation of space required the display of the 1/3 octave results to be compressed, and in the Table heading only the octave band centre frequency is given. The sound pressure level in the 1/3 octaves belonging to that octave band are printed beneath one another, e.g.:

	63 → octave band centre frequency
Microphone position:	54.6 → sound pressure level in 50 Hz 1/3 oct. band
0.1	59.9 → sound pressure level in 63 Hz 1/3 oct. band
	59.7 → sound pressure level in 80 Hz 1/3 oct. band

The referred sound pressure level L_{Mo} and sound power level L_p are calculated out directly from:

$$L_{Mo} = L_M + \Delta L \quad \text{(dB)}$$

and $\quad L_p = L_M + \Delta L + 17.5 \quad$ (dB) respectively

where $\quad 17.5 = 10 \times \log 2 \pi r_o^2$

Computation of Octave-Levels

The sound pressure level in octave bands is computed according to the following relation:

$$L_{oct} = 10 \log (10^{0.1\, L_1} + 10^{0.1\, L_2} + 10^{0.1\, L_3}) \quad \text{(dB)}$$

where L_1, L_2 and L_3 are the respective 1/3 octave sound pressure levels.

Spectrum

This print-out should give a general view of the noise content of the measured object. The averaged sound pressure levels of the octave bands are displayed in the form of a step-wise diagram with the centre frequency as abscissa and the sound pressure level as ordinate. The accuracy is ± 1 dB and is defined by the outline of the hatched area, as in Fig. 9. An improved display using digital to analogue conversion and a Level Recorder as a plotter is projected for the future.

Various explanations, etc., complete the printout sheet on the Teletypewriter.

MASCH. TYP QU 355 M6A   FABR.NR A 8Ø298

PROTOKOLL-NR 2Ø9   VERSUCH-NR 13   DATUM 26.6.69

A= +Ø.1589999E+Ø1
B= +Ø.1349999E+Ø1
C= +Ø.17Ø9999E+Ø1
R= +Ø.1559711E+Ø1
S= +Ø.15285Ø9E+Ø2

	63	125	25Ø	5ØØ	1K	2K	4K	8K	16K	DB A	DB B	DB C	LIN
	54.6	6Ø.8	66.9	69.1	66.4	62.1	53.4	45.2	4Ø.Ø				
Ø1	58.9	63.2	77.3	71.6	66.4	58.4	48.5	42.6	4Ø.Ø	78.3	82.3	83.2	83.Ø
	59.7	63.1	79.Ø	73.9	64.8	54.3	47.1	4Ø.1	4Ø.Ø				
	6Ø.5	65.1	69.3	74.4	7Ø.5	62.5	52.4	45.3	4Ø.Ø				
Ø2	65.8	65.2	8Ø.6	78.3	68.8	57.7	48.6	43.9	4Ø.Ø	81.9	86.Ø	87.Ø	87.Ø
	63.8	67.9	82.8	78.7	66.6	54.6	47.6	41.8	4Ø.Ø				
	56.8	63.9	67.4	73.6	69.1	6Ø.1	53.8	49.6	44.Ø				
11	67.Ø	63.4	81.7	75.7	68.8	58.8	51.8	48.Ø	42.Ø	8Ø.9	85.8	87.2	87.Ø
	61.4	64.8	83.8	72.4	65.6	55.7	51.Ø	46.1	4Ø.Ø				
	55.3	59.9	66.5	72.4	67.1	61.7	56.6	51.2	43.2				
12	57.9	61.1	76.6	71.6	67.5	58.9	54.5	48.7	4Ø.Ø	77.5	81.2	83.Ø	82.5
	59.3	63.1	77.4	71.Ø	62.4	57.1	53.3	46.Ø	4Ø.Ø				
	6Ø.5	65.4	7Ø.6	76.2	71.6	65.1	58.6	53.3	47.2				
LM	67.7	66.5	82.1	78.5	71.4	62.2	56.2	51.5	45.3	83.1	87.2	88.3	88.3
	64.4	67.9	84.3	77.7	68.2	59.7	55.Ø	49.4	43.7				
	54.8	59.7	64.9	7Ø.5	65.9	59.4	52.9	47.6	41.5				
LMØ	62.Ø	6Ø.8	76.4	72.8	65.7	56.5	5Ø.5	45.8	39.6	77.4	81.5	82.6	82.6
	58.7	62.2	78.6	72.Ø	62.5	54.Ø	49.3	43.7	38.Ø				
	72.3	77.2	82.4	88.Ø	83.4	76.9	7Ø.4	65.1	59.Ø				
LP	79.5	78.3	93.9	9Ø.3	83.2	74.Ø	68.Ø	63.3	57.1	94.9	99.Ø	1ØØ.1	1ØØ.1
	76.2	79.7	96.1	89.5	8Ø.Ø	71.5	66.8	61.2	55.5				

Fig. 8. Example of Print-out: Results in 1/3 octave bands.

17000 1

```
            63   125  250  500   1K   2K   4K   8K  16K  DB A

9Ø.Ø                                                             9Ø.Ø
88.Ø                                                             88.Ø
86.Ø              -----                                          86.Ø
84.Ø              -----                              -----       84.Ø
82.Ø              ----------                         -----       82.Ø
8Ø.Ø              ----------                         -----       8Ø.Ø
78.Ø              ----------                         -----       78.Ø
76.Ø              ---------------                    -----       76.Ø
74.Ø              ---------------                    -----       74.Ø
72.Ø         --------------------                    -----       72.Ø
7Ø.Ø    -------------------------                    -----       7Ø.Ø
68.Ø    ------------------------------               -----       68.Ø
66.Ø    ------------------------------               -----       66.Ø
64.Ø    ------------------------------               -----       64.Ø
62.Ø    -----------------------------------          -----       62.Ø
6Ø.Ø    -----------------------------------          -----       6Ø.Ø
58.Ø    -----------------------------------          -----       58.Ø
56.Ø    ----------------------------------------     -----       56.Ø
54.Ø    ----------------------------------------     -----       54.Ø
52.Ø    ----------------------------------------     -----       52.Ø
5Ø.Ø    --------------------------------------------------       5Ø.Ø
48.Ø    --------------------------------------------------       48.Ø
46.Ø    --------------------------------------------------       46.Ø
44.Ø    --------------------------------------------------       44.Ø
42.Ø    --------------------------------------------------       42.Ø
4Ø.Ø    --------------------------------------------------       4Ø.Ø
```

E R L A E U T E R U N G E N Z U D E N B E I L A G E N:

A:B:C	SIEHE DECKBLATT (SKIZZE)	
$R = \sqrt{Ø\cdot 5\ A(B+C)}$	RADIUS DER AEQUIVALENTEN HALBKUGELFLAECHE	(m)
$S = 2\pi R^2 = A(B+C)$	AEQUIVALENTE HALBKUGELFLAECHE	(m^2)
63,125,.........,16K	OKTAVBANDMITTENFREQUENZEN	(HZ)
Ø1,Ø2,.........,N	MESSPUNKTE	
LM	MITTLERE SCHALLDRUCKPEGEL *)	(DB,DB(A))
LMØ	MITTLERE SCHALLDRUCKPEGEL BEZOGEN AUF REFERENZ-RADIUS 3 METER *)	(DB,DB(A))
LP	MITTLERE SCHALLEISTUNGSPEGEL	(DB)

*) UMGERECHNET (3 DB) / GELTEN FUER ABSTRAHLUNG IN EINEN HALBRAUM

B E M E R K U N G E N :...
....
....
....

DIE GERAEUSCHSTAERKE IST ZULAESSIG / NICHT ZULAESSIG NACH VDE Ø53Ø
TEIL 1, PARAGRAPH 54.

BIRR,MF-VL,DEN............ GEMESSEN:........... GESEHEN:..........

Fig. 9. Example of Print-out: Mean Octave band spectrum.

17ooo2

Statistics

If required a tape can be punched, which contains information in 1/3 octave bands, and which facilitates further statistical analysis.

Future Developments

In the measurement of noise and vibration, future developments are almost certain to make greater use of automatic data collection [3].

The objective is seen as using the computer as an integrated part of the total system, with due regard paid, of course, to the cost. It is believed that the system described here is a step in this direction, at a relatively low outlay.

Literature Cited

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[2] B. Ploner: Ein neues Akustiklabor für Zweckforschung. Brown Bovery Mitt. Bd. 53 (1966), Nr. 6/7 Seite 428 . . . 431.

[3] J. Søeberg: The use of Digital Systems in Acoustical Measurements. Brüel & Kjær Technical Review No. 1-1969.

[4] Digital Equipment Corporation: Small Computer Handbook. 1967 Edition.

1/3 Octave Spectrum Readout of Impulse Measurements*)

by

Bent Frederiksen

ABSTRACT

Problems concerning the choice of time constants are discussed. Also a measuring technique using the Real-time 1/3 Octave Analyser Type 3347 for the measurements of unrepeatable impulses is outlined. The measuring principle is based upon the following: A 1/3 octave filter responds to an impulse step function with a rounded tone burst pulse. The energy of this tone burst is equal to the energy within the filter bandpass of the original impulse. The time function of the signal is squared and integrated in the running exponential analog detector, and the highest RMS value for each 1/3 octave band is measured. Calculated and measured values are shown for tone bursts of sine, square waves and "N"-curves (sonic booms).

SOMMAIRE

Les problèmes concernant le choix d'une constante de temps sont discutés. Une technique de mesure utilisant l'analyseur 1/3 d'octave en temps réel est soulignée dans le cas d'impulsions uniques. Le principe est le suivant: La réponse, à une fonction impulsion, d'un filtre 1/3 d'octave, est une impulsion dodulée et amortie. L'energie de cette dernière est égale à l'energie dans la bande considérée de l'impulsion originale.

La fonction temporelle du signal est elevée au carré, integrée dans le detecteur analogique exponentiel, et la plus grande valeur quadratique moyenne pour chaque 1/3 d'octave est mesurée. Les valeurs théoriques et expérimentales sont données pour des impulsions de forme sinusoidalee, rectangulaire et les courbes en «N» (bangs soniques).

ZUSAMMENFASSUNG

Die mit der Wahl der Zeitkonstanten auftretenden Probleme werden diskutiert. Weiterhin wird ein Meßverfahren für die Messung von sich nicht wiederholenden Impulsen mit Hilfe des Ectzeit-Terzanalysators Typ 3347 aufgezeigt. Die Messung stützt sich auf folgendes Prinzip: Ein Terzfilter liefert bei einer Sprungfunktion im Eingang einen abgerundeten tonfrequenten Schwingimpuls (Tonfrequenzburst) im Ausgang. Die Impulsenergie ist der Energie des ursprünglichen Impulses innerhalb des Filterbandpasses gleich. Die Zeitfunktion des Signals wird quadriert und im Analoggleichrichter mit »atmende« Exponentialkurve integriert. Der höchste Effektivwert für jedes Terzband wird gemessen.

Für Sinus-, Rechteck- und "N"-Impulse (Überschallknall) werden gemessene und berechnete Werte angegeben.

Introduction

In our serial type frequency analysis equipment, the read-out is provided by the instrument using its "fast" or "slow" characteristics, or by the Level Recorder.

Thus the meter with its detector or the writing speed of the graphic level recorder provides the averaging of rapidly fluctuating signal levels throughout the whole spectrum.

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*) Based on a paper presented at the 78th meeting of the Acoustical Society of America. San Diego, November 1969.

B & K Real-Time 1/3 Octave Analyzer Type 3347 is a true parallel analyzer where only the time constant of the integrator provides the averaging element, and where the averaging time can be chosen for each channel independently. Updating of RMS analog signals occur on all channels continuously.
Fig. 1 shows the sequence of detector, integrator, store gate, store capacitor and scan gate for each of the channels. The common buffer amplifier is also shown.

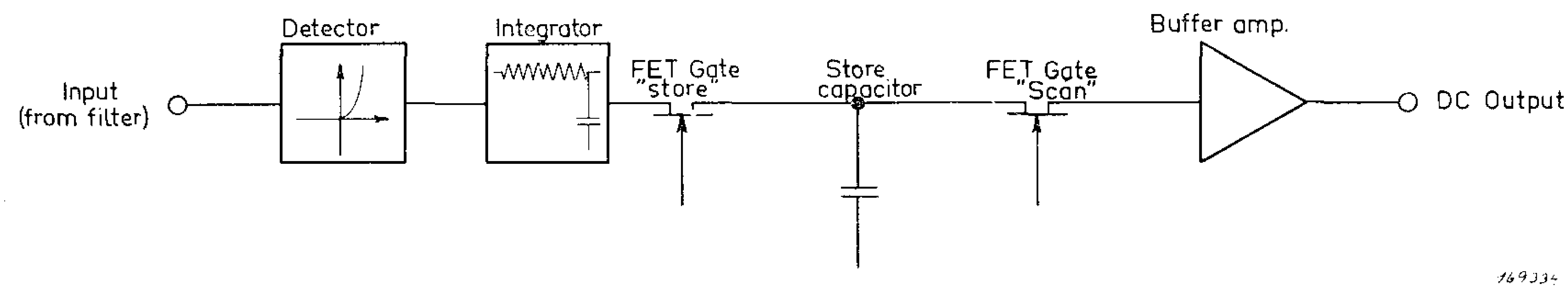


Fig. 1. Block diagram of analog signal processing.

Time Constant

Anyone of three different time constant modes may be selected to suit sinusoidal, fast random, or slow random signals. See Fig. 2.

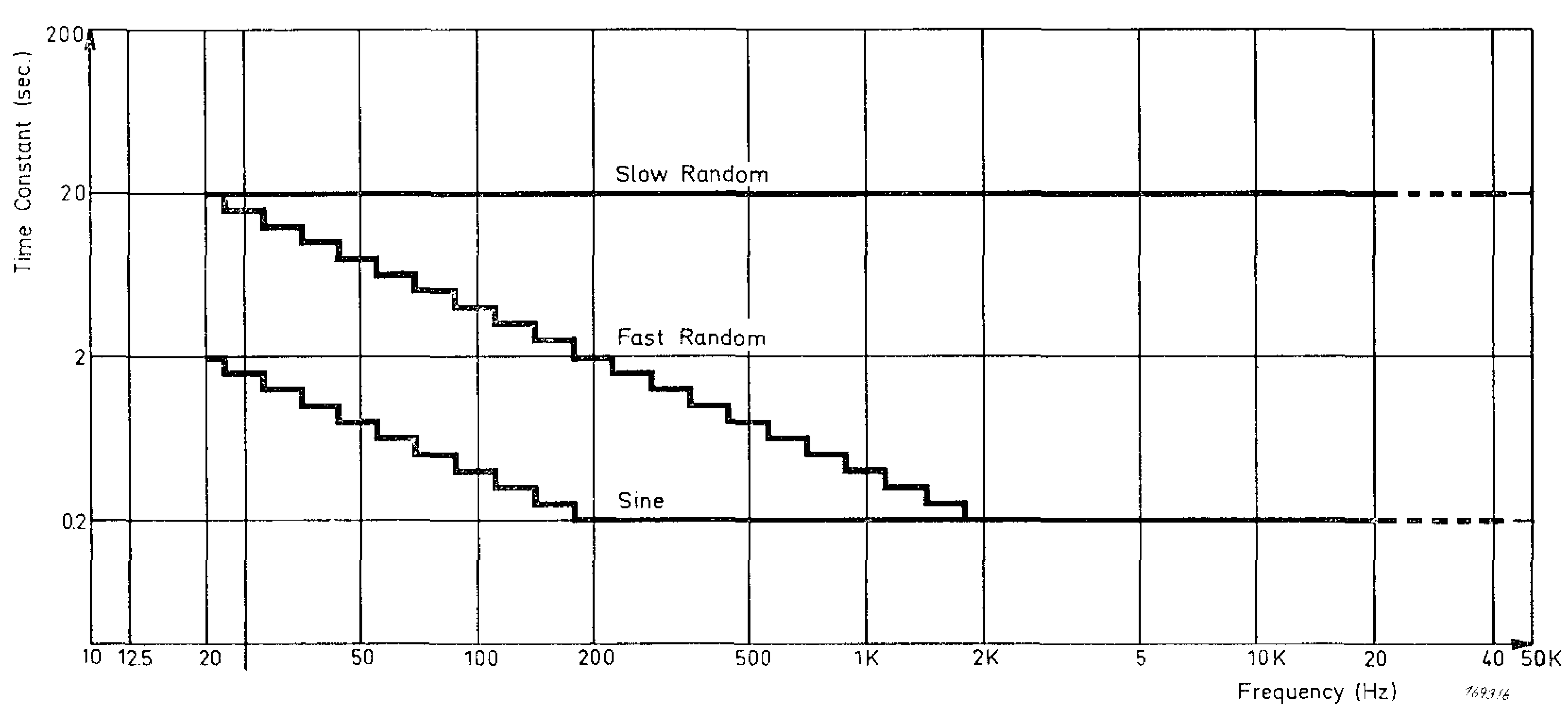


Fig. 2. Variation with frequency of the time constants of the 1/3 octave filters.

The applications for the three standard time constant modes are

"Sine" for continuous spectra,

"Fast Random" for non stationary data where speed of measurement overrides accuracy and a varying time constant is acceptable, to obtain a constant confidence level.

»Slow Random« where a good confidence level is essential.

(Nevertheless any time constant in the range 20 msec. to 20 sec. can be made available for any filter. However if a non standard time constant is needed, considerations should be given to detector ripple and error related to measurement bandwidth, versus the type of signal to be measured).

Ripple from Detectors

As the output from 1/3 octave filter is approximately a symmetrical sine wave, a half wave rectification is used in the filter detectors.

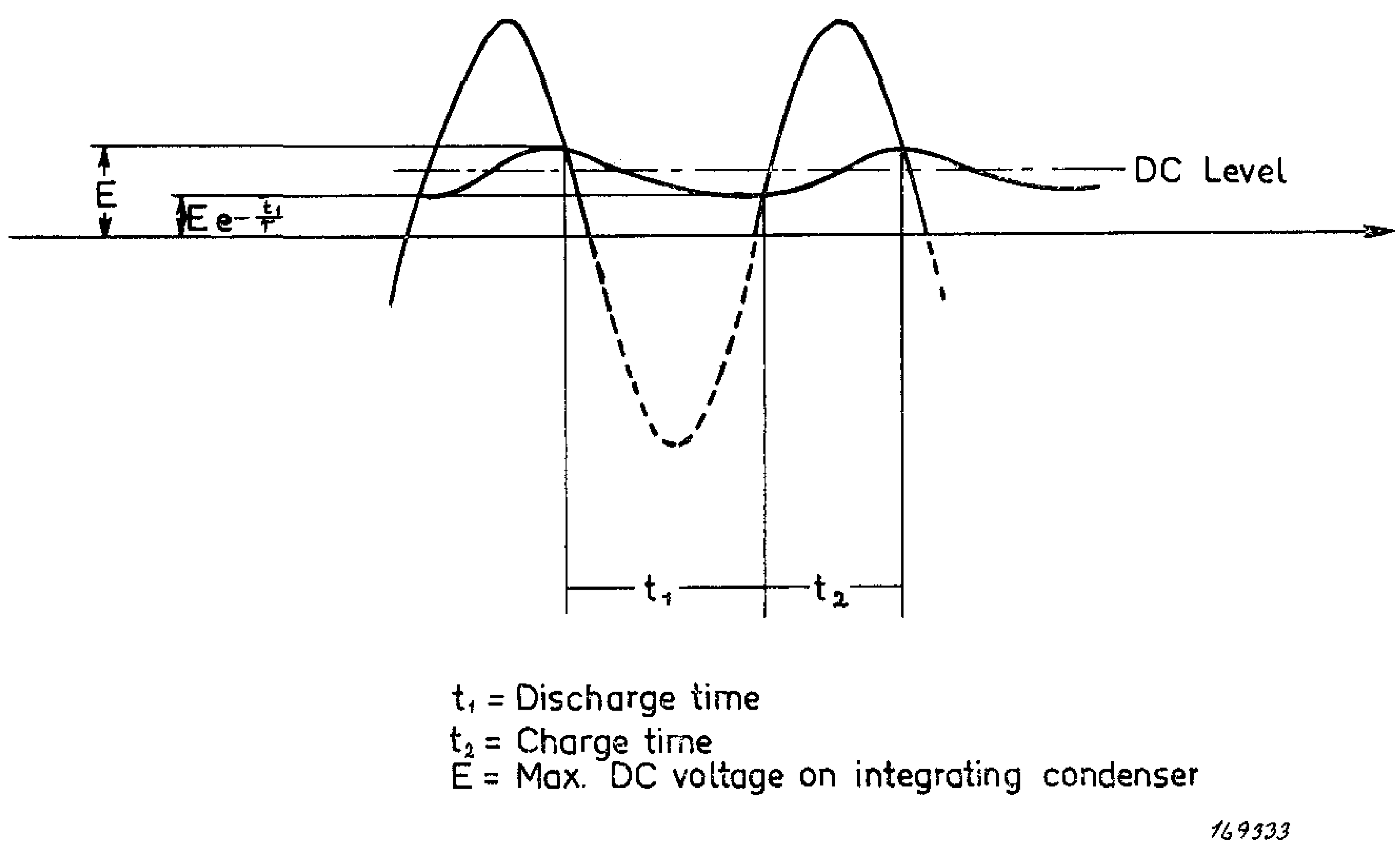


Fig. 3. Half wave rectification of a sine wave signal.

Ripple resulting from sine wave signals and time constants of 200 msec., 500 msec. and 1 sec. is shown in Fig. 4.

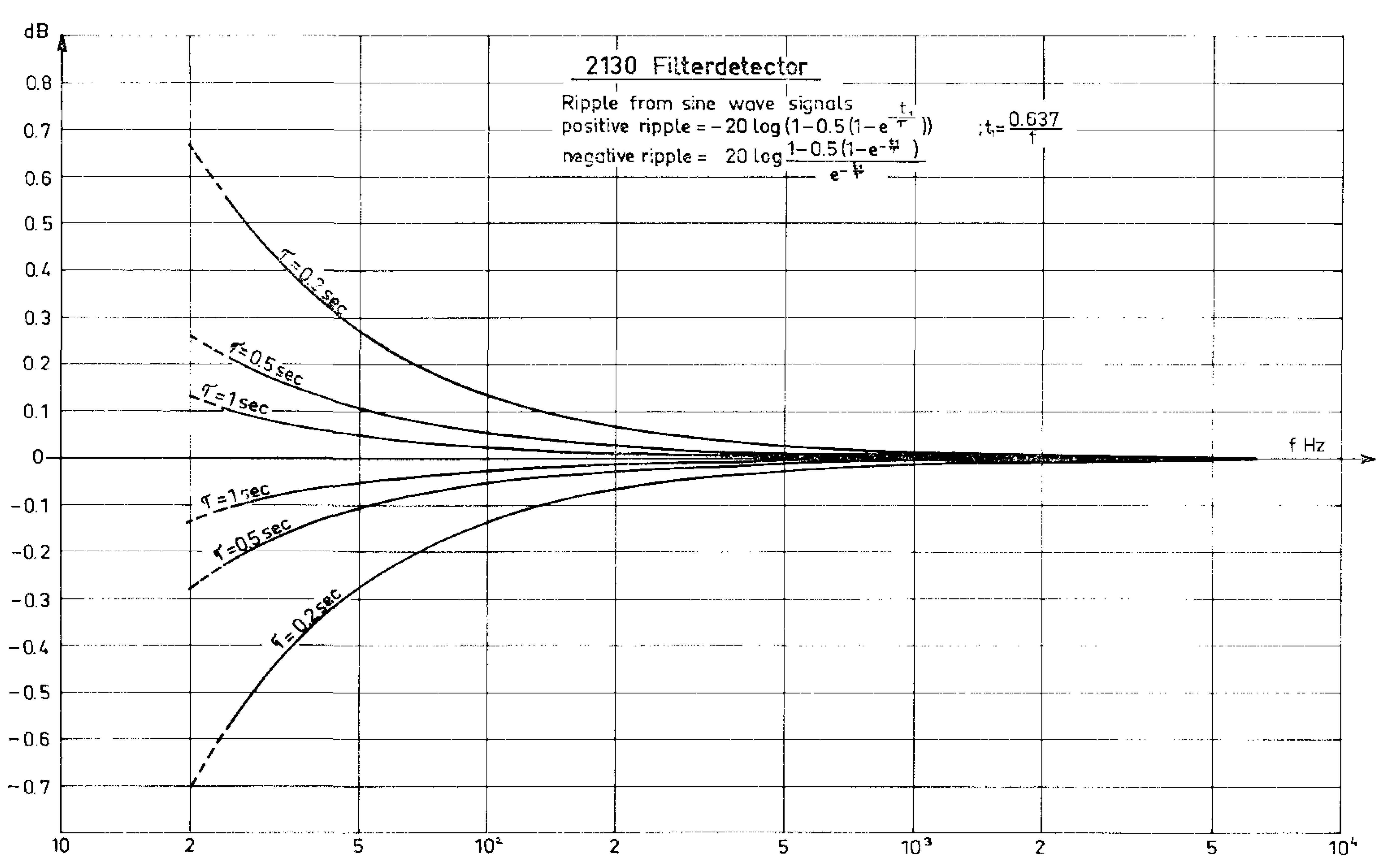


Fig. 4. Ripple from Sine wave signals.

It can be seen that the time constant must be increased with decreasing frequency to avoid large ripple. The low frequency weighting of our "Sine" time constant mode ensures a ripple of less than 0.2 dB.

Influence of Measurement Bandwidth

On random noise measurements the fluctuation in RMS level caused by random amplitude distribution in the narrow band spectrum passed by a 1/3 octave filter is much greater than the ripple discussed above for sine waves.

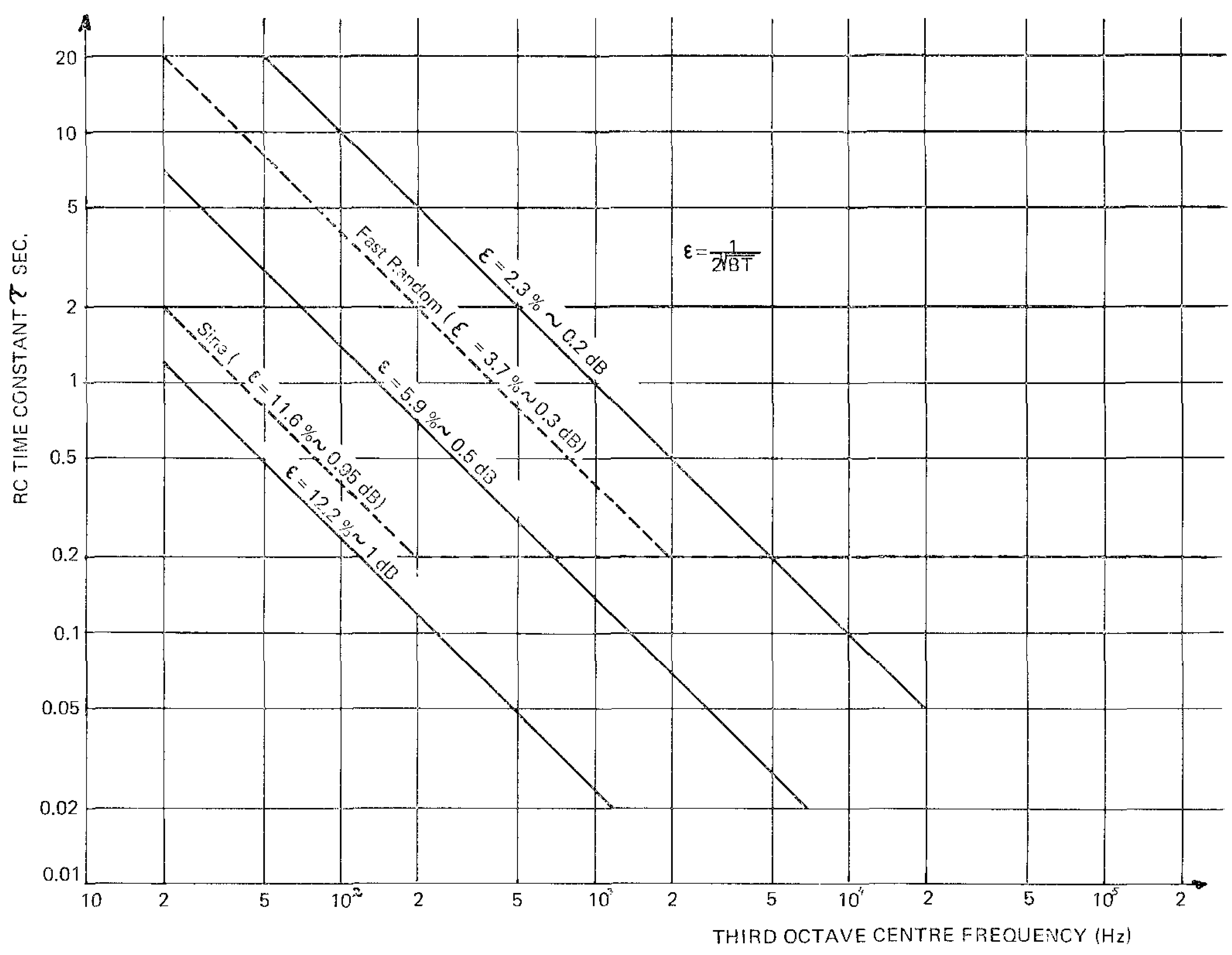


Fig. 5. Standard deviation related to bandwidth and time constant.

In Fig. 5 three full lines are drawn, representing the one δ standard deviation of 1 dB, 0.5 dB and 0.2 dB.

For comparison the standard deviations resulting from the time constant modes "Sine" and "Fast Random" are shown as dotted lines. The graph shows what time constant to choose for each filter in order to obtain a desired confidence level. The need for maintaining high confidence with digital readout is obvious because the computer sees only the value of the detector signal at the instant when it samples the detector.

Impulse Measurements

The conditions of impulse signal measurements will now be discussed:

If a square wave is applied at the input of a filter, the output from the filter will be two rounded tone burst pulses, one from the leading edge, the other from the trailing edge of the square wave impulse (see Fig. 6).

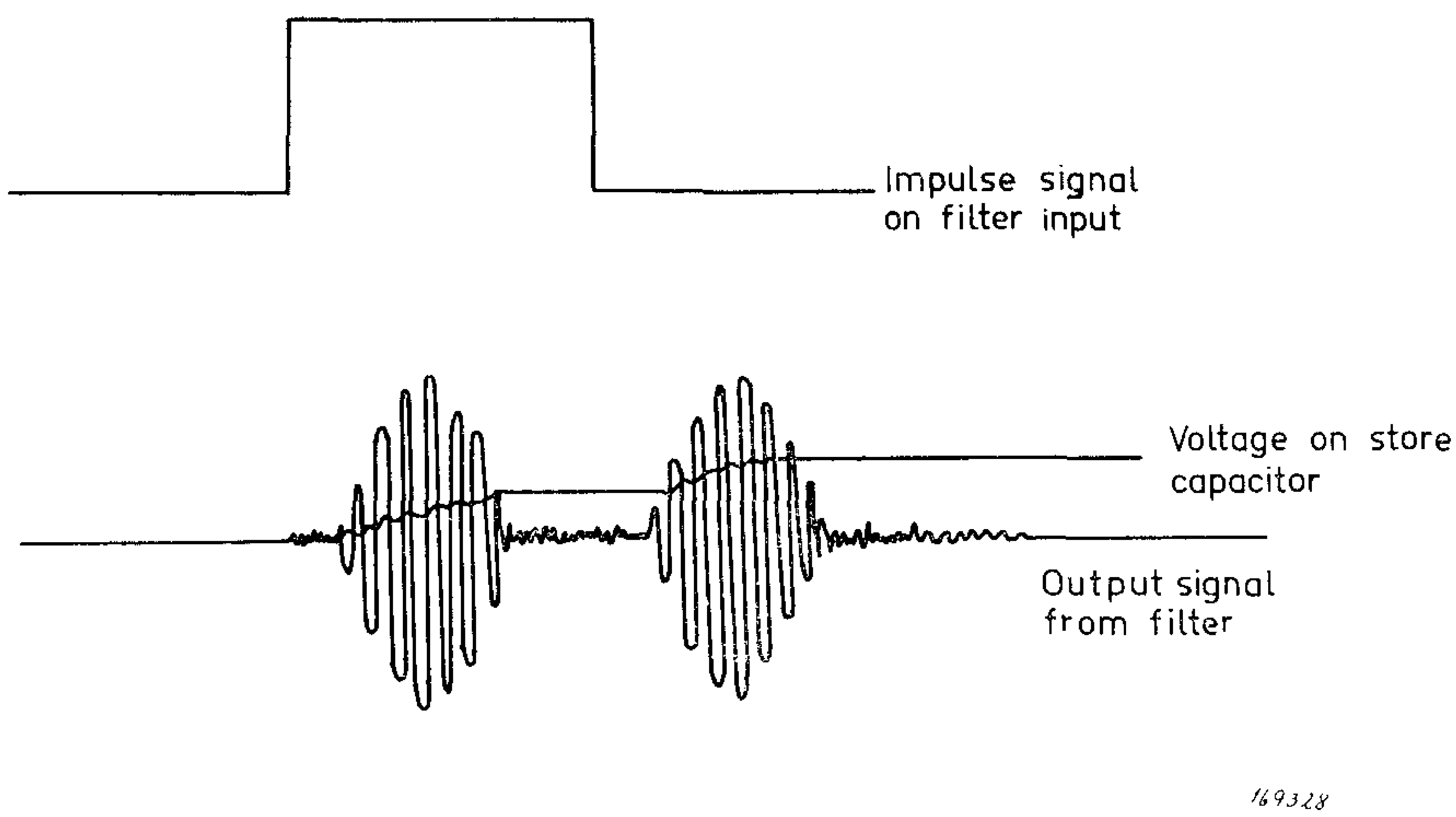


Fig. 6. Filter response to a square wave input.

The time function of an impulse squared and integrated is a measure for the total energy in the impulse.

A connection between time function and frequency function is given by

$$\int_{-\infty}^{\infty} f^2(t)\, dt = \frac{1}{\pi} \int_{-\infty}^{\infty} |F(\omega)|^2\, d\omega$$

where $f(t)$ is the time function of the impulse
and $|F(\omega)| = A(\omega)$ is the amplitude density spectrum
as $f(t)$ is the Fourier transform of $F(\omega)$.

If $y(t)$ is the signal at the output of a filter when an impulse is applied at the filter input, and $H(\omega)$ is the transfer function of the filter, the following equation is found:

$$\int_{-\infty}^{\infty} y^2(t)\, dt = \frac{1}{\pi} \int_{-\infty}^{\infty} A^2(\omega)\, |H(\omega)|^2\, d\omega$$

For ideal 1/3 octave filters, $|H(\omega)|^2$ is equal to 1 within $\omega_0 \pm 1/6$ oct. and equal to 0 outside $\omega_0 \pm 1/6$ oct.

In this case:

$$\int_{-\infty}^{\infty} y^2(t)\, dt = \frac{1}{\pi} \int_{\omega_o - 1/6 \text{ oct.}}^{\omega_o + 1/6 \text{ oct.}} A^2(\omega)\, d\omega$$

This means that the total energy of the signal from each filter will be equal to the energy within the filter band-pass frequencies of the original impulse signal.

When the energy is averaged over a time equal to the time constant of the integrator the signal level within each 1/3 octave filter is found from

$$\mathrm{dB}_{\text{re } 10\,\mu\text{V RMS}} = 20 \log \left(\frac{1}{10^5} \sqrt{\frac{1}{\pi\tau} \int_{\omega_o - 1/6 \text{ oct.}}^{\omega_o + 1/6 \text{ oct.}} A^2(\omega)\, d\omega} \right)$$

where τ = averaging time
and $\omega_o \approx$ filter centre frequency.

It is assumed that

1. The 1/3 octave filters are "ideal"
2. Integration is true linear
3. The complete output signal from the filters is found within the integrating period.

Measurements on such pulses have been made with the B & K Real-Time Analyzer using its "Sine" time constant mode and "Store Max" feature.

"Store Max" means that, in each filter channel, the highest RMS value produced during the measurement is stored on the "store" capacitor.

Fig. 6 shows the output signal from a 1/3 octave filter and the voltage on the store capacitor using the "Store Max" mode of the Analyzer.

Calculations and measurements will be shown for three kinds of impulses: Sinusoidal tone burst, square wave, and "N"-curves (Sonic Booms).

Sinusoidal Tone Burst

The spectrum is given by

$$|F(\omega)| = \frac{2a(-1)^p}{j\omega_o} \frac{1}{(\omega/\omega_o)^2 - 1} \sin p \pi \omega/\omega_o$$

where p is the number of periods.

In Fig. 7, a tone burst of four periods of 1000 Hz is shown together with its amplitude density spectrum. Also the effect of using constant percentage bandwidth filters in the frequency analysis is illustrated.

The calculated and measured spectra are shown in Fig. 8 for a one-period and a four-period tone burst of a 1 kHz sine wave.

To show how closely the measured spectrum resembles the calculated, theoretical spectrum, the deviation between the two were plotted for each channel

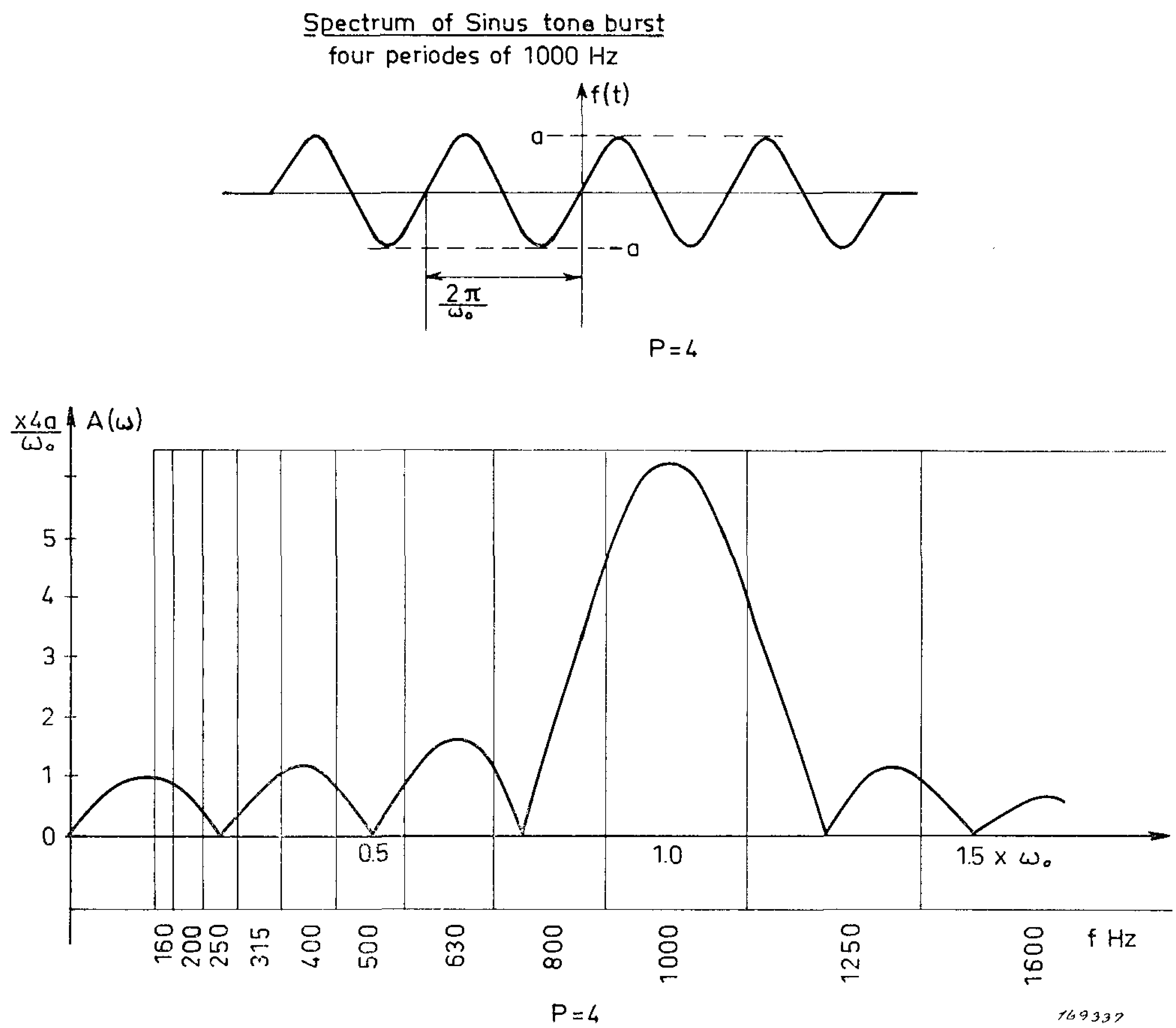


Fig. 7. Amplitude density spectrum of a 1 kHz 4 period tone burst.

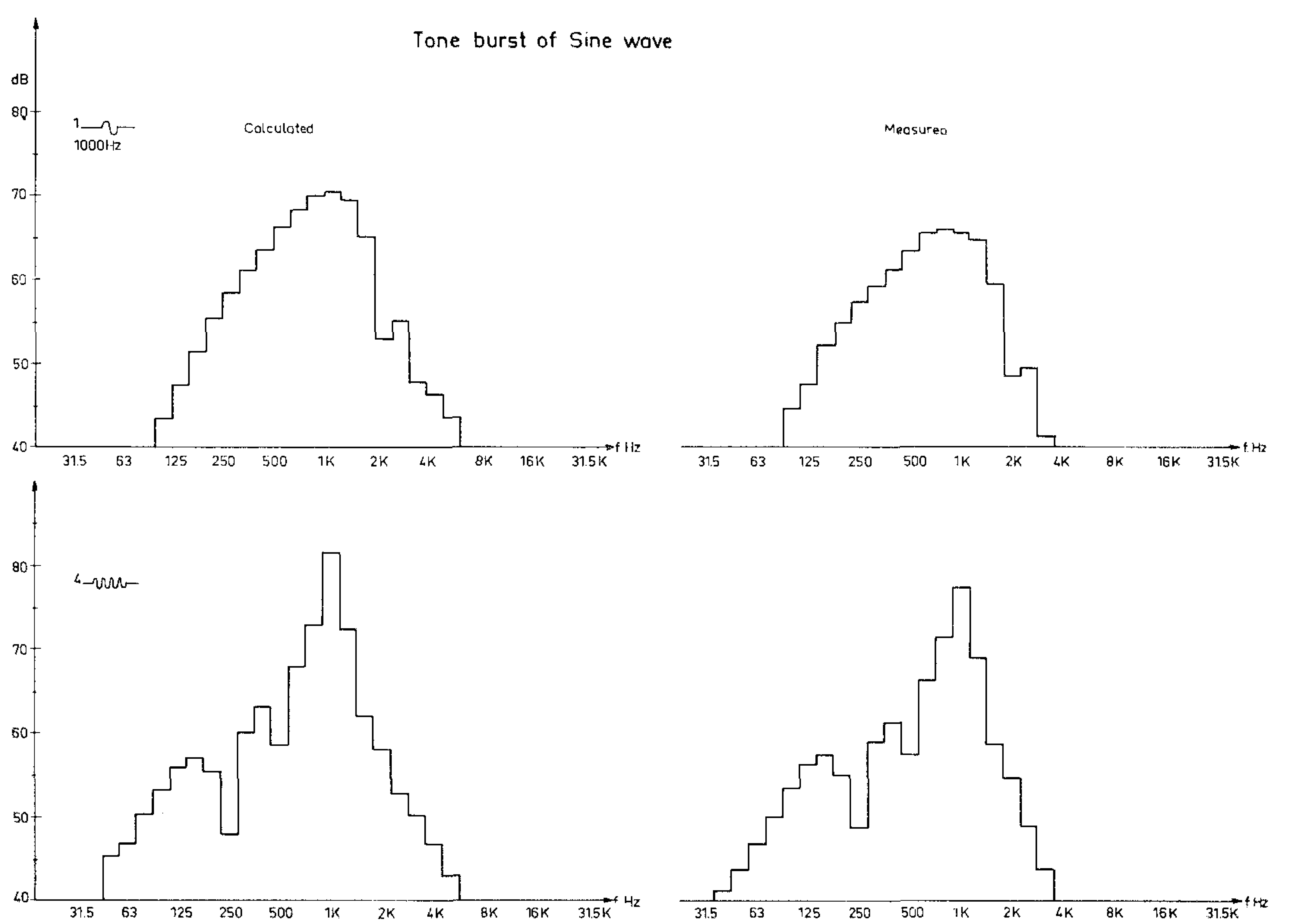


Fig. 8. 1/3 octave spectrum of two sinusoidal tone bursts.

during measuremens of tone burst impulses with 1–2–4–8–16–32–64 and 128 periods of the 1 kHz sine wave.

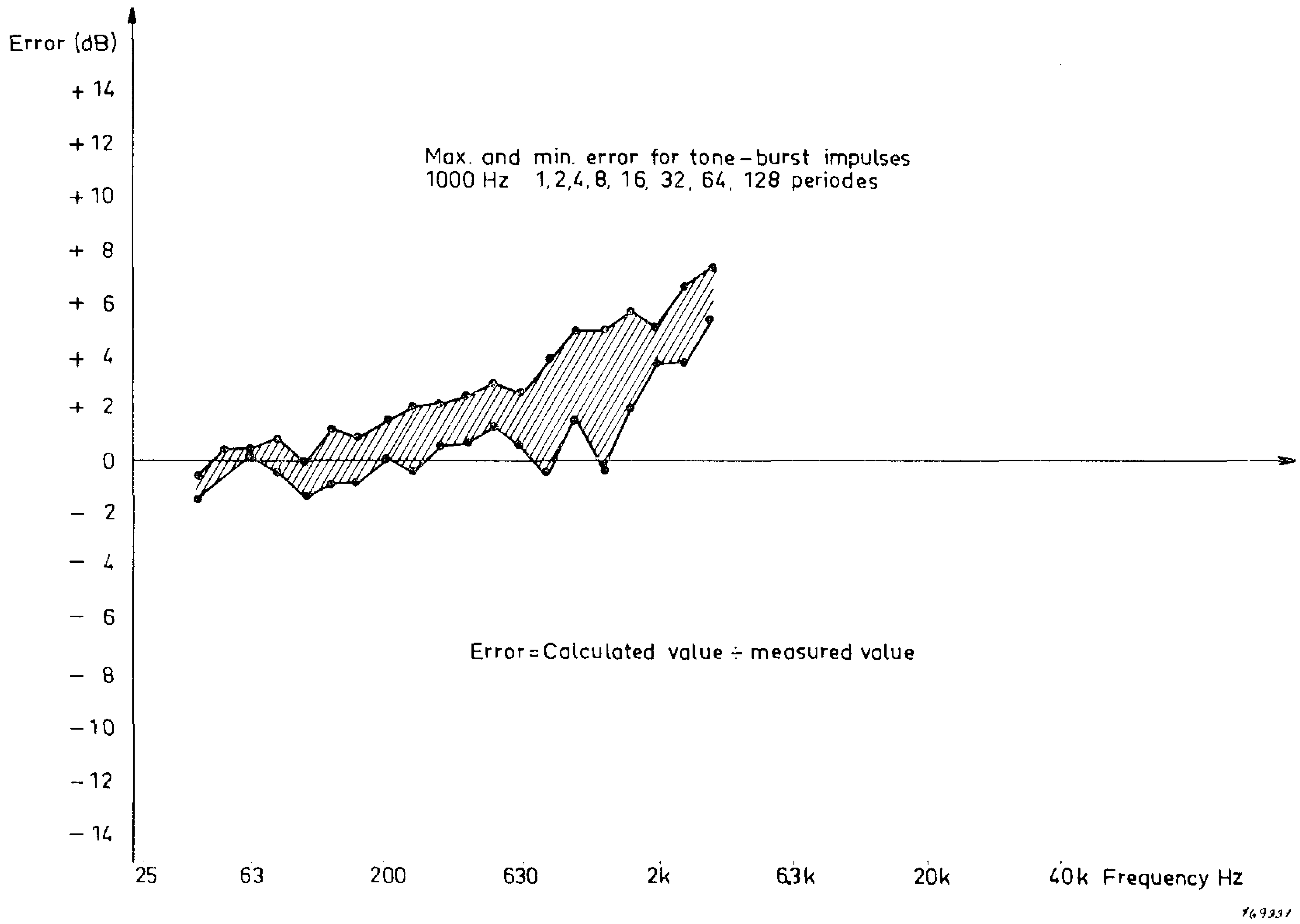


Fig. 9. Error spread. Sinusoidal tone bursts.

Max. positive and negative error are the two curves of Fig. 9, the shaded area representing the error spread of the above mentioned measurements.

Square Wave Impulse

The spectrum is given by

$$F(\omega) = aT \frac{\sin \omega T/2}{\omega T/2} e^{-j\omega T/2}$$

The graph on Fig. 11 shows the theoretical frequency (Fourier) spectrum of a square wave impulse.

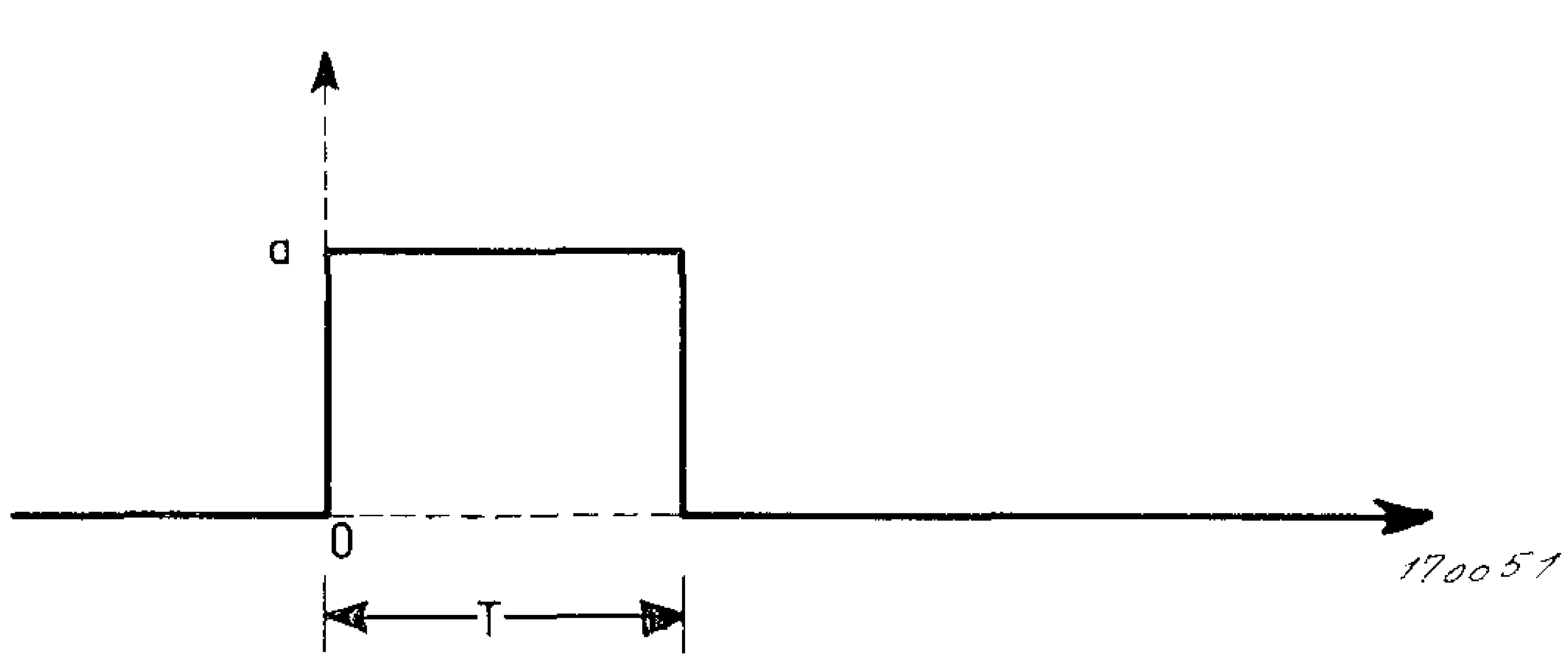


Fig. 10. Square wave impulse.

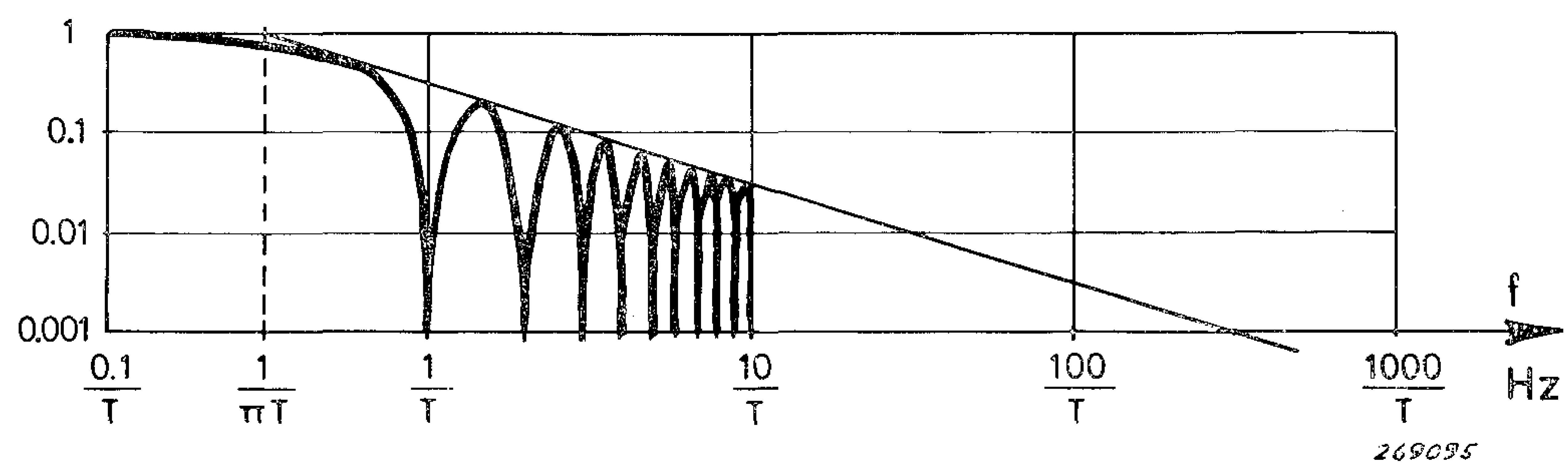


Fig. 11. Theoretical frequency spectrum of a square wave impulse.

Calculated and measured curves for two square wave impulses are shown, a 1 msec. 5 Volt impulse and a 4 msec. 5 Volt impulse (see Fig. 12).
The characteristic high frequency variations in the spectrum is of course lost, because of the wide bandwidth of the high frequency filters.
Nevertheless the error spread for a number of measurements is very small, considering the calculations are made for ideal filters and true linear integration.

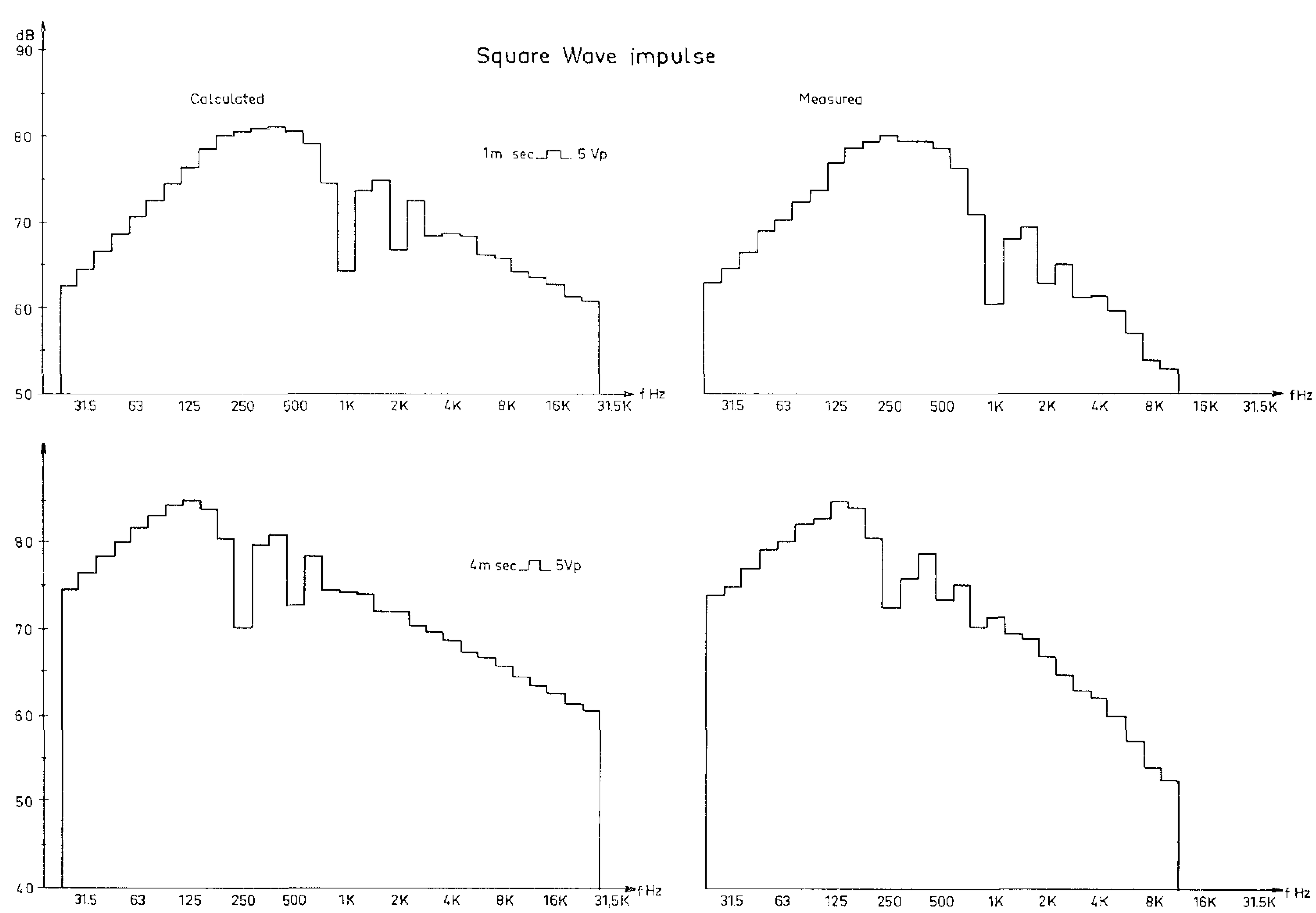


Fig. 12. 1/3 octave spectrum of two square wave impulses.

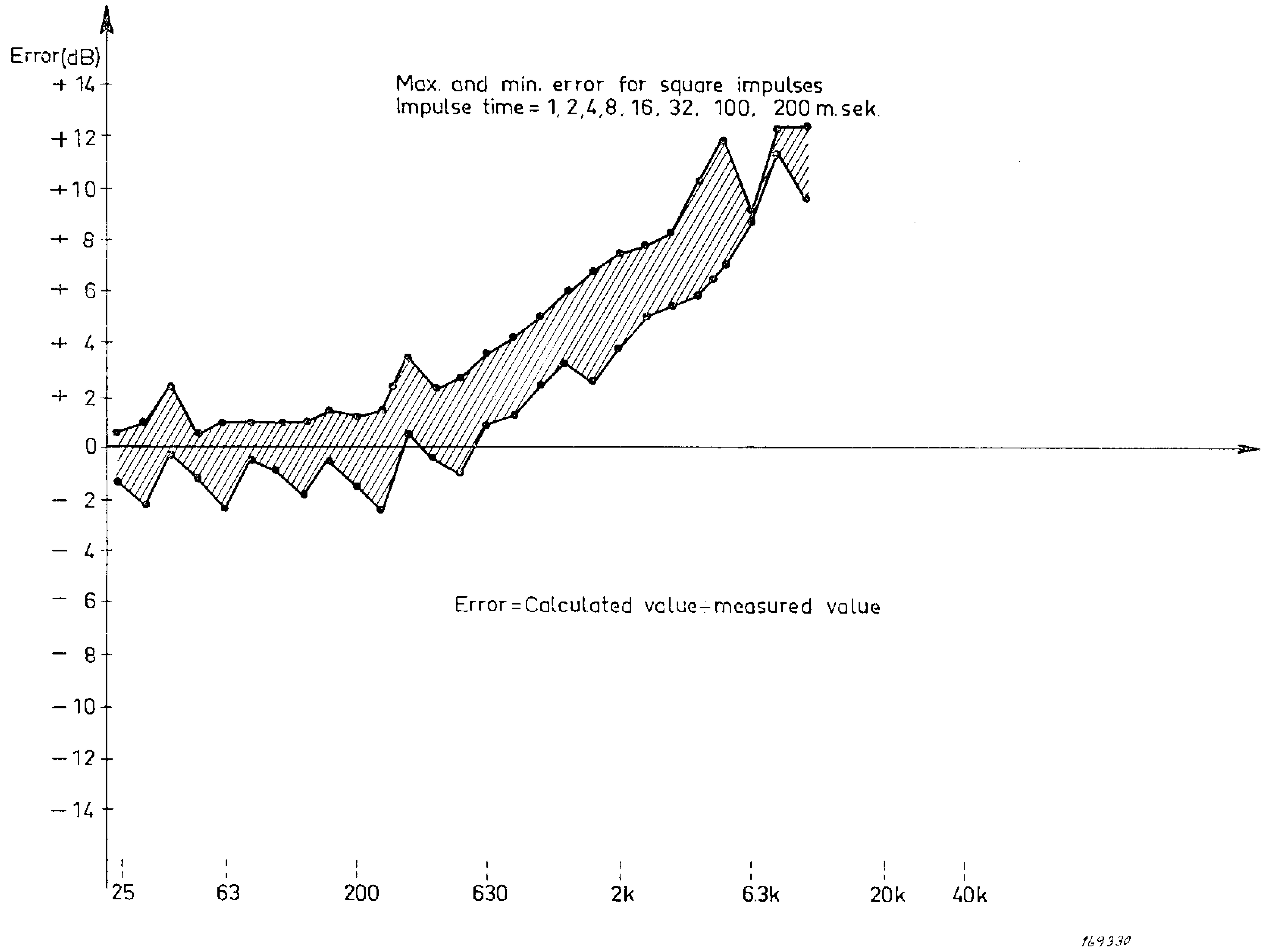


Fig. 13. Error spread. Square wave impulses.

For the highest frequencies rather low values are measured. This can be explained as follows:

In Fig. 6 is shown that the filter response to a step function is a rounded tone burst pulse of about 10 periods of the filter frequency. This tone burst pulse of a 20 kHz filter will last approx. 0.5 msec. This pulse will act as too short a transient for the filterdetector and will cause the parabola to average the signal rather than squaring it.

If the time constant is made short enough for the parabola to follow the high frequency signal, other problems will arise with long impulses, as only part of the energy will be measured.

"N"-curve (Sonic Boom)

The spectrum of a symmetrical "N"-impulse is given by

$$F(\omega) = j\frac{2a}{\tau T}\frac{1}{\omega^2}\left[(T+\tau)\sin\omega T - T\sin\omega(T+\tau)\right]$$

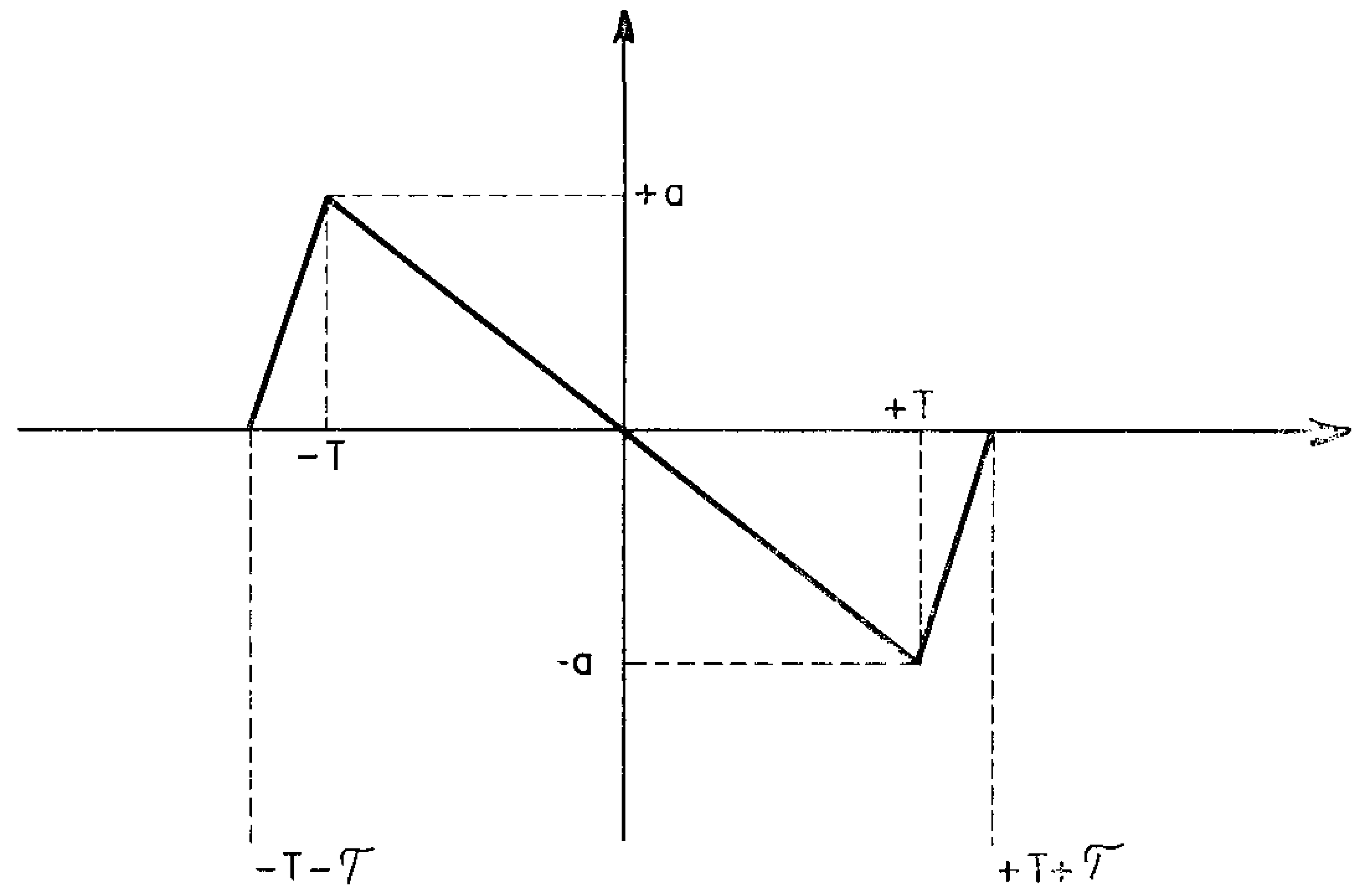


Fig. 14. "N"-impulse. 170050

Calculated and measured values for two "N"-curves (simulated sonic booms) are shown in Fig. 15.

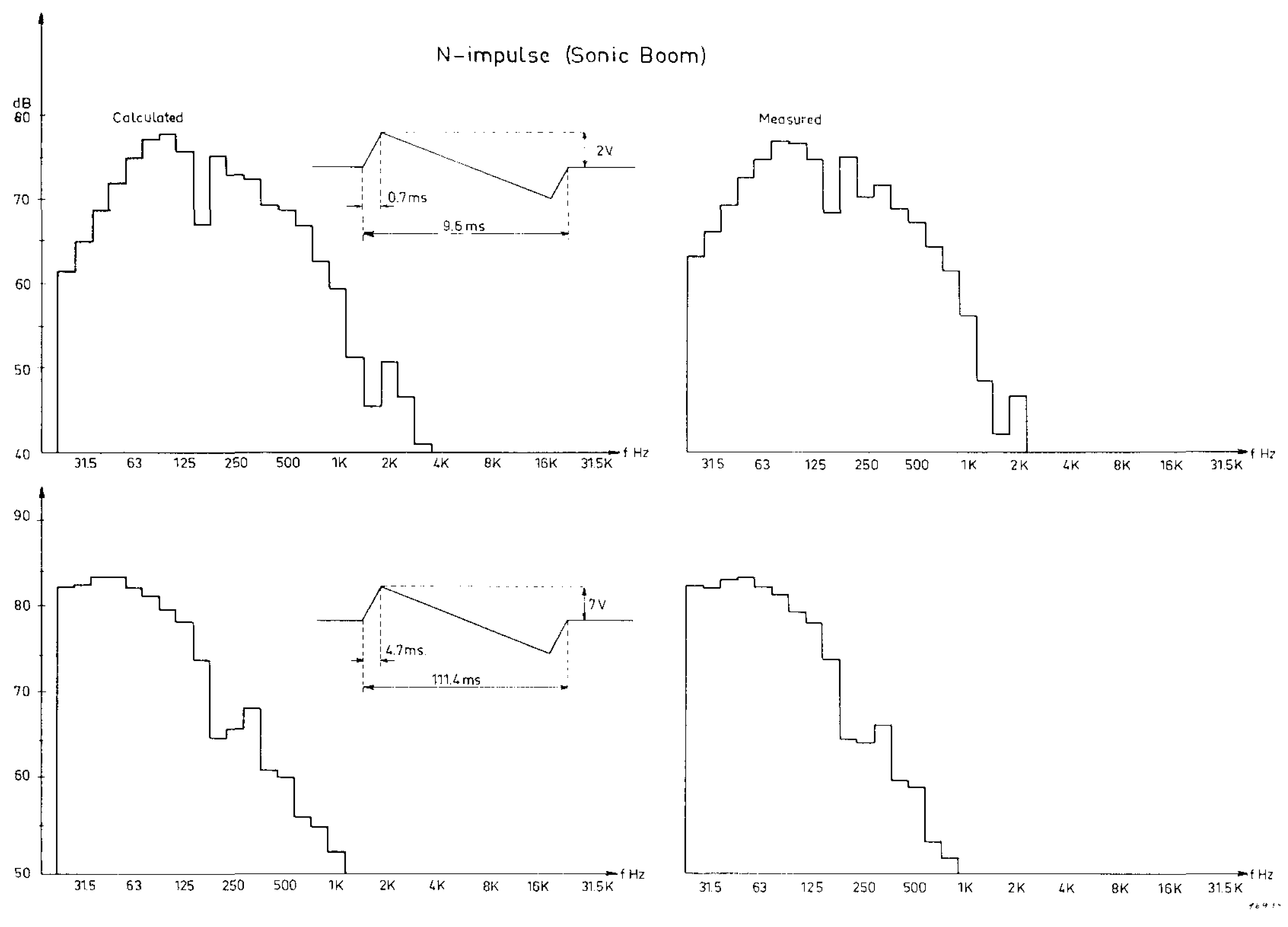


Fig. 15. 1/3 octave spectrum of two "N"-impulses.

The error spread for three measurements is shown in Fig. 16.
Typical repeatability for all the measurements made on sinusoidal tone bursts, square wave impulses, and "N"-impulses is 0.4 dB.

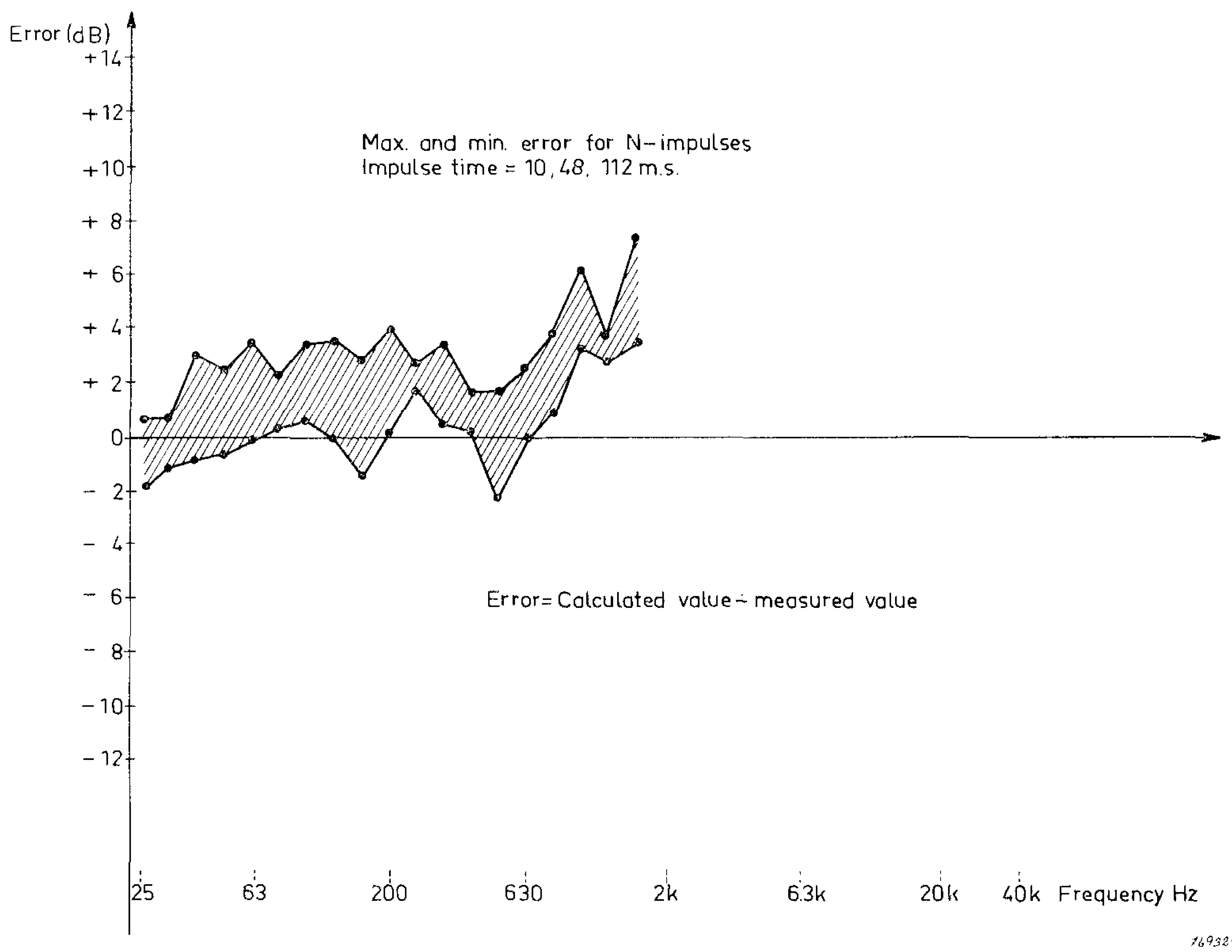


Fig. 16. Error spread. "N"-impulses.

Conclusion

The B & K Real-Time Analyzer has proven to be a very useful instrument for the measurement of steady-state noise or stationary signals, periodic and random signals, and single impulses with a duration from a chosen time-constant down to a duration of only a fraction of a msec.

With all the analog as well as digital features, the B & K Real-Time 1/3 Octave Analyzer conveniently links together current analog measuring procedures and new high speed automatic data processing techniques.

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 (A 4).

Measurements of Lowest Vibration Levels

The weakest vibration signal that can be usefully detected is limited by randomly varying voltages and currents existing in the circuits of the amplifier. Every electrical conductor produces an irregularly varying voltage across its terminal as a result of the random motion of the free electrons in the conductor, caused by thermal action. This phenomenon is termed as noise in the circuit. The magnitude of this noise (R.M.S.) is given by

$$e_{RMS} = \sqrt{4\,k\,TR\,(f_2 - f_1)}$$

where k = Boltzmann's constant,
T = absolute temperature,
R = resistance component of impedance across which agitation is produced,
$(f_2 - f_1)$ = bandwidth of the noise,

if the resistance R is constant over the frequency range.

The largest source of noise is normally the amplifying element, i.e. the tube or transistor, in the first stage. It gives a shot noise which is normally several dB higher than the resistor noise, and a flicker noise which at low frequencies is even higher.
However, in the measurement of low level signals picked up by accelerometers, additional noise is introduced, either due to pick-up from nearby electrical or magnetic fields or by internal generation due to cable motion. The pick up problem could be quite severe on account of the high impedances involved. The primary cause of generated cable noise is the "triboelectric effect" which is, however, minimized by a special treatment of the coaxial cables.

Lowest Measurable Vibration Levels (Worst Case)

	ACCELEROMETERS				PRE-AMPLIFIERS								
					1606 POSITION FULL GAIN 38 dB						2616	2622	
	Accelerometer Set No. / Type No.	Capacity In pF	Voltage Sensitivity (minimum) mV/g	Charge Sensitivity (minimum) pC/g	ACC 10^3 cm/sec² L.L.F. 2 Hz mg	Velocity 10^2 cm/sec L.L.F. 3 Hz mm/sec	Velocity 10 cm/sec L.L.F. 30 Hz μm/sec	Displacement 10 cm L.L.F. 3 Hz μm	Displacement 10^{-1} cm L.L.F. 30 Hz nm	Displacement 10^{-3} cm L.L.F. 300 Hz nm	Gain 0 dB L.L.F. 0.13 Hz mg	Voltage Amplifier L.L.F. 2 Hz mg	Charge Amplifier L.L.F. 2 Hz mg
WITH 1/3 OCTAVE FILTER 1614 FREQUENCY RANGE 2–10 kHz	4312/32 4314/34	1000	45	40	0.07	0.02	2	0.9	8	0.07	0.13	2.5	.5
	4313/33 4315/35 4320/40	1000	14	14	0.2	0.06	6	3	26	0.2	0.4	2.5	3.5
	4318/38	1000	100	100	0.03	0.01	1	0.4	4	0.03	0.06	2.5	3.5
	4319/39 4323/43	1000	10	10	0.3	0.1	10	4	40	0.3	0.6	2.5	3.5
	4324/44	900	1.7	1.3	2	0.5	50	30	220	2	3		
LINEAR FREQUENCY RANGE 2–40 kHz	4312/32 4314/34	1000	45	40	0.5	0.2	* 15	20	* 100	* 2	0.8	14	15
	4313/33 4315/35 4320/40	1000	14	14	1.5	0.7	40	70	300	7	2.5	14	15
	4318/38	1000	100	100	0.2	0.1	6	10	50	1	0.4	14	.5
	4319/39 4323/43	1000	10	10	2	1	60	100	500	10	4	14	15
	4324/44	900	1.7	1.3	15	6	400	600	3000	60	20		

L.L.F. = Lower Limiting Frequency.
* With 20 Hz High Pass Filter Values are possibly better.
m = 10^{-3}.
μ = 10^{-6}.
n = 10^{-9}.

$\frac{\text{Signal}}{\text{Noise}} \geq 5$ dB

Lowest Measurable Vibration Levels (Worst Case)

PRE-AMPLIFIERS														
23	2624			2625 POSITION 20 dB GAIN										
.16 Hz g	Sensitivity 0.1 mV/pC L.L.F. 0.003 Hz mg	Sensitivity 1 mV/pC L.L.F. 0.03 Hz mg	Sensitivity 10 mV/pC L.L.F. 0.3 Hz mg	ACC 10 m/sec² L.L.F. 1 Hz mg	Velocity 3 m/sec L.L.F. 1 Hz mm/sec	Velocity 0.3 m/sec L.L.F. 10 Hz μm/sec	Velocity 0.03 m/sec L.L.F. 100 Hz μm/sec	Displacement 1000 mm L.L.F. 1 Hz μm	Displacement 100 mm L.L.F. 3 Hz μm	Displacement 10 mm L.L.F. 10 Hz μm	Displacement 1 mm L.L.F. 30 Hz nm	Displacement 0.1 mm L.L.F. 100 Hz nm	Displacement 0.01 mm L.L.F. 300 Hz nm
0.0'	1.8	0.3	0.18	0.08	0.05	2	0.3	6	0.8	0.04	4	0.4	0.04
0.3	5	0.8	0.5	0.3	0.15	6	0.8	20	2.5	0.13	13	1.3	0.13
.04	0.7	0.1	0.07	0.04	0.02	1	0.15	3	0.4	0.02	2	0.2	0.02
.4	7	1	0.7	4	2	10	1.5	30	4	0.2	20	2	0.2
.2	55	8	5.5	2.2	1.3	50	7	170	20	1	100	10	1
				POSITION 0 dB GAIN									
.6	9	1	0.35	1	0.7	70	* 7	220	22	2.2	* 220	* 22	* 2.2
2	26	3	1	3	2.3	230	23	720	72	7.2	720	72	7.2
[illegible]	3.5	0.4	0.15	0.4	0.3	30	3	100	10	1	100	10	1
3	35	4	1.5	4	3	300	30	1000	100	10	1000	100	10
17	280	35	10	25	19	1900	190	6000	600	60	6000	600	60

VELOCITY AND DISPLACEMENT VALUES ARE REFERRED TO INPUT AS IF INTEGRATION WAS CARRIED OUT BEFORE THE AMPLIFIER.

In order to determine the lowest vibration levels measurable by B & K accelerometers and preamplifiers, an experiment was carried out to investigate the noise levels in the measuring systems. Using the measuring set-up shown in Fig. 1 the noise spectra were obtained for each of the preamplifiers.

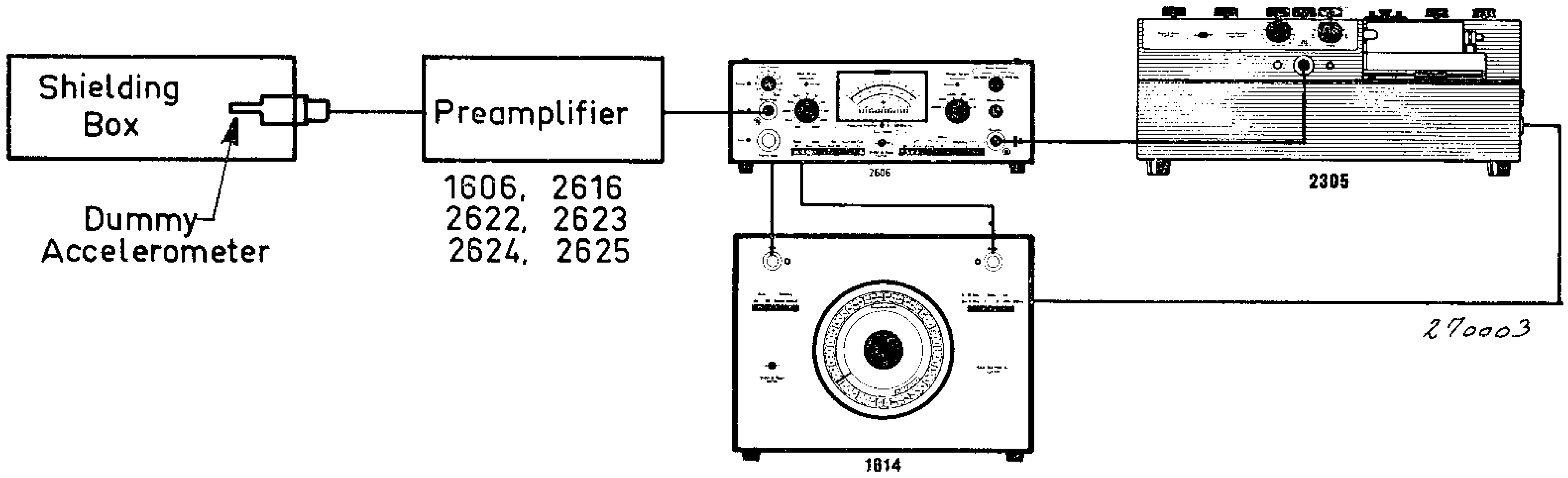
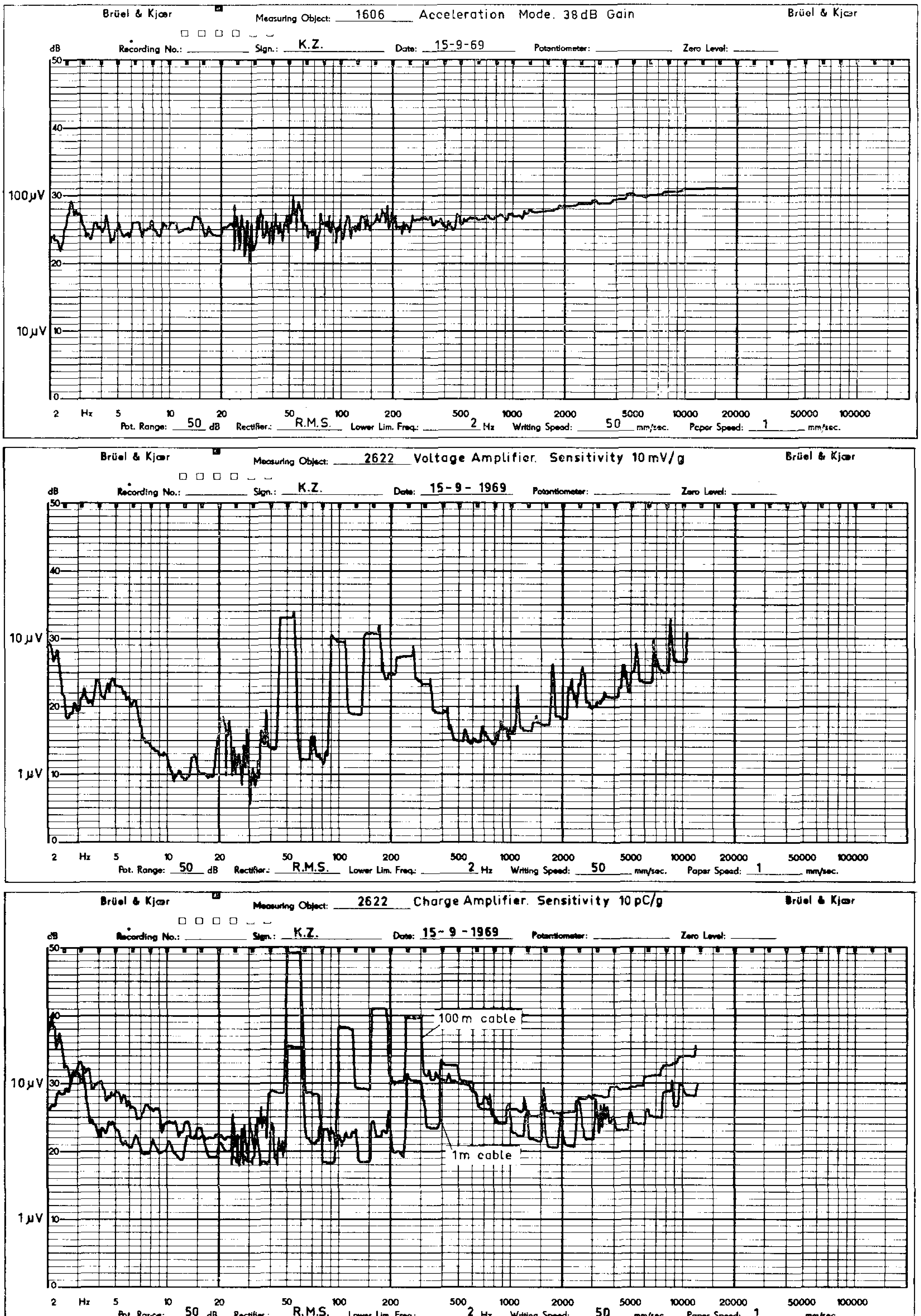


Fig. 1. Measuring set-up.

A dummy accelerometer (capacitance 1000 pF) was used instead of an actual accelerometer to effectively isolate the system from mechanical vibrations and it was placed in a "shielding box" to minimize pick up from environmental electrical fields. A 1/3 octave frequency analysis of the noise was carried out for each of the preamplifiers in turn. The recorded noise spectra are shown in Figs. 2–10. The maximum noise level (referred to input) at the various settings of the preamplifiers with full gain were converted to the appropriate acceleration, velocity or displacement assuming that any signal 5 dB above the noise level is easily detectable (ISO Recommendation 177).
The tabulated results give the (worst case) minimum vibration levels (R.M.S.) measurable in the frequency range 2 Hz to approximately 10 kHz except where frequency limitations are imposed by integration networks. The bottom half of the table gives similar values for linear frequency range 2 Hz – 40 kHz. From the noise spectra it can be seen that lower values than those quoted can be obtained if the operating frequencies are above 10 Hz and away from the mains frequency and its harmonics.
The values given here are for a cable length of 1 meter. For voltage amplifiers larger cable length lead to a reduction in sensitivity. However, in the case of charge amplifiers 100 meter cable length was also used and the difference in noise levels can be seen from the spectra.
The frequency spectra obtained with battery operation of the preamplifiers 2623, 2624 and 2625 are superimposed on those obtained with Power Supply Type 2805 and are shown with finer lines. A slight reduction in hum level is obtained with battery operation.
The above-mentioned effects can all be seen in Figs. 2–10.

K. Zaveri.



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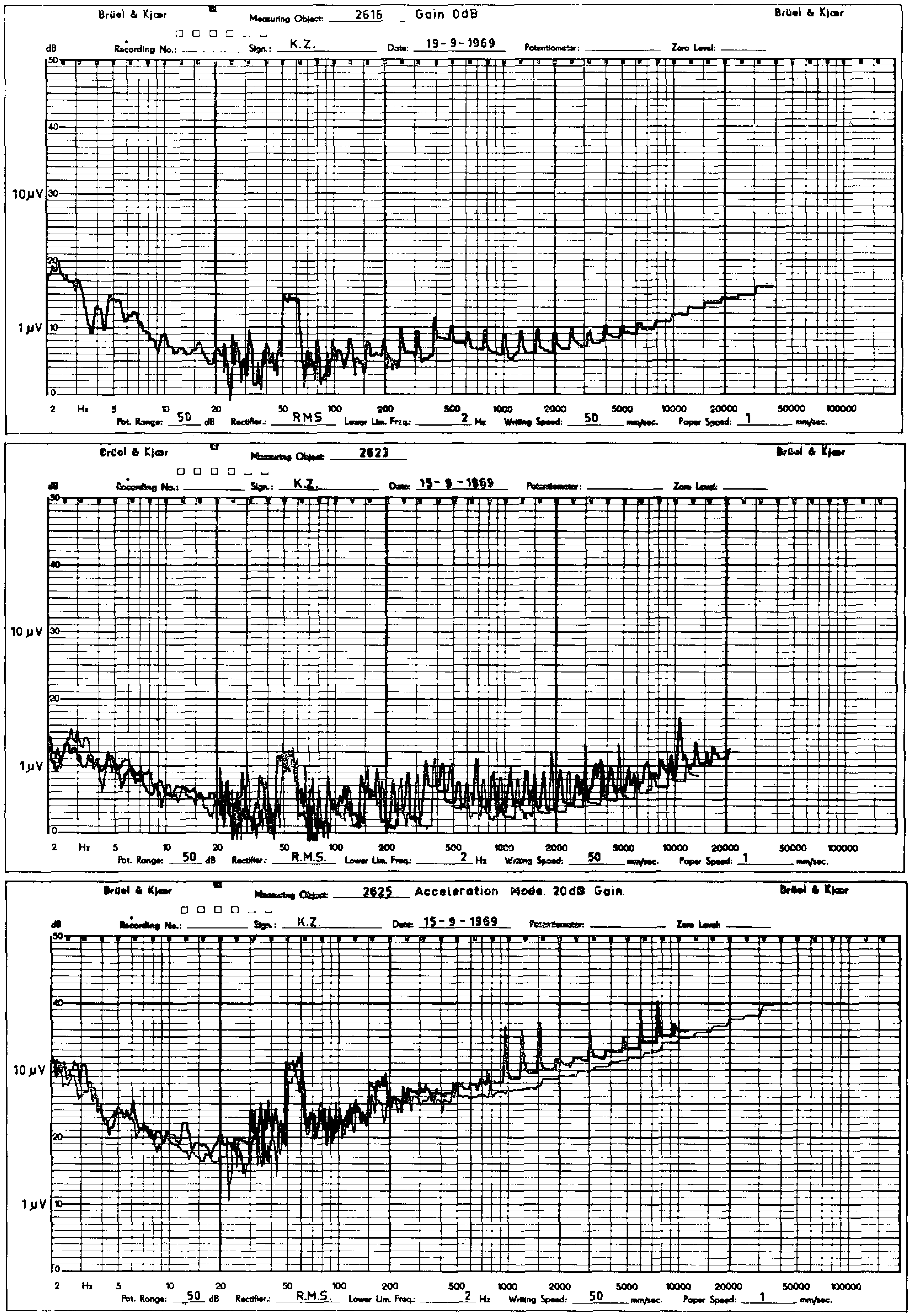
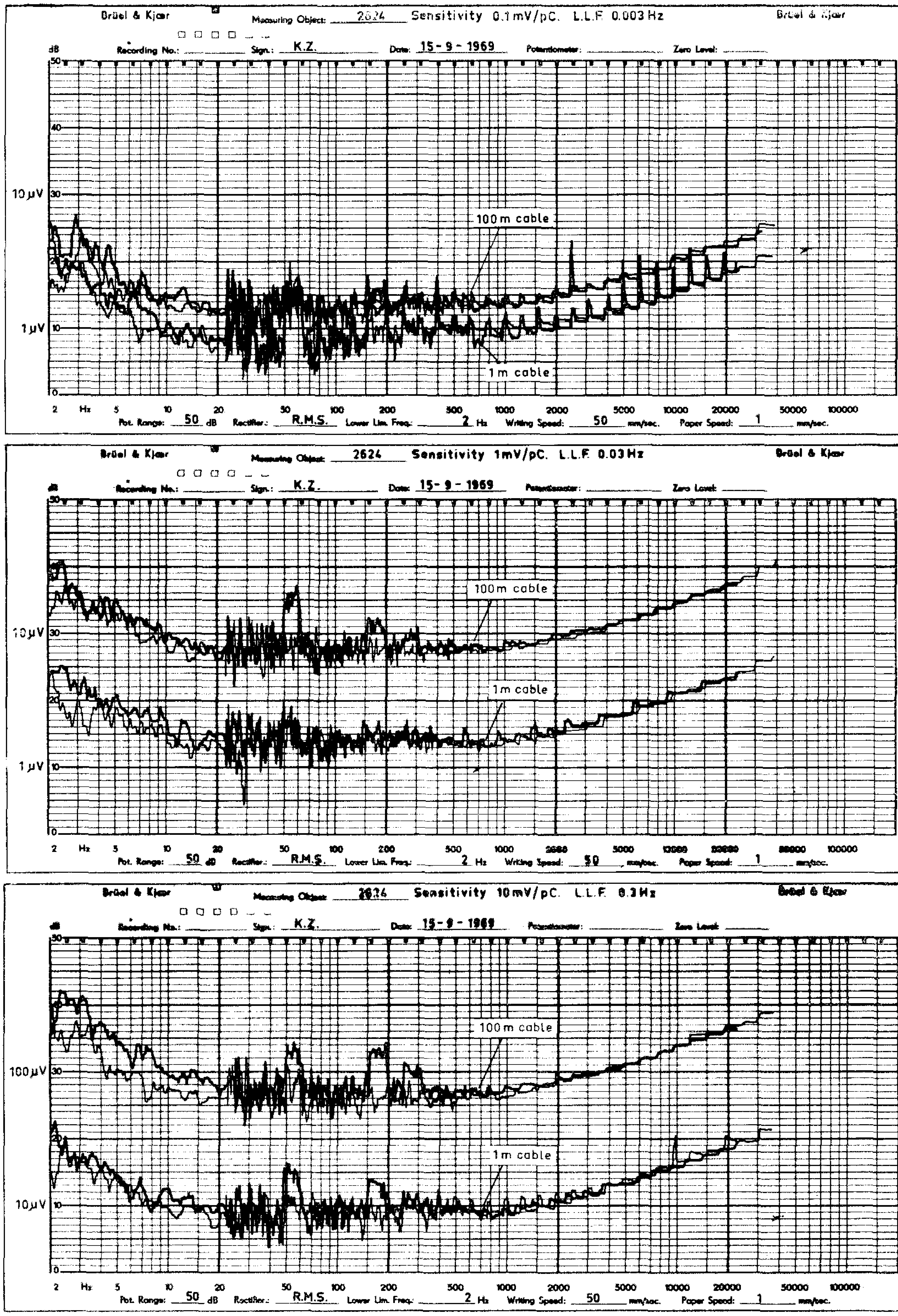
Figs. 2–4. 1/3 octave noise spectra of preamplifiers Type 1606 and 2622.



270006

Figs. 5–7. 1/3 octave noise spectra of preamplifiers Type 2616, 2623 and 2625.
Operation with Power Supply Type 2805 ———————
Operation with batteries - - - - - - - - - - -



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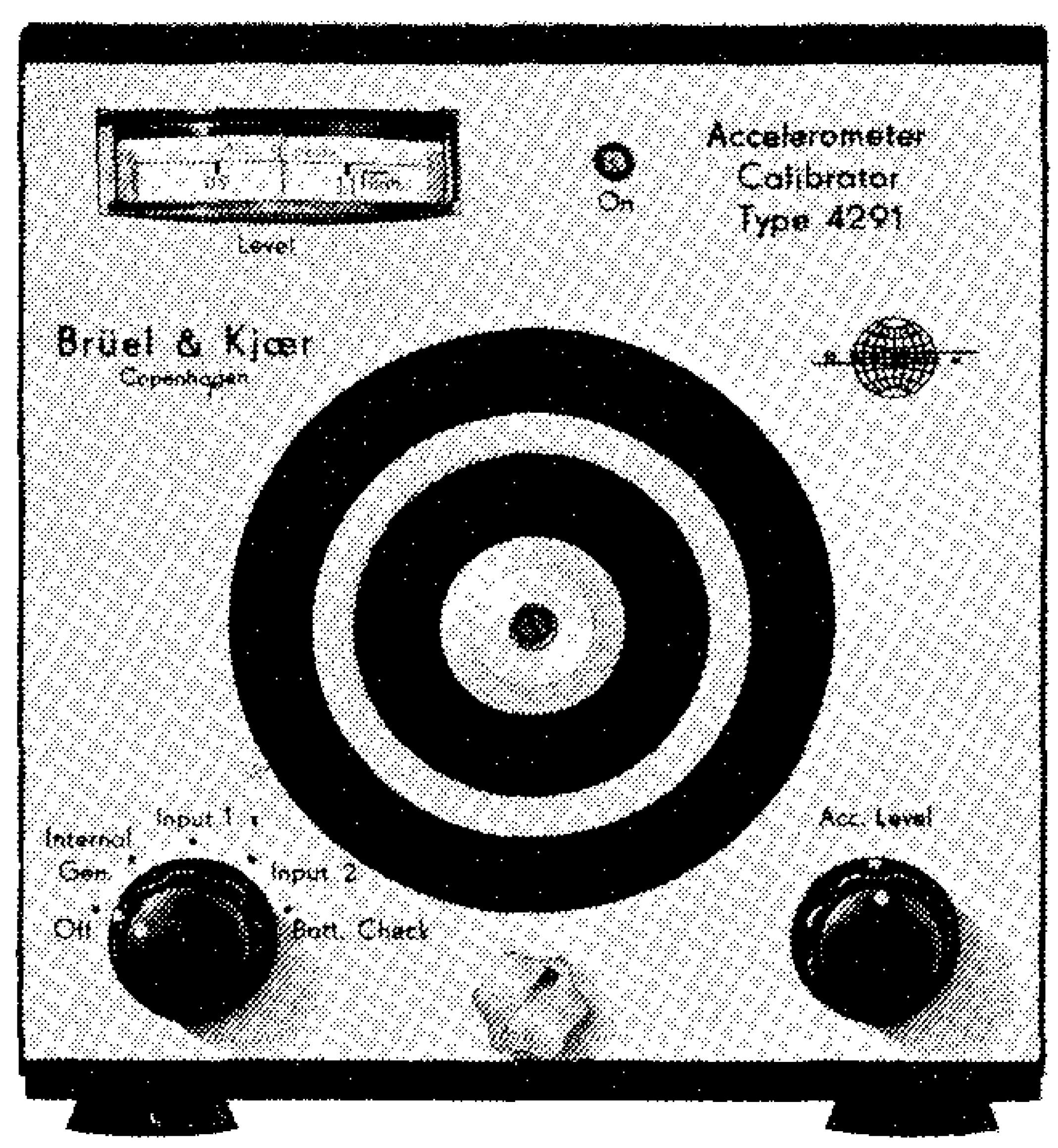
Figs. 8–10. 1/3 octave noise spectra of preamplifier Type 2624.
Operation with Power Supply Type 2805 ————
Operation with batteries ————

News from the Factory

Vibration Calibrator Type 4291

The Vibration Calibrator Type 4291 is a portable unit for calibration of accelerometers and other vibration transducers in the laboratory or in the field.

It consists of a small electromagnetic shaker system with two coils, one for driving the moving mass and one for velocity measurements.



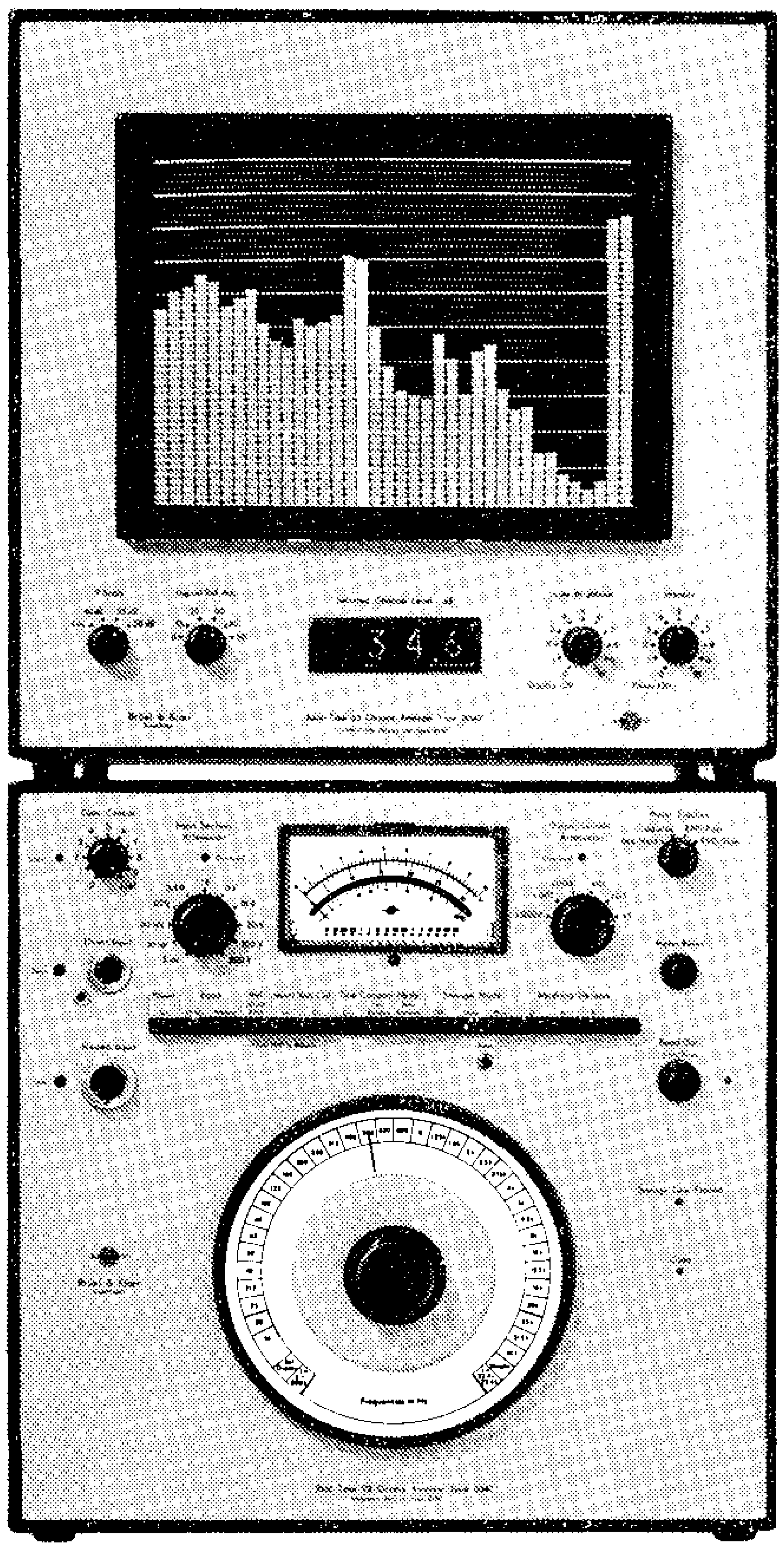
A built in Wien Bridge Oscillator gives a frequency of $\omega = 500$ (USAS - S2.2. 1959 and DIN 45666) and the amplitude is adjustable to 1 g peak, read off a built-in meter.

For a more accurate calibration the back to back calibration method or the reciprocity calibration method can be utilized. Also the insert voltage method of measuring open circuit voltage is facilitated. The power requirement of the Calibrator is supplied from internal batteries or from exernal 28 V DC.

Real-time 1/3 Octave Analyzer Type 3347

The 3347 is a combination of the 1/3 Octave Frequency Analyzer Type 2130 and the Control and Display Unit Type 4710. Some of its features have been described in "Real Time Analysis", Technical Review number 4, 1969 and in "1/3 Octave Spectrum Readout of Impulse Measurements" in this volume.

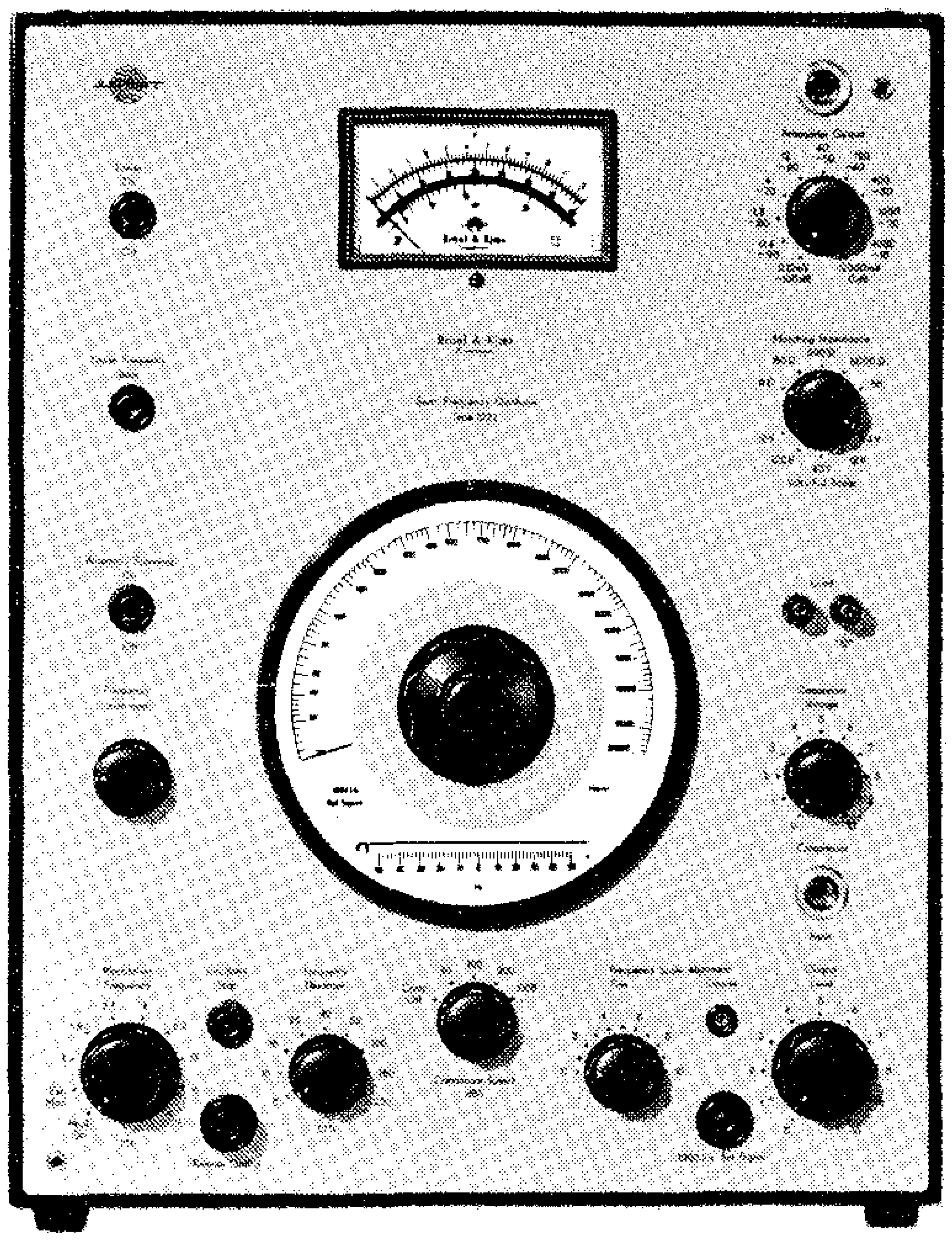
The Real-time display and readout facilities of the 3347 makes it an ideal instrument for all applications which include 1/3 octave analyses. The 12" CRT-display gives the operator instant information on the state of the analyzed system and thus enables him to continuously adjust until the desired conditions are obtained.



To obtain full advantage of the 3347 it should normally be operating in conjunction with a computer – either on line or via punched or magnetic tape. This combination facilitates various intricate types of analysis which were earlier too tedious and time-consuming to carry out.
Standard software will be made available for some such procedures as for example PNdB analysis.

Beat Frequency Oscillator Type 1022

Type 1022 has been redesigned using all solid state electronic circuits thus reducing power requirement to 26 W and weight to 15 kg (33 lbs.).
Other improvements are:
The compressor dynamic range has been extended (to more than 55 dB), and the compression slope has been raised to 55 dB for an input change of 1.5 dB.
The "Oscillator-Stop" function is now obtained by earthing the compressor output signal. In this way the output signal is switched off and on without overshoot when the "Oscillator-Stop" button is pressed or when the "dead zone" of the frequency scale is swept over. The new way of stopping the



output signal allows the internal generators to be operating continuously, which is desirable when used in conjunction with the Type 2020 Heterodyne Slave Filter.
An output current limiter has been applied so that output current can not exceed $\pm$ 300 mA peak.

PREVIOUSLY ISSUED NUMBERS OF BRÜEL & KJÆR TECHNICAL REVIEW

1-1962 Artificial Ears for the Calibration of Earphones of the External Type, part 2.

2-1962 Loudness Evaluation.

3-1962 Testing of Stereophonic Pick-ups by means of Gliding Frequency Records.

4-1962 On the Use of Warble Tone and Random Noise for Acoustic Measurement Purposes.
Problems in Feedback Control of Narrow Band Random Noise.

1-1963 Miniature Pressure Microphones.
Methods of Checking the RMS Properties of RMS Instruments.

2-1963 Quality Control by Noise Analysis.
A. F. Nonlinear Distortion Measurement by Wide Band Noise.

3-1963 Effects of Spectrum Non-linearities upon the Peak Distribution of Random Signals.

4-1963 Non-linear Amplitude Distortion in Vibrating Systems.

1-1964 Statistical Analysis of Sound Levels.

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Sweep Random Vibration.

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4-1964 The Accuracy of Condenser Michrophone Calibration Methods. Part I.

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The Use of Comparison Bridges in Coil Testing.

3–1965 Analog Experiments Compare Improved Sweep Random Tests with Wide Band Random and Sweep Sine Tests
The Frequency Response Tracer Type 4709.

4–1965 Aircraft Noise Measurement, Evaluation and Control.

1–1966 Windscreening of Outdoor Microphones.
A New Artificial Mouth.

2–1966 Some Experimental Tests with Sweep Random Vibration

3–1966 Measurement and Description of Shock.

4–1966 Measurement of Reverberation.

(Continued on cover page 2)

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