A MULTI-CHANNEL DIGITAL NOISE MEASURING APPARATUS FOR THE MEASUREMENT OF NOISE PROPAGATION

H.W. Jones and F. Babott
Acoustics Group,
University of Calgary
Calgary, Alberta
Canada T2N 1N4

ABSTRACT

This paper describes a multi-channel system designed to make simultaneous measurements at up to six widely spaced points. The apparatus was used to determine the attenuation of traffic noise as it propagated over various forms of terrain in urban communities.

The method of data acquisition, digital coding, interfacing and subsequent computational treatment to obtain any of the A weighted statistical properties of the noise is described.

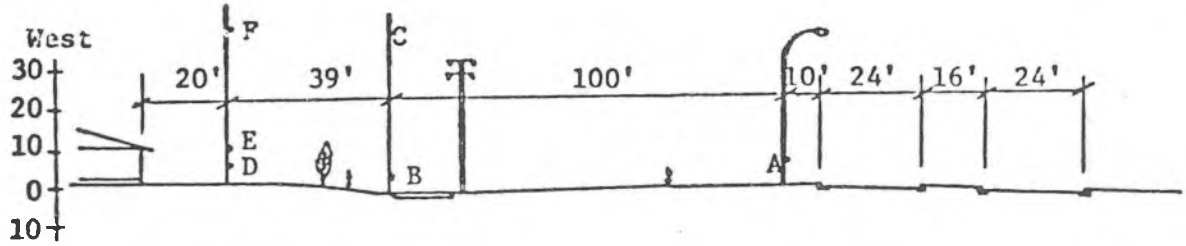
The use of the instrument as a long term (24 hours) survey meter is described.

Calibration methods are outlined.

INTRODUCTION

This paper describes a noise measuring apparatus which was designed and built at the University of Calgary in support of studies relating to traffic noise in the urban environment. The study required detailed measurements of the attenuation of road and railway noise by an array of microphones placed at right-angles to the right of way, as shown for example in Figure 1. These measurements were used to determine the accuracy of traffic noise prediction methods and the adequacy of barrier attenuation desigh formulae[1] and for the development of scaled model analogues.[2] The instrumentation was subsequently used to assist in social surveys and human reaction studies related to the original traffic noise problems.

The nature of the study implied the satisfaction of two operational requirements. First the accuracy of the differences between the microphone readings was of greater consequence than the absolute accuracy of any individual reading. Second, the knowledge of the temporal sequence of the readings was important. These considerations led to the design of an apparatus which was capable of sampling six microphones in succession with the microphones being up to 150 feet from the central measuring instrument.



SITE #13 Middle Section	MIC's	A	B	C	D	E	F
Blackfoot Trail S.W., Calgary	Heights	5	5	40	5	9	40

(160' north of n PL of lane between
Fyfree Road & Fleetwood Drive)

Figure 1. Typical Microphone Array Used in Field Measurements.

DESIGN CONSIDERATIONS

The primary design criteria was set by the need to use an array of microphones in a sampling mode. The rate of sampling is set by the statistical problems it invokes. Once a sampling rate is accepted as the prime criterion then it implies limits for the integration time and the dynamic range of the instrument. Fisk[3] has provided an excellent analysis of the statistical problems of sampling traffic noise. He concluded that the error introduced by sampling is given by:

$$\frac{\Delta L_{eq}}{\Delta L'_{eq}} = \left[\coth \left(\frac{\pi t_c}{\Delta t} \right) \right]^{\frac{1}{2}} \qquad (1)$$

where $\Delta L'_{eq}$ is the error for continuous sampling. The characteristic time t_c is given by $t_c = 2a/V$, where V is the vehicle speed and a is the source receiver distance perpendicular to the road. Δt is the sampling period and ΔL_{eq} the error associated with this period. Equation (1) is shown graphically in Figure 2.

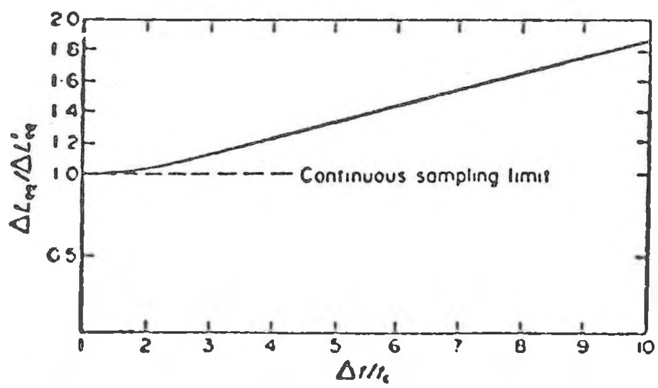


Figure 2. Error in L_{eq} Related to Sampling Period Δt.

We concluded that sampling rates close to 1 per second were required for the most adverse conditions under which it would be necessary to operate. Our initial version of the apparatus used one sample per microphone per second for three microphones. A second version used up to eight microphones (but normally six) and these were sampled at a rate of one microphone each one third of a second.

The acceptance of a particular rate implied that the dynamic range and frequency response of the instrument had to be limited. Previous experience[4] had shown that a dynamic range of 40 dB was more than adequate for this type of work provided that the minimum level could be chosen to satisfy any particular circumstance i.e. ranging from 40 to 80 dB or 50 to 90 dB and so on. It can be supposed that the successive reading between two microphones can differ by 40 dB i.e. the full dynamic range of the instrument. If a single analogue circuit is used for all microphones the time constant of the true r.m.s. circuit has to be chosen so that any reading does not mask the one which immediately follows it. This requirement leads to a slight modification of the bandwidth of the analogue circuit such that the output at frequencies below 60 Hz are slightly reduced. The effect is very small, however, and of no consequence for traffic noise measurements which are made using an A weighting network.

In order that the temporal information should be preserved it was decided to store the results as they were obtained so that they could be reproduced in correct sequence. The cheapest method available to us for this purpose was an inexpensive cassette tape recorder.

The need for maximum accuracy in the differences between the microphones was met by feeding the sampled signals to a single analogue and digital circuit. Consequently the processing was identical for all microphone outputs and no matter what drifts occurred the effects were the same for each reading.

DESCRIPTION OF CIRCUITS

Figure 3 shows the schematic arrangement which was used.

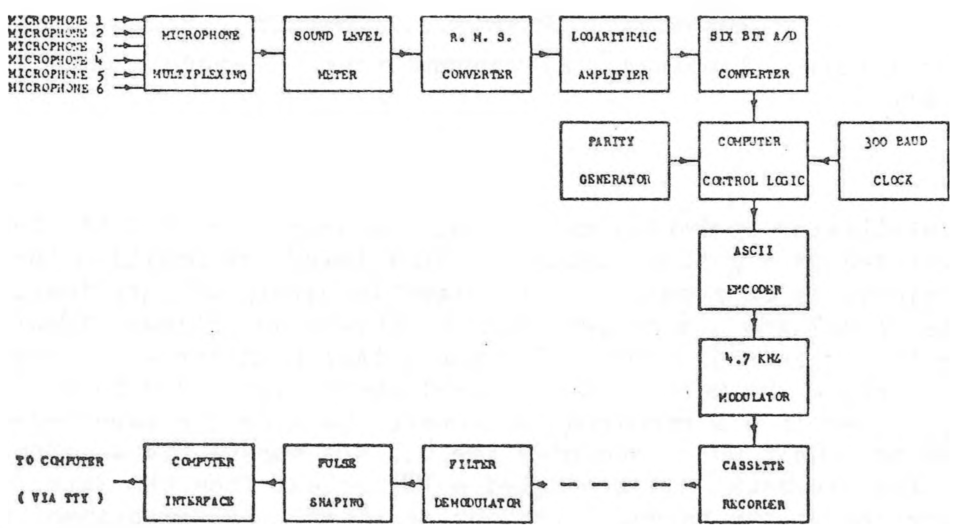


Figure 3. Schematic of Sampling Sound Level Meter.

The output from the six microphones was fed to a reed relay multiplexing unit. Each microphone was connected in turn to a sound level meter. This meter served two functions; it provided an approximate direct reading of the signal being measured and it served as an A weighted preamplifier. The output from this stage was fed to a true r.m.s. circuit and then to a logarithmic amplifier. The output from the logarithmic amplifier was fed to an analogue-to-digital converter which sampled the signal immediately prior to the end of the microphone switching period. A 6 bit converter was used because it was cheaper than more complex types and provided sufficient accuracy (5/8 dB steps). The digital output was displayed on light emitting diodes in binary form so that the operator could check the functioning of the instrument.

The six bit binary output was buffered by additional bits, to provide an ASCII code characters according to the scheme shown in Table 1. The clock circuit generated bits at a rate of 300 per second i.e. at a rate of 100 bits per microphone sampling period. There was thus an opportunity to generate and store associated information. Consequently the data sequence shown in Table 2 was used for each sampling period.

Table 1. Coding for Tape Recording.

<table>
<tr><td>Binary Level</td><td>0</td><td>1 → 26</td><td>27,28,29,30,31,32</td><td>33 → 58</td><td>59,60,61,62,63</td></tr>
<tr><td>ASCII coded character</td><td>@</td><td>A → Z</td><td>[\ ] ^ > `</td><td>a → z</td><td>{ , | , } , ~ , ?</td></tr>
</table>

Example of Coding

Binary Level 23	Character W.
Binary output	010111
Additional bits	1 1
Final character (W)	1 010111 1

Table 2. Sequence of Data Recording.

Carriage return, linefeed, microphone number, space, data, carriage return etc.

An oscillator operating at 4.7 KHz was used to record the data. This signal was fed to a gating circuit which allowed the positive logic digital signal to be recorded on the tape in bursts of pure tone. A frequency of 4.7 KHz was chosen because it gave an optimum signal to noise ratio in the recovered signal. The cheap tape recorder which was used accepted tapes which would run at normal speeds for a period of 90 minutes per side. When it was required to recover the data the tape recorder was connected to a unit which accepted the 4.7 KHz bursts via a narrow band filter. The playback unit generated a TTY signal from the data for transmission by any of the normal telephone couplers. An arrangement was made to use the play-back unit in association with a normal computer terminal so that computer control instructions could be generated as required.

The instrument was adapted to serve as a community noise monitor. In this role the instrument sampled a single microphone once per second and stored the information on a tape which ran at one third normal speed. It was arranged that the circuit operated for ten minutes in a 25 minute period. The tape recorder was able to store data for an 18 hour period when it was operated in this manner. Consideration was given to the provision of short term digital storage which would store the data obtained from continuous sampling (with the usual loss of temporal order) for periods of up to 30 minutes. It was planned that at the end of that time stored data would be read onto tape, and the registers reset for the next period. The expense of this modification has not justified its incorporation up to this time because the additional accuracy it could produce was not required.

The apparatus, which is shown in Figure 4, operated from storage batteries which required charging after 48 hours of continuous operation.

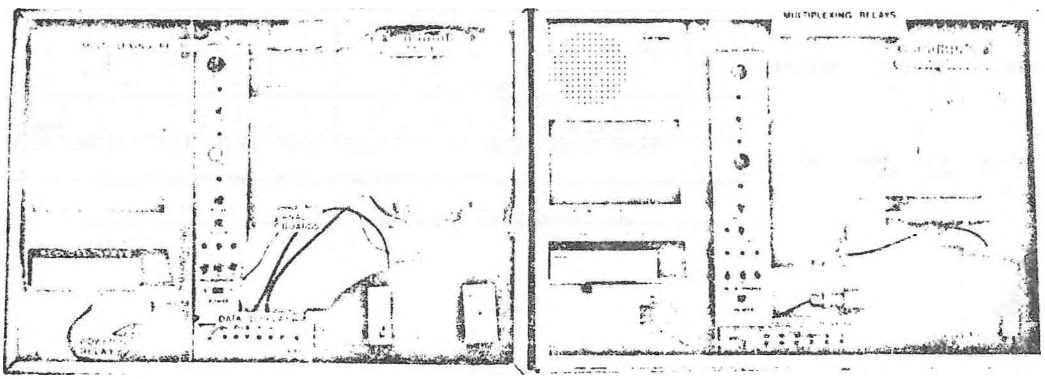


Figure 4. Early and Later Version of Apparatus.

Figure 5 a shows the output obtained directly from the magnetic tape and Figure 5 b shows the data sorted and interpreted. Figure 6 shows a histogram obtained from the data together with the derived descriptors and the microphone calibrations. It is to be noted that the sensitivity of the microphone placed next to the roadway was reduced by 10 dB relative to its neighbours. Table 3 shows the data obtained for field readings associated with the microphone array shown in Figure 1.

Traffic Noise Data Sample Before Statistical Analysis

6 Microphones ASCII Coded Data With Microphone Nos.		6 Microphone Data After Conversion from ASCII to Numerical Values 0 to 63					
1	f	38	34	40	32	38	4
2	b	30	36	42	30	38	3
3	h	38	35	40	27	35	4
5	`	30	30	35	23	29	4
6	f	21	25	35	26	31	3
7	D	22	30	37	26	35	5
1	^	27	33	40	29	39	3
2	d	40	37	39	28	34	3

Figure 5 a. Figure 5 b.

Sample Statistical Computer Output For 1 Hour Data in 6 Microphone Mode of Traffic Noise

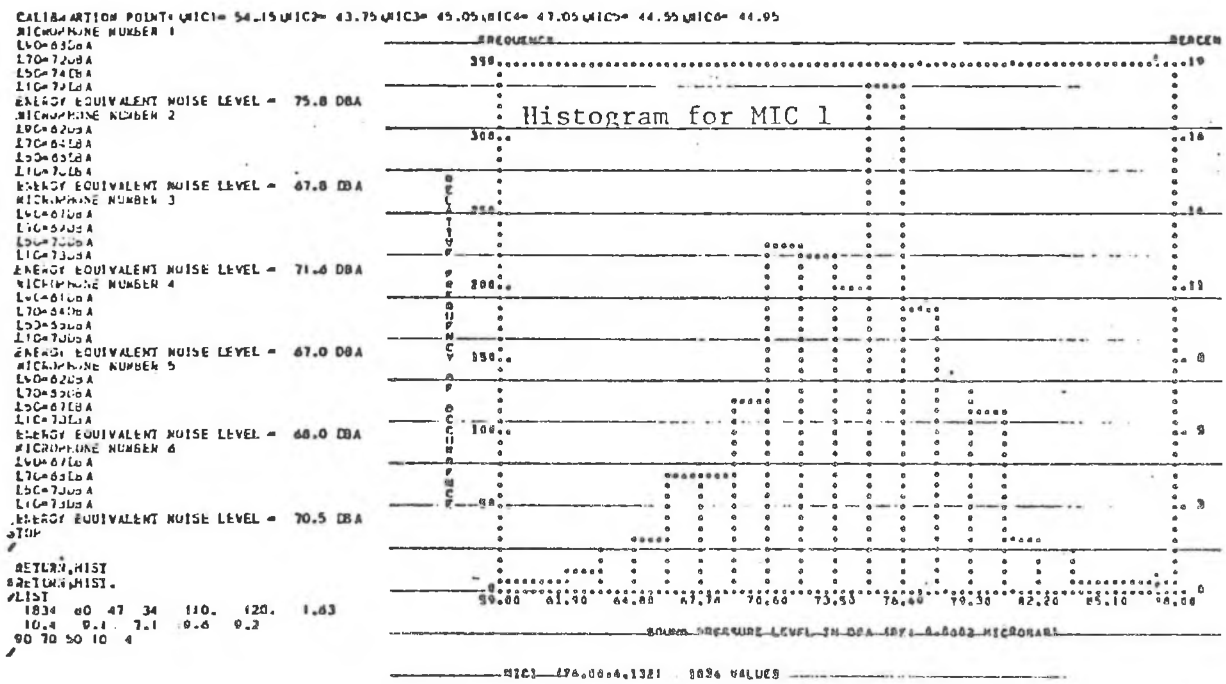


Figure 6.

CALIBRATION

The calibration of the system with its 6 microphones was achieved in two stages. First each microphone was placed in an enclosure with an approximately white noise field. The intensity of the sound was a constant when averaged over a period of time. This average level was measured by a precision sound level meter which had been calibrated separately. Each microphone with its associated head amplifier and cable was placed in the same position in the sound field in turn and the digital output recorded. Second, in the field, a precision sound level meter was used to monitor the same sound field as the microphone placed closest to the road. This data was used for comparison with the information obtained by the sampling instrument. At the start and finish of each field measurement a pistophone operating at several frequencies was used to check the outputs of each microphone in turn.

CONCLUSION

A measuring system, designed primarily for field measurements of traffic and railway noise and its attenuation by barriers and buffers is described. The method uses relatively cheap integrated circuits and records its data on a cheap tape recorder. The advantages of the instrument in reducing errors related to the differences in the readings between the microphones is described. Sample field data is given. The operation of the unit as a survey monitor is also described.

Table 3.

TRAFFIC NOISE MEASUREMENT DATA

SITE NO. 11 SECTION Middle

LOCATION Blackfoot Trail S.W., Calgary

DATE 26-5-76 TIME PERIOD 11:40 - 12:42

MICROPHONE POSITION ON DIAGRAM	MEASURED NOISE LEVEL (dBA)				
	L_{10}	L_{50}	L_{70}	L_{90}	L_{eq}
A	82	76	73	69	79.4
B	67	61	60	58	63.7
C	73	70	69	66	71.0
D	62	57	55	53	59.8
E	65	59	57	55	61.9
F	70	68	67	61	67.3

DIRECTION	VEHICLE VOLUME (VEHICLES/HR)			
	AUTOS	MEDIUM TRUCKS	HEAVY TRUCKS	MOTOR-CYCLES
Southbound	1021	32	48	2
Northbound	472	45	89	0

AVERAGE VEHICLE SPEED N.B. - 47.5 S.B. - 47 mph

AMBIENT TEMPERATURE 16 °C

WIND SPEED 5 mph DIRECTION S.E.

SKY DESCRIPTION Clear, light clouds and haze

GROUND COVER 2" grass from Blackfoot to Mic B, gravel and clay from Mic B to Mic D

ROADWAY GRADIENT 1.5 %

ABSOLUTE INSTRUMENT ERROR 0.7 dBA max.

STATISTICAL STANDARD DEVIATION OF L_{eq} (ΔL_{eq}, ref. D.J. Fisk) 0.2 dBA

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