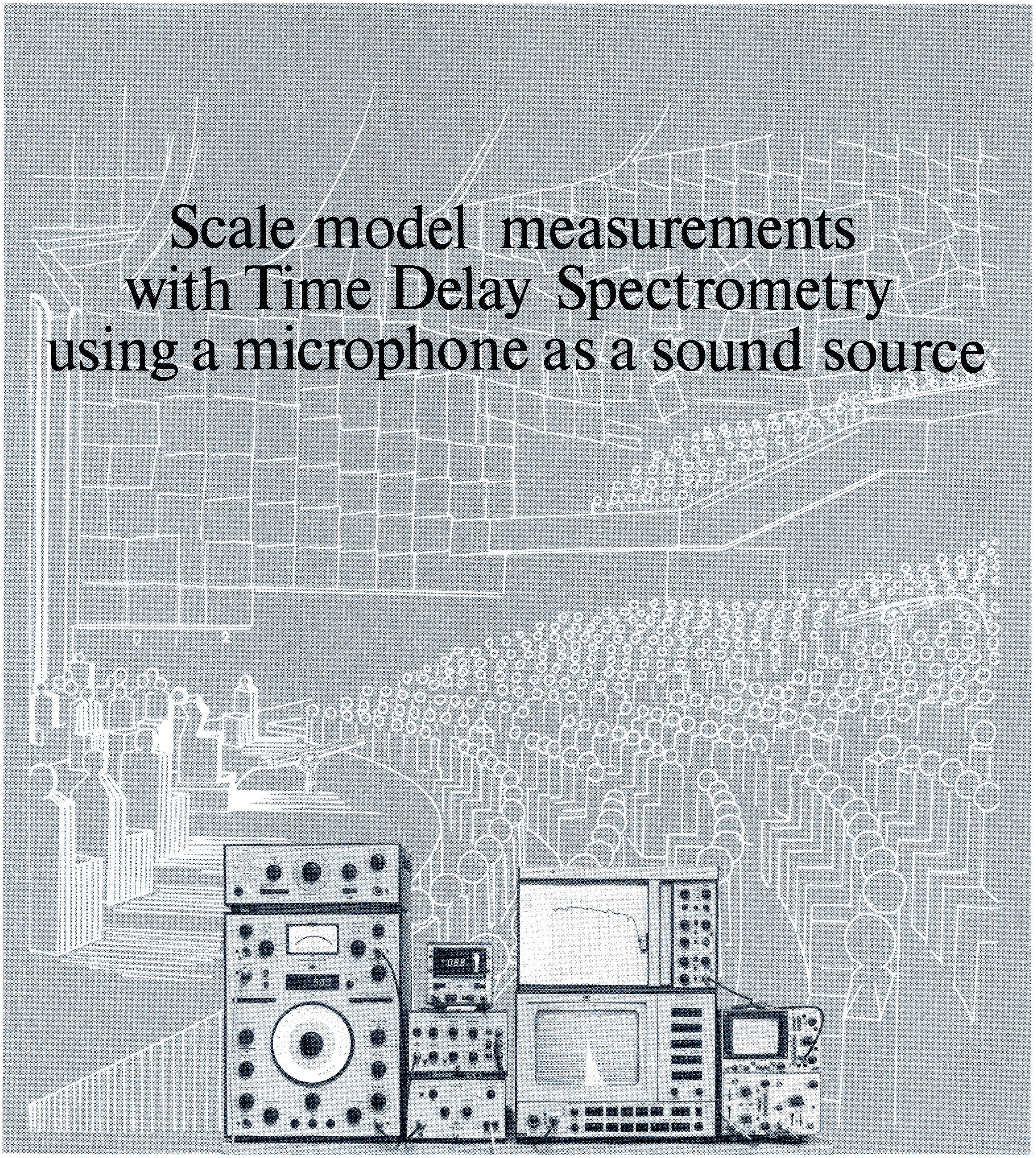
Scale model measurements with Time Delay Spectrometry using a microphone as a sound source



Scale model measurements with TDS using a microphone as a sound source

by *André Perman*
Brüel & Kjær

Introduction

A high quality, condenser microphone is undoubtedly the best device available for use on the receiving side of an acoustical measurement chain. The condenser microphone's sound emitting counterparts such as loudspeakers or spark generators are generally the "weakest links" in such a chain. In terms of stability, flat frequency response or omnidirectionality, they tend to be substantially inferior to the condenser microphone.

The condenser microphone can be used reciprocally as a sound source which possesses many desirable features: a very flat frequency response over a wide frequency range, omnidirectionality and small physical dimensions. However, a major limitation of a condenser microphone sound source is the relatively low sound pressure level produced. Consequently a poor signal to noise ratio is obtained with most measurement techniques.

By employing Time Delay Spectrometry (TDS), this limitation is largely overcome. Due to the outstanding noise rejecting properties of TDS, signal to noise ratios better than 40 dB are obtained in normal office surroundings and far better than this in very quiet environments or within acoustic couplers.

Thus with TDS a condenser microphone sound source becomes a straightforward tool for a multitude of otherwise cumbersome, two-port measurements. The applications include: scale model tests, investigations of absorbers and reflectors, acoustical impedance, free field calibration of microphones, production of well defined sound fields, leak detection in air ducts and quality control of musical brass instruments. Examples of the first two of these applications as well as the necessary basic TDS and microphone theory are described in this note.

Measurements

The measurements were performed by the author inside a 1:10 scale model of the old hall of Trinity College, Dublin, Ireland. The original hall dates back to 1865. In 1973 a renovation of the hall was initiated to convert it into a concert hall. The responsibility for the acoustical design was entrusted to Dr. V. L. Jordan [1]. The successful renovation was completed in 1980. The 1:10 hardboard scale model used by Dr. Jordan for his investigations is to-day situated at the Technical University of Denmark, where it is used for educational and research purposes (Fig.1).

Sound distribution on stage

The set-up for investigating the sound transmission on stage is shown in Fig.2. This set-up models the manner in which two orchestra members hear one another during a performance on stage.

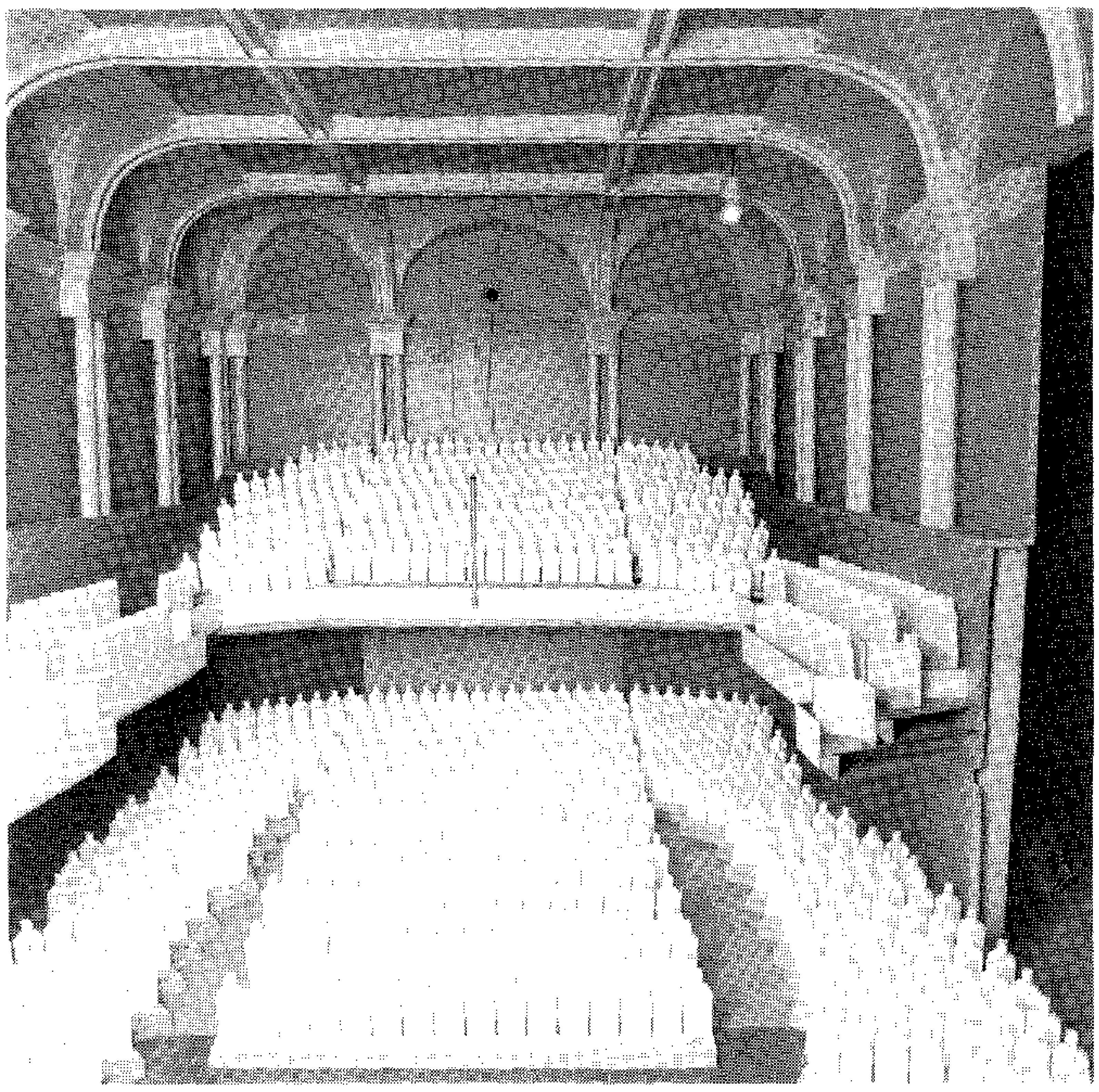
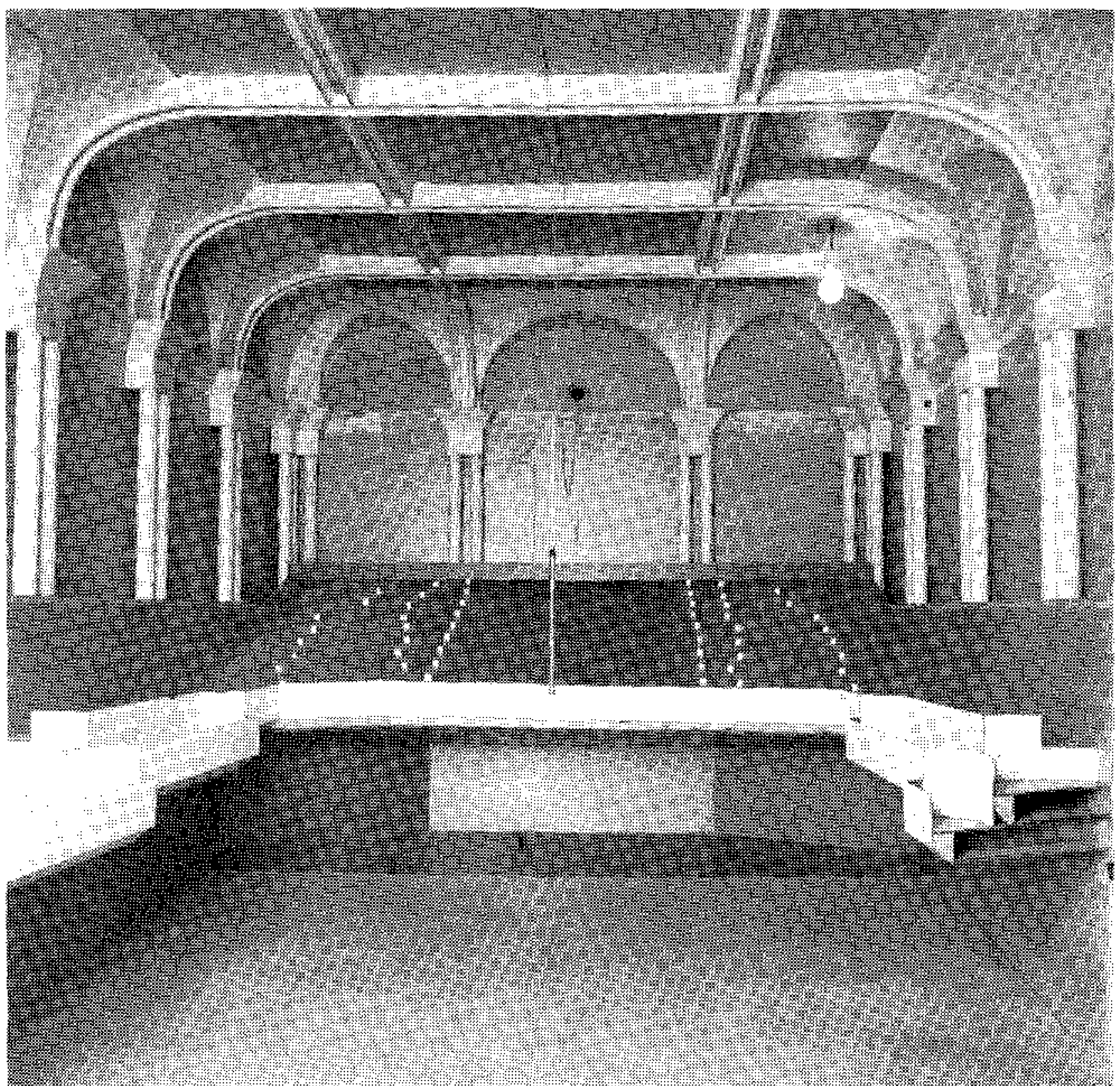



Fig. 1. The 1:10 scale model of Dublin Concert Hall, viewed from the stage. Left: with audience. Right: without audience

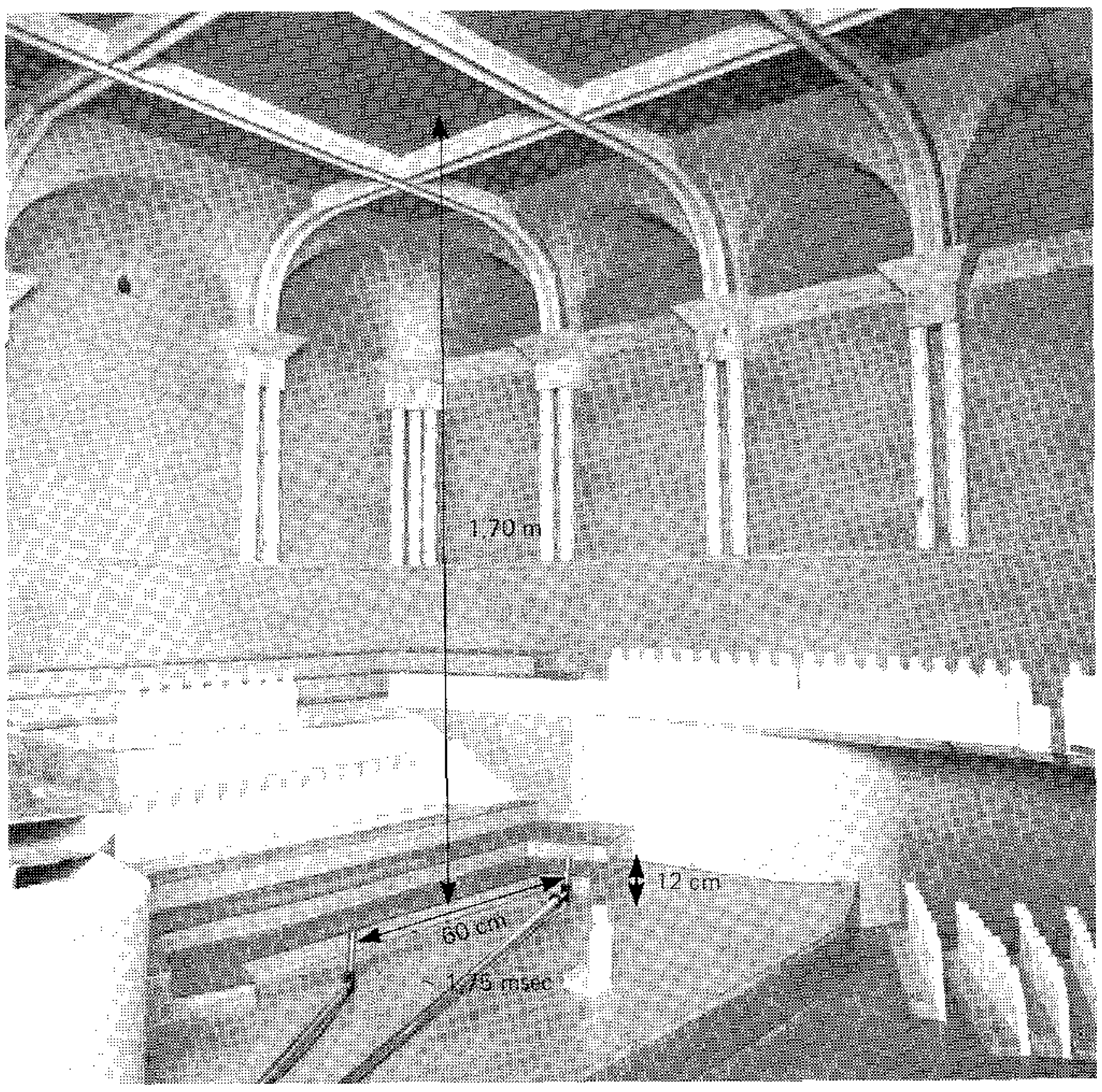


Fig. 2a. Sound transmission on stage. No reflector over stage. Measurement set-up

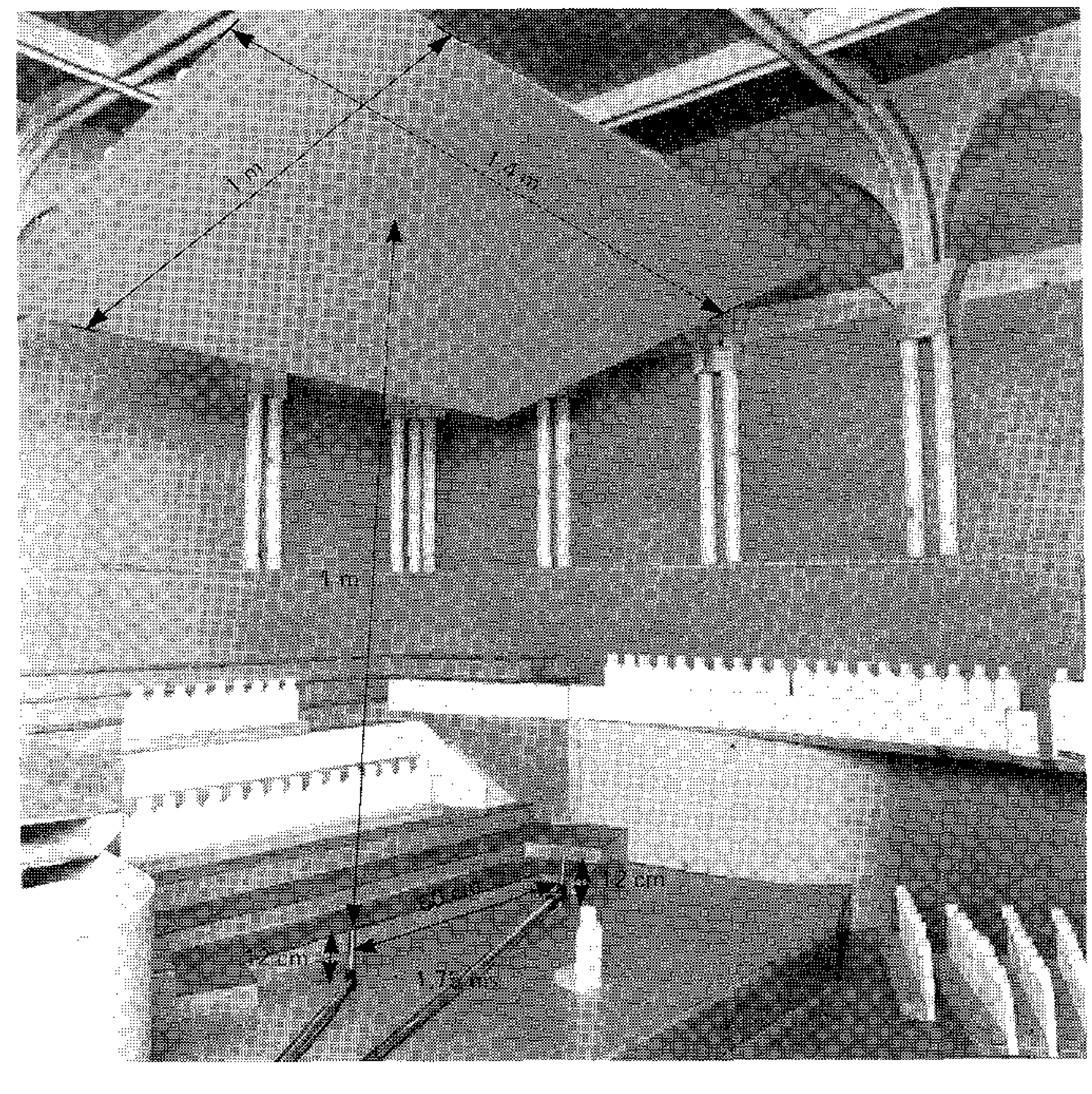


Fig. 3a. Sound transmission on stage. With reflector over stage. Measurement set-up

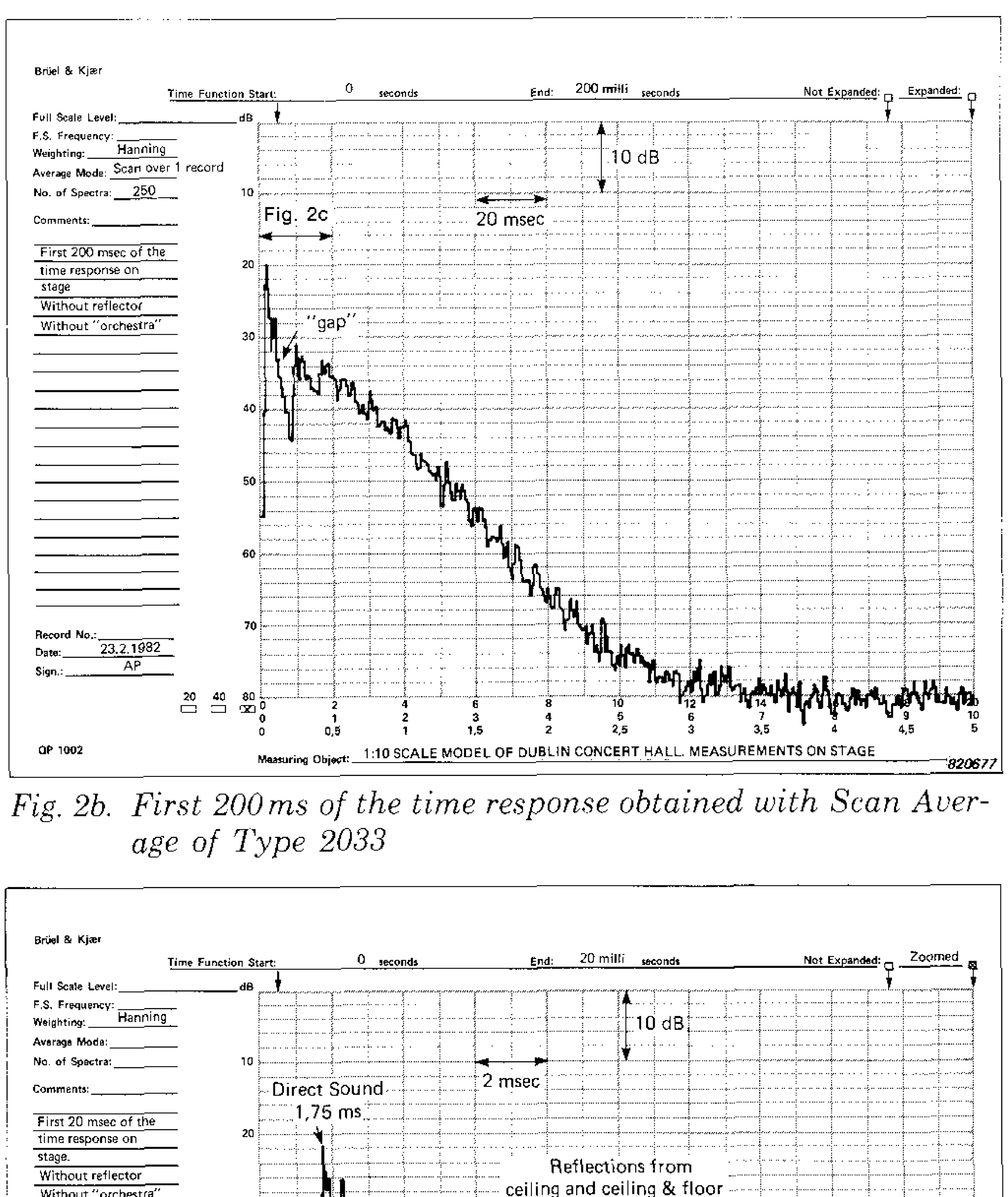


Fig. 2b. First 200 ms of the time response obtained with Scan Average of Type 2033

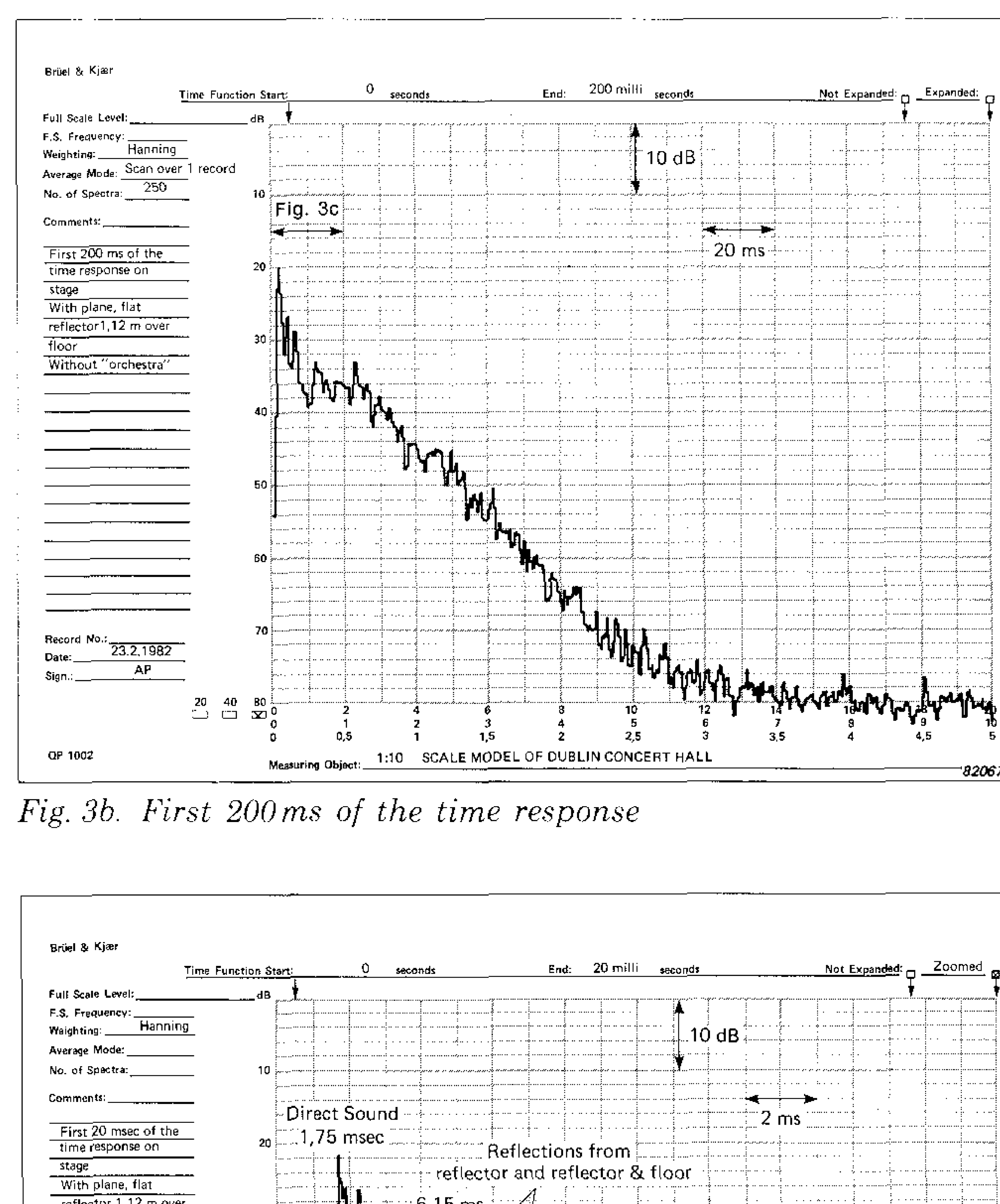


Fig. 3b. First 200 ms of the time response

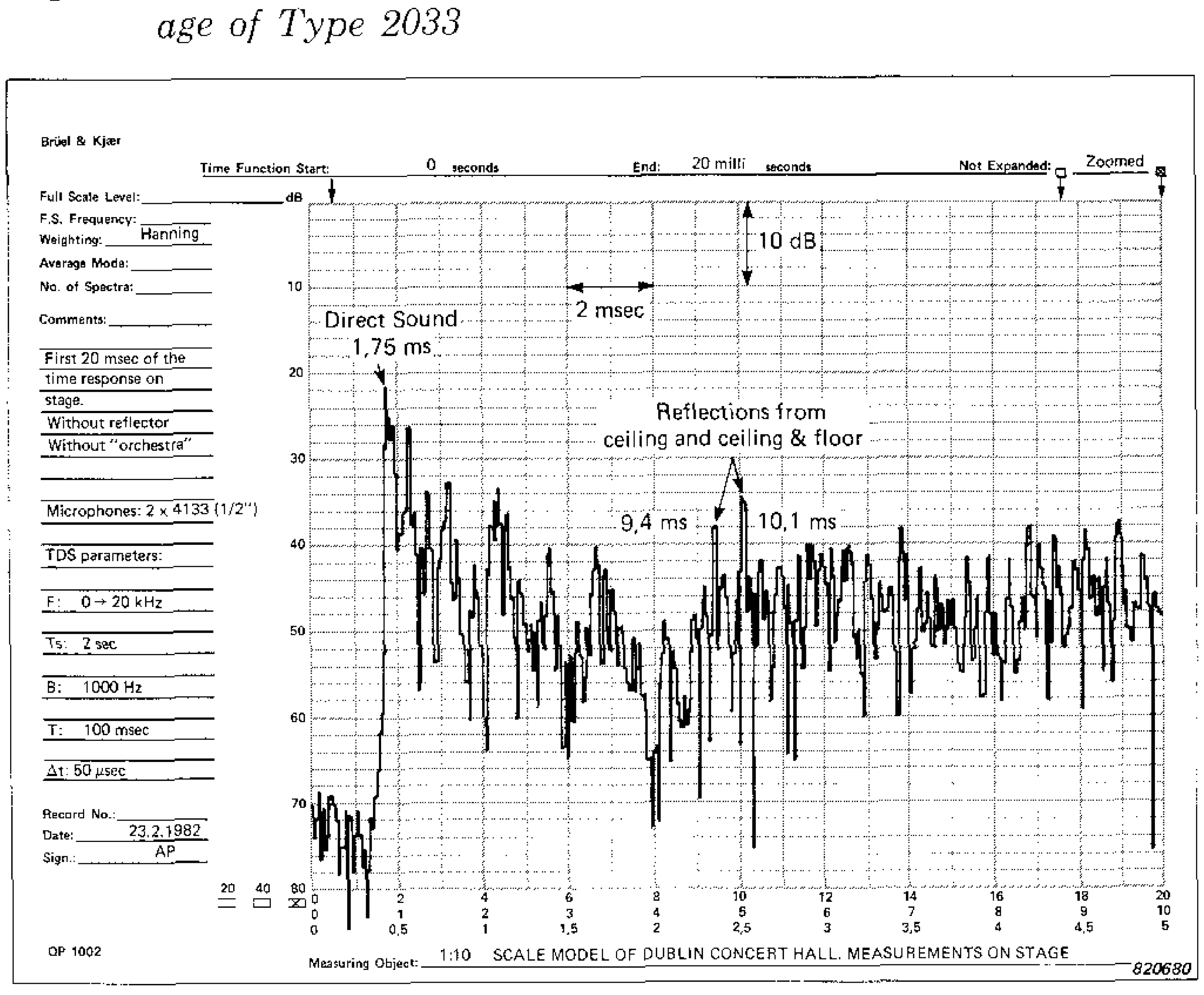


Fig. 2c. First 20 ms of the time response (10 × zoom of Type 2033)

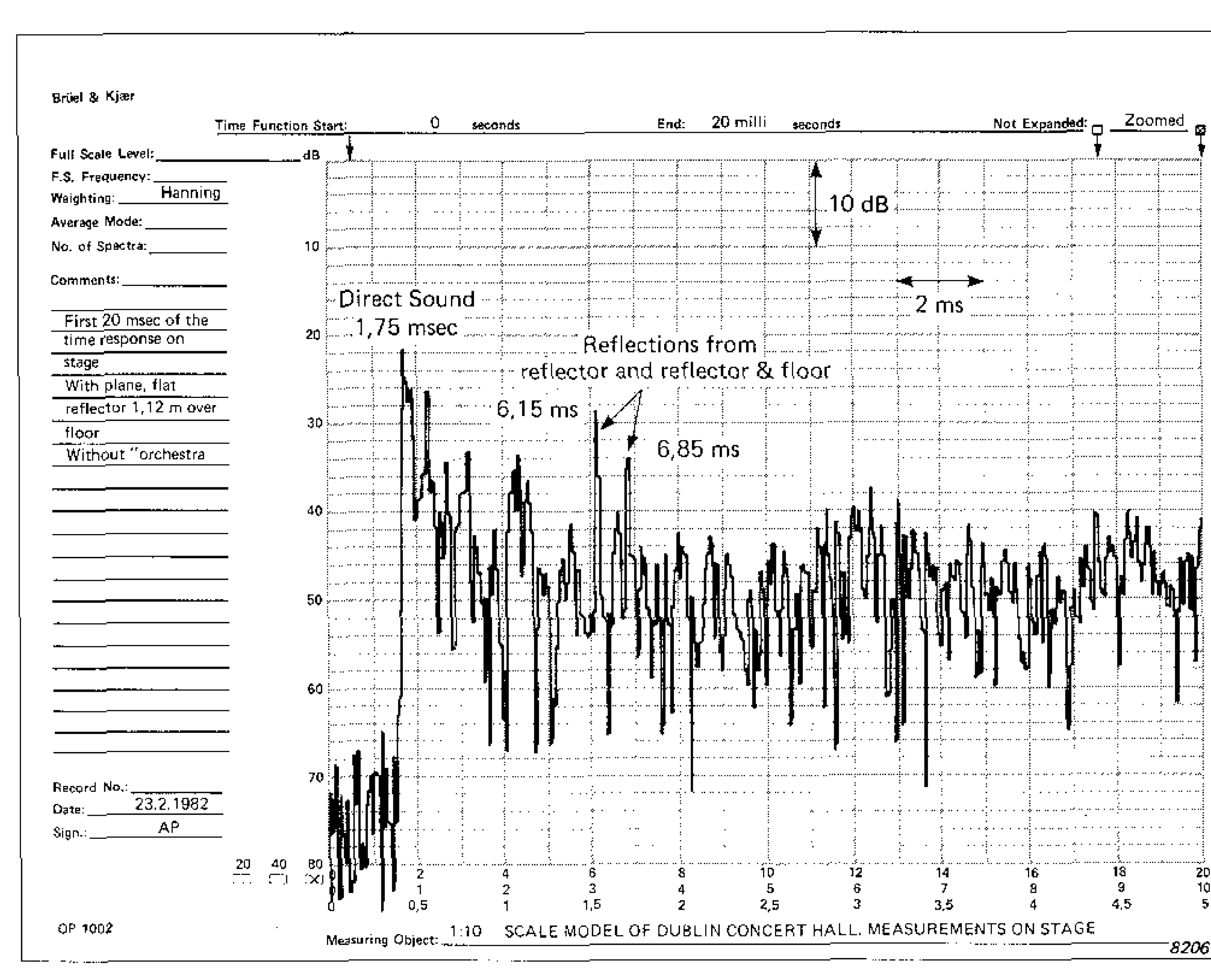


Fig. 3c. First 20 ms of the time response
Measurements 3b & 3c were made with a 1 m × 1,4 m flat reflector, 1,12 m over the stage i.e. 1 m over the microphones

A transmitter and a receiver microphone (both 4133, ½ inch) were placed on the empty stage, 60 cm from one another, both at a height of 12 cm. This corresponds to the elevation of the head of a person seated in a chair (i.e. 120 cm). The frequency range swept was from 0 to 20 kHz. This corresponds to 0 to 2 kHz range in a full scale hall and agrees well with the rule of thumb that when evaluating the acoustics of a full scale concert hall at least the four main octaves 250 Hz to 2 kHz should be included in the spectrum [2].

The rods supporting the microphones shown in Fig. 2, were found to be too big and were replaced for the measurement by acoustically more transparent stands made of wire. Furthermore the microphones were tilted 45° towards one another to obtain a subjectively correct balance between the emphasis on the direct sound relative to the diffuse sound.

The results of the measurement (Fig.2) as seen on the 400-line screen of the FFT Analyzer Type 2033, yield very interesting information. The magnitude of the time response (i.e. the energy time curve or ETC or reflectogram) Fig.2b clearly shows a somewhat unhealthy picture with a very pronounced "gap" in the total sound received after approximately 5 ms. In Fig.2c the 10 × zoom facility of the FFT Analyzer Type 2033 was employed to obtain a picture of the first 20 ms of Fig.2b. The picture shows: a) direct sound, then b) a few rapidly decaying early reflections then c) after a delay of approximately 8 ms the room's response arrives with particularly strong reflections from the ceiling and ceiling/floor. Note that all the time intervals given in the figures refere to the 1:10 scale model. Therefore 8 ms in the model corresponds to 80 ms at full scale.

It is well known [3], that a reflection arriving after a delay of approximately 70 ms or more will be perceived by the human ear as an echo provided that the level is high enough relative to the direct sound. In the case shown, the dominant reflection has just about the threshold level and delay to produce an audible discrete echo. When an orchestra is seated in the path of the direct sound, the level of the direct sound will further decrease while leaving the level of the ceiling-reflection unaffected. An audible echo will then certainly be present. With a scale model "orchestra" made of expanded polystyrene figures placed between the microphones, a 7 dB attenuation of the direct sound was measured (Fig.4.)

By advancing the ceiling-reflection in time towards the direct sound the pronounced gap in Fig.2 will to a certain degree be "filled out" and the hard reflection will effectively merge into a reinforcement of the direct sound, instead of acting as an echo (a phenomenon known as the Haas effect).

This straightforward remedy was investigated by lowering the ceiling by means of a hard reflector suspended 1,12 m above the scale model-stage (Fig.3). The gap is now much less pronounced than in Fig.2.

The convenient representation of the time response in the above figures is worthy of mention. This advantage is a direct result of the TDS method.

TDS yields the *magnitude of the complex time response* rather than the impulse response itself. References [4] and [5] show that the magnitude of the time response is simply the envelope of the impulse response and is thus often referred to as the Energy Time Curve (ETC).

In contrast to the impulse response itself, the ETC is much easier to interpret as the picture is not confused by oscillations at the various peaks in the frequency response. Furthermore, the logarithmic amplitude scale, not applicable to the impulse response, improves the dynamic range visually and exponential decays appear as straight lines. Note also the signal to noise ratio of 45 dB to the direct sound and the repeatability and stability of the set-up testified by the simularity between Figs.2c & 3c.

Measurements of absorbers and reflectors

TDS is a time selective measuring technique which enables measurements to be performed on sound reflected from a surface of interest while rejecting the direct sound as well as undesired reflections. Measurements on material samples can therefore be performed in situ, in ordinary rooms or even out-doors [6].

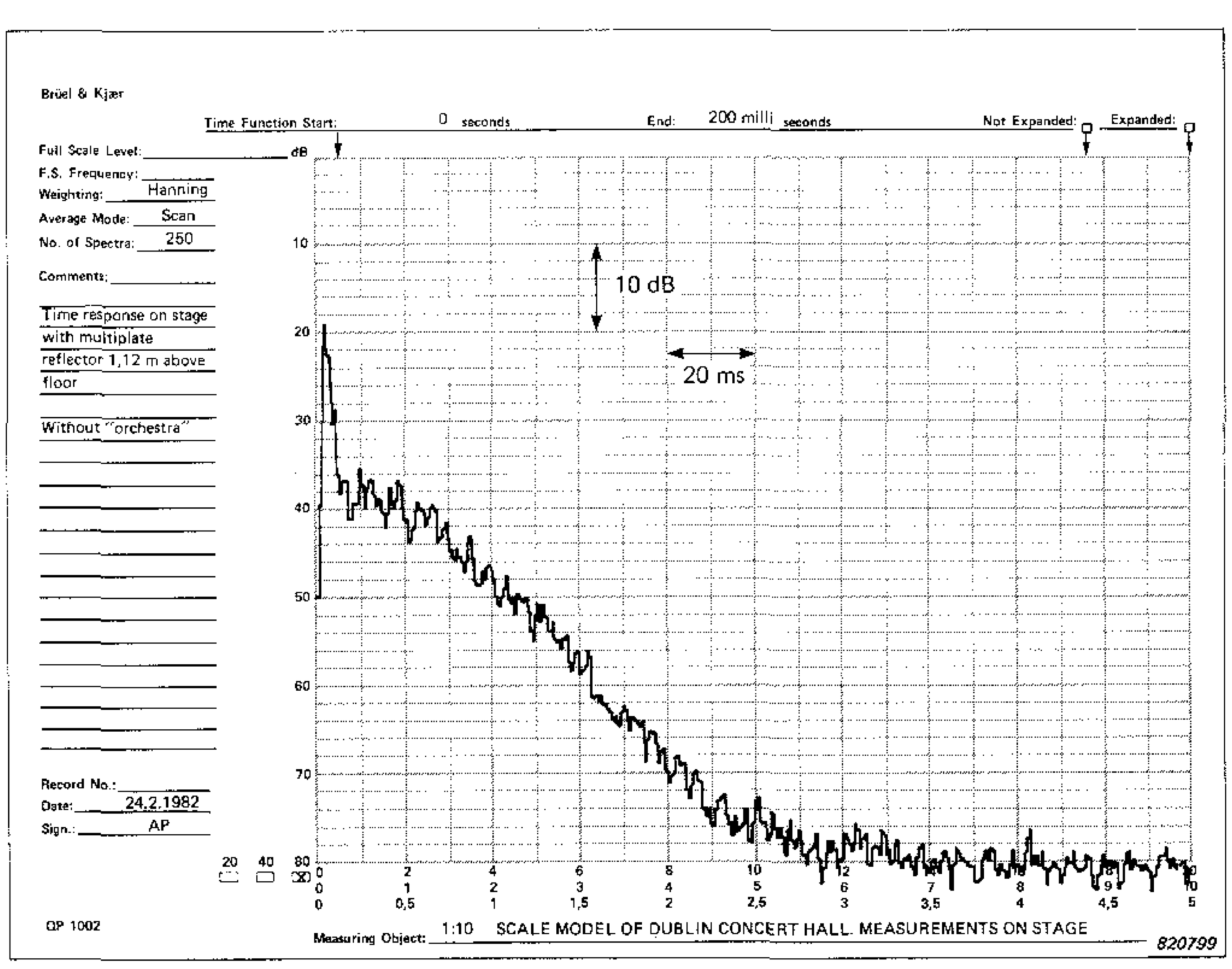


Fig. 4a. Sound transmission on stage, without an "orchestra" of polystyrene figures between the microphones. This measurement was made with 1 m × 1,4 m multi-element reflector, 1,12 m over the stage

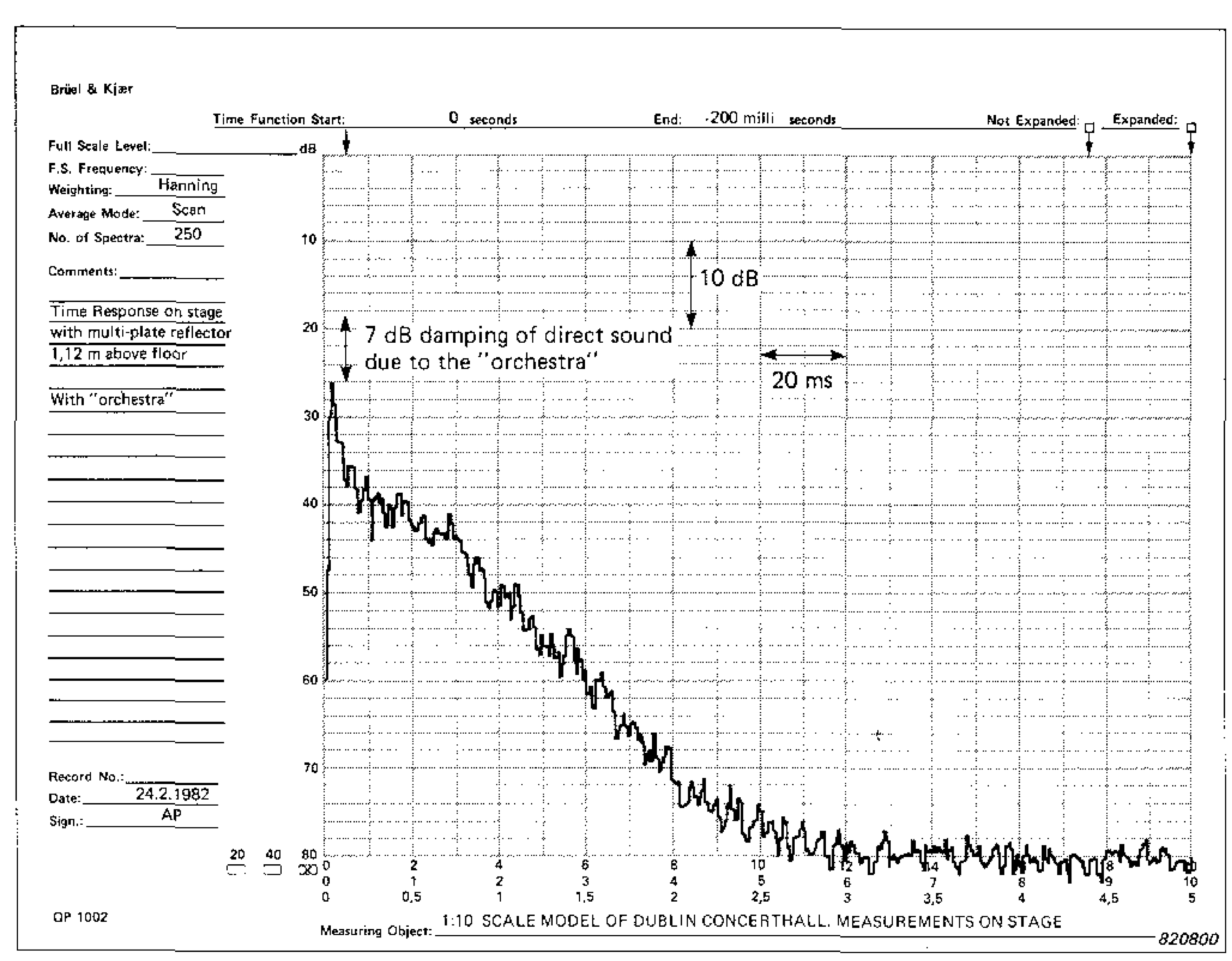


Fig. 4b. As for Fig. 4a but with a full "orchestra" on stage

Fig.5 shows the set-up in the Acoustics Laboratory at the Technical University of Denmark. The amount of sound reflected by an ideal reflector in a certain direction can be regulated by bending the reflector concave (i.e. focussing) or convex (i.e. diffusing). The TDS system measured the level of this reflection using a ½″ 4133 microphone as a sound source. The same distances as in Figs. 2 & 3 were employed.

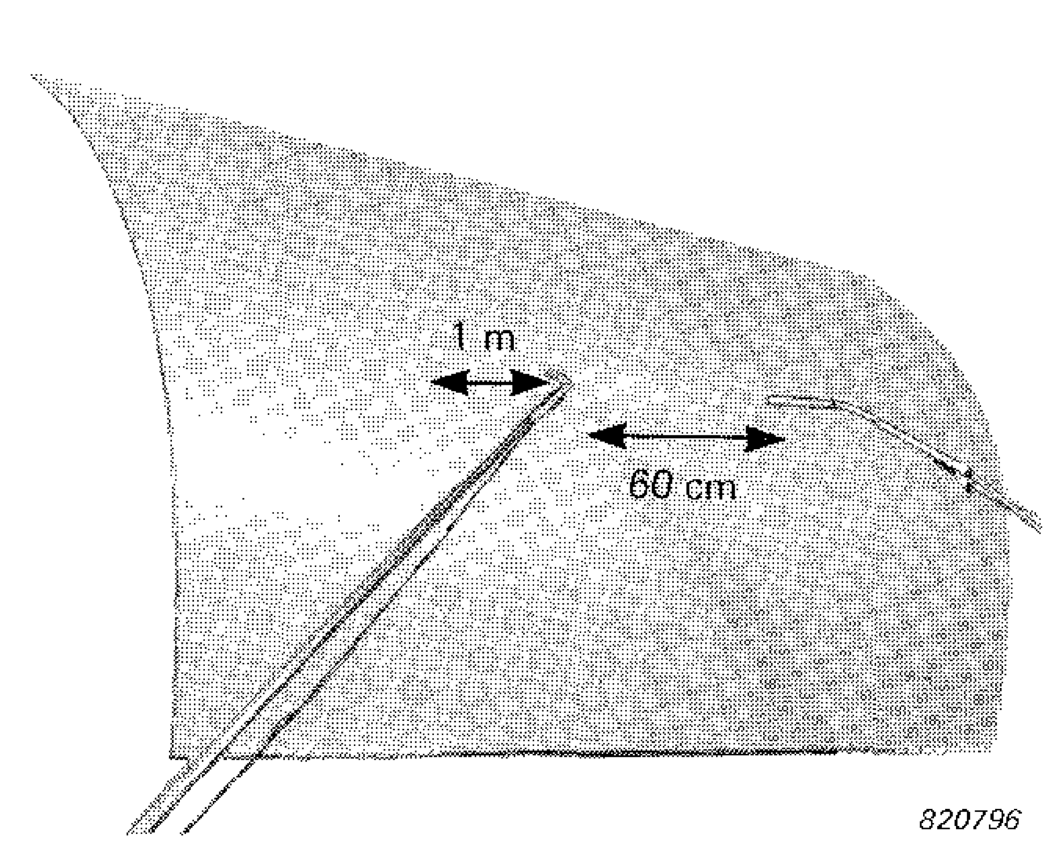


Fig. 6. Convex reflector

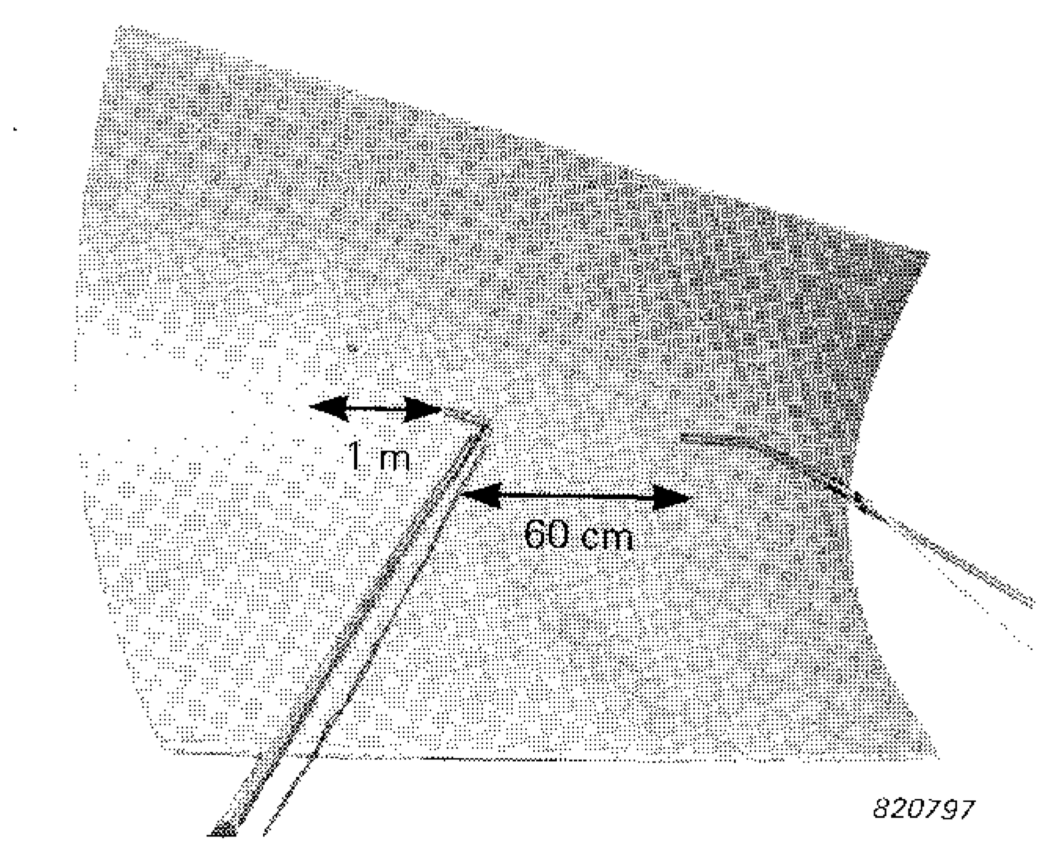


Fig. 7. Concave reflector

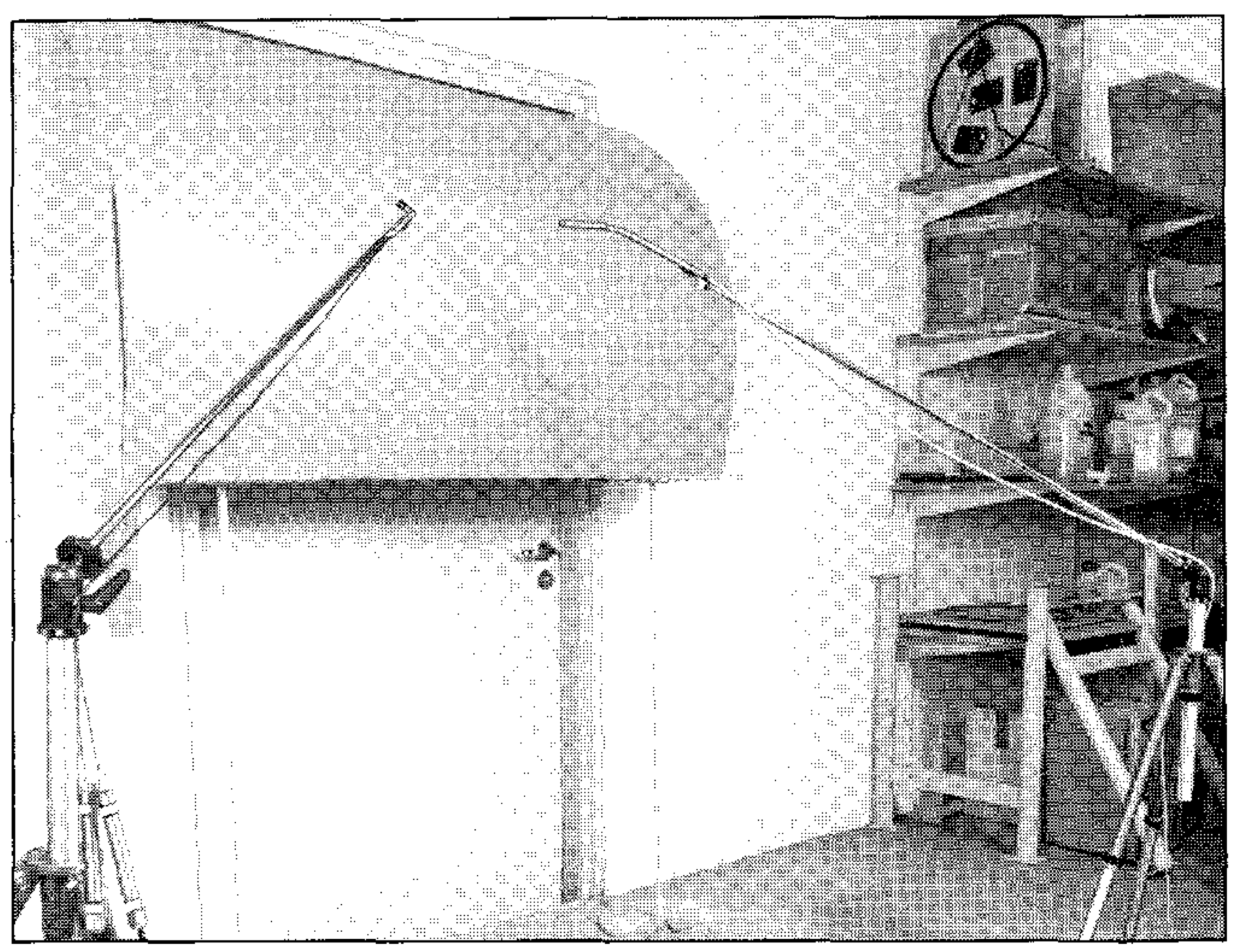


Fig. 5. The measurement set-up for evaluating differently shaped reflectors. Although the acoustic environment is far from anechoic, the TDS system enabled meaningful measurements to be performed

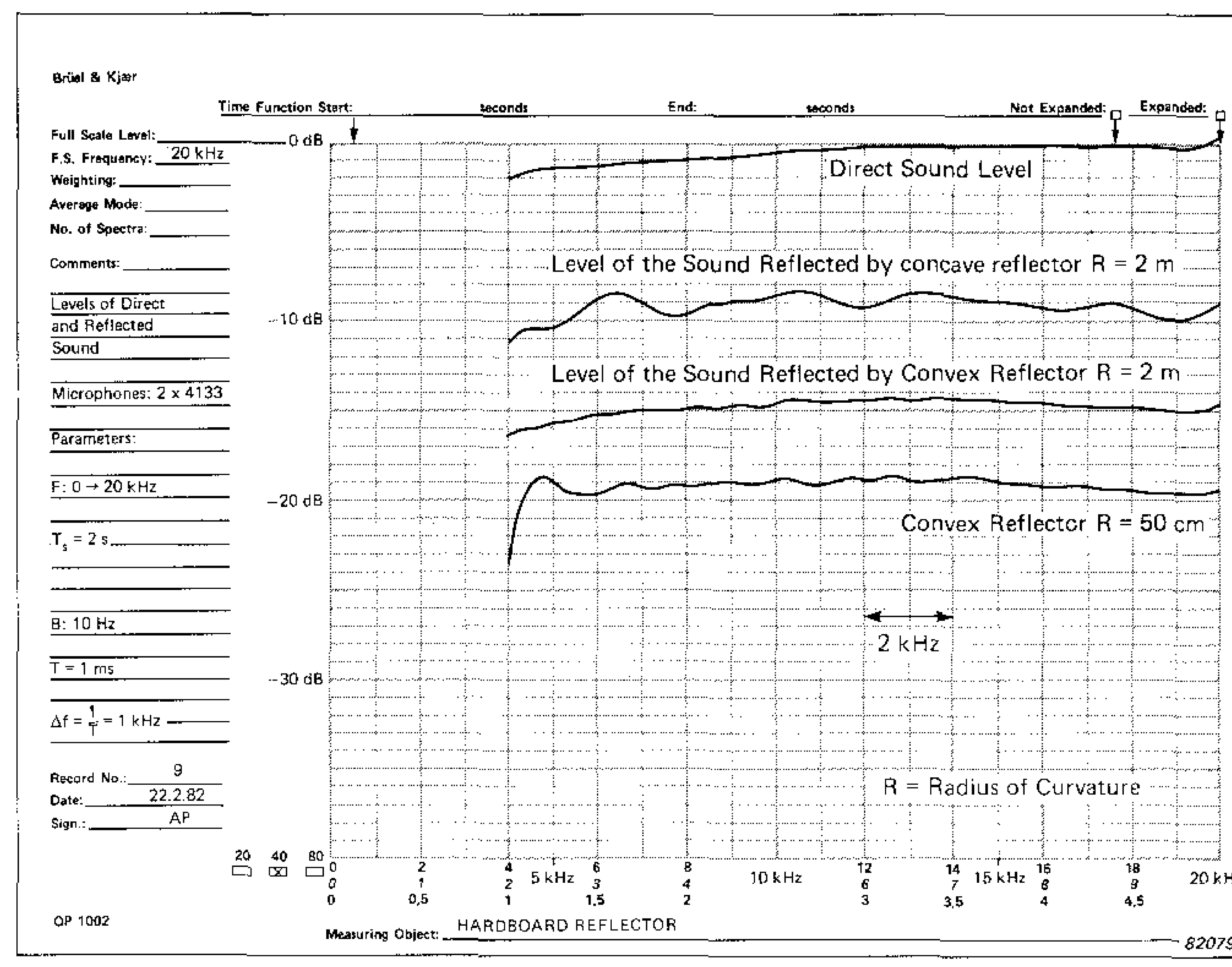


Fig. 8. Some measurement results obtained with concave and convex reflectors

The results are compared in [7] with the theoretical values obtained from a mathematical model for sound reflection from cylindrical surfaces. Here it need only be stated that the agreement was within a fraction of a dB!

An experimental evaluation of this computation with any method other than TDS would be very tedious. With TDS the only requirement is that there is a time delay, T, at the receiving microphone, between the arrival of the direct sound and the reflection of interest. The frequency response of the reflection can then be measured with a resolution of $\Delta f \sim 1/T$. This is not a limitation of the TDS technique but merely a statement of Heisenberg's Uncertainty Principle.

Instrumentation

Time Delay Spectrometry

Time Delay Spectrometry (TDS) is a general two-port measuring method allowing time selective measurements of frequency response (magnitude and phase) and the magnitude of the time response. For a given measuring time and a given ambient noise level, the method offers a signal to noise ratio superior to all other equivalent measuring techniques [8]. For general description of Time Delay Spectrometry and the TDS Measuring System Type 9550 refer to [5] & [9]. Only the aspects relevant to scale modelling are discussed here.

A spark generator can be used to measure directly the impulse response between two points. TDS employs instead the fact that the impulse response is the inverse Fourier transform of the frequency response. The latter is measured by simultaneously detecting the magnitude and phase on a linear sine sweep. To optimise the signal-to-noise ratio of a particular measurement, as much signal energy as possible must be fed into the system under test in the measuring time available. Therefore a short, amplitude limited impulse of a spark source is the worst possible excitation signal. The excitation signal of TDS, the sine sweep, has the lowest possible crest factor and is applied throughout the entire measuring time, with the result that the energy input into the scale model is a maximum.

Simultaneously this method of excitation allows a tracking filter to be employed on the receiving side resulting in rejection of distortion and noise. The narrower the filter used, the better noise rejection. The price paid for this advantage is a longer measuring time needed to maintain the BT_s product and hence the time- and frequency- resolutions. Note, however, that typical TDS measuring times, T_s, are of the order of 1 to 5 s. All the measurements in this note have been performed with a T_s of 2 s.

The TDS System is able to track the sound source output all the way down in level, significantly below the non-filtered noise level. TDS thus constitutes an economical alternative to quietness of an anechoic chamber.

This feature together with the wide frequency range (2 Hz to 200 000 Hz) makes the TDS system eminently suitable for use with condenser microphone sound sources.

Condenser microphone as a sound source

In normal use as a sound receiver, the condenser microphone produces an electrical output proportional to the sound pressure on its diaphragm. For B & K measuring microphones the ratio between pressure applied and voltage produced, the free-field sensitivity, M_f, is uniform throughout the frequency range of the microphone.

A condenser microphone is, however, a reciprocal device and thus is capable of operating in reverse where, by the rules of reciprocity, the ratio between input current applied and the volume velocity produced then also equals the free field sensitivity, M_f. To find the pressure produced for a voltage applied the output and input impedances of the microphone must be known. For free field conditions (see [10] for other acoustical loads) these impedances are well defined and are almost purely reactive. Hence:

$$p \sim M_f \pi \rho_0 U C f^2 / d \quad \text{(Pa)} \qquad \text{Eqn.1}$$

where

M_f = free field sensitivity (V/Pa). For axial direction its value may be found on the individual calibration chart

ρ_0 = density of air (1,2 kg/m^3 at 20°C and 1013 mbar)

U = AC voltage applied (V)

f = frequency (Hz)

C = the electrical capacitance of the microphone (F)

d = distance from source microphone (m)

From eqn.1, it is seen that apart from a +40 dB/decade slope (due to the f^2 term) the relation between pressure and voltage is still governed by the almost constant, free-field sensitivity M_f. Thus, compensating the slope by inserting a –40 dB/decade filter in the signal path effectively yields a sound source with uniform frequency response which is fully equivalent to that of the receiving microphone (Fig.9 & 10).

It is reasonable to assume that the airborne transmission path is linear, therefore the compensation filter can be placed anywhere along the signal path. This should then be done on the receiving side of the set-up, to ensure maximum acoustical output into the test object.

Below some frequency, f_{min}, (which depends on the sound source microphone used, low frequency ambient noise level and TDS-settings), the acoustical output from the microphone becomes no longer traceable and the filter is made to roll off in order not to amplify the ambient noise. The Microphone Compensation Filter WB 0504 has 3 compensation filters which have cut-off frequencies of 500 Hz, 3 kHz and 15 kHz. These values were found optimal for the ½, ¼ and ⅛ inch microphones (Types 4133, 4135, 4138) respectively, when used in environments where the background noise levels were from about 40 to 60 dB(A) and with measuring times up to 5 s. For scale model measurements in a quiet environment (e.g. an anechoic chamber) or with extended measuring times somewhat lower cut-off frequencies may be employed, if necessary.

With at least 20 dB signal-to-noise ratio at the low frequencies and standard compensation-filter values, the following flat frequency response-ranges were measured:

Type	Size	Range
4133	1/2″	1 kHz – 40 kHz (± 1 dB. Fig.8)
4135	1/4″	6 kHz – 100 kHz (± 1,5 dB)
4138	1/8″	30 kHz – 160 kHz (± 2 dB)

(Drive voltage: 200 Vpp equivalent to 70 V rms, DC-polarisation voltage: 200 V, Measuring distance: 20 cm, Measuring microphone: same type as source microphone, Tracking filter: 10 Hz , Measuring time: 2 sec. Note that for best results the 1/2 inch microphones should be used with protection grid on, whereas smaller microphones should be used without grid.)

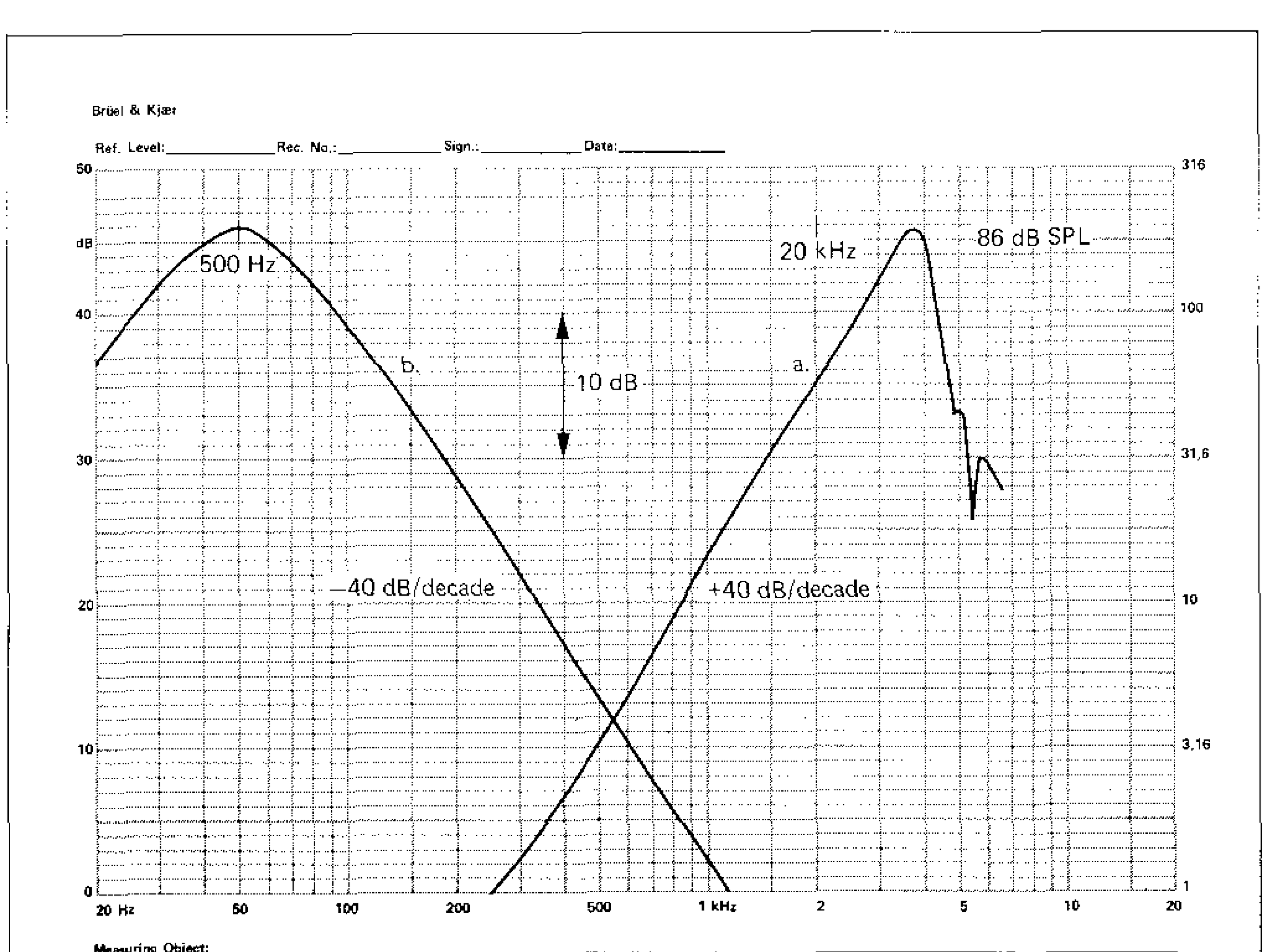


Fig. 9. a) The sound pressure level emitted by a ½″ Microphone Type 4133 when driven by 200 Vpp AC voltage. Measured at a distance of 20 cm. b) The compensation filter necessary to obtain a flat frequency response

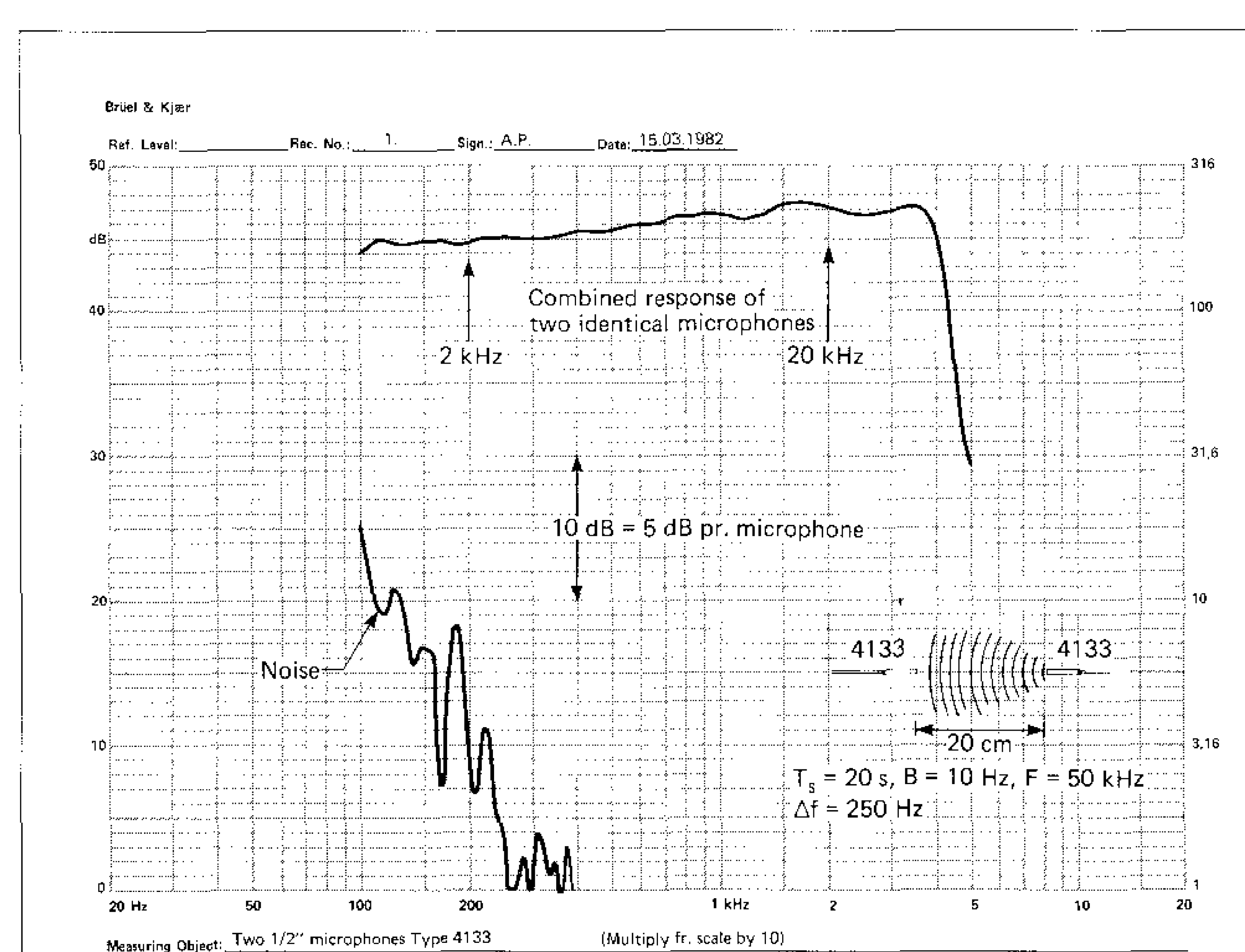


Fig. 10. Total, compensated frequency response of two ½″ Microphones Type 4133 using one as a sound source and the other as a receiver. Deviations from the ideal flat response can be equally attributed to the emitter and the receiver thus making the 10 dB division on the vertical axis equivalent to 5 dB per microphone. For signal to noise evaluation the 10 dB division is valid

Best results are normally obtained using the same type of microphone as transmitter and receiver. Due to reciprocity, the frequency and directional characteristics of the microphone are the same whether it is used as a transmitter or as a receiver [10]. For most practical purposes the microphones are omnidirectional.

The measuring chain

The TDS System Type 9550 together with the auxillaries suggested for scale model measurements are shown in Fig.11.

The Heterodyne Analyser Type 2010 serves as sine sweep generator and has a frequency locked tracking filter section. It possesses a wide frequency range 2 Hz to 200 kHz and is thus eminently suitable for scale model purposes. The TDS Control Unit Type 5842 controls the sine sweep generation, detects the magnitude of the frequency response and then directs and mixes incoming signals for the appropriate analysis. The Narrow Band FFT Analyzer Type 2033 performs the Fourier transform into time domain from the frequency response measurement.

The X-Y Recorder 2308 can be connected to give hard copies of the measurements i.e. frequency vs. magnitude, frequency vs. phase or time response.

The Phase Meter Type 2971 measures the phase of the frequency response. The Distortion Control Unit Type 1902 serves as a delay compensation. The 2 latter instruments may be omitted if only the time response is of interest.

The above mentioned instruments comprise the standard 9550 TDS Measuring System. Normally a storage oscilloscope is included for continuous display of the frequency response (magnitude and phase).

The left hand side of Fig.11 shows the special equipment required for scale model measurements. The Octave Filter Type 1617 can be included e.g. for measurement of reverberation time in octave and third-octave bands (up to 200 kHz). The Microphone Compensation Filter Box WB 0504 is capable of amplifying the preamplified microphone signal up to 1 V rms which is the nominal input level for the Filter Type 1617.

The high voltage Power Amplifier Type 2713 has a frequency range of 10 Hz to 200 kHz and is capable of delivering the maximally allowed 200 V pp into the purely capacitive load presented by the transmitter microphone. This limit is valid for the Microphones Types 4133, 4135 & 4138 with 200 V DC polarisation. See [10] for other types. Finally a small but important detail is the Microphone Sender Adaptor WA 0160 which enables the DC polarisation voltage from the rear of the 2010 and the AC drive voltage to be connected to the diaphragm of the transmitter microphone. For places where access is difficult the flexible connectors UA 0122, UA 0123 or UA 0196 between the microphone capsule and the preamplifier or the sender adaptor may prove useful.

Additional data processing with the TDS System

Once the time response is measured, all the other types of room acoustical responses (e.g. steady state, reverberation, the build-up of sound) as well as the large number of single-figure room acoustical criteria (e.g. EDT, T_{60}, Clarity, Rise-time [2], [11]) can be either read directly off the 2033 FFT analyzer's screen or a copy of the screen drawn by 2308 X-Y recorder or calculated by a computer from the digitally stored time response data inside the 2033. See [12] for some useful algorithms.

A multitude of postprocessing routines such as averaging, comparison with prerecorded reference data, 10 × zoom, scan and scan average is available in 2033 without the need for a computer. The scan feature enables a time response to be calculated based on only 10% of the frequency range. Thus it is possible to scan through the frequency domain while viewing the corresponding time responses.

As previously mentioned, the frequency response (magnitude and phase) of a reflecting surface can be assessed using the time-window of TDS to reject the direct sound and other non-desired reflections from a measurement. In the time domain, with the time window wide open, the TDS System allows the viewed time response to be compensated for the damping due to distance. The average damping coefficient of a particular reflector can hence be read directly from the screen of the 2033 as the difference in level between the direct and the reflected sound. With the scan feature of the 2033 the variation of the damping coefficient as a function of frequency can be roughly monitored.

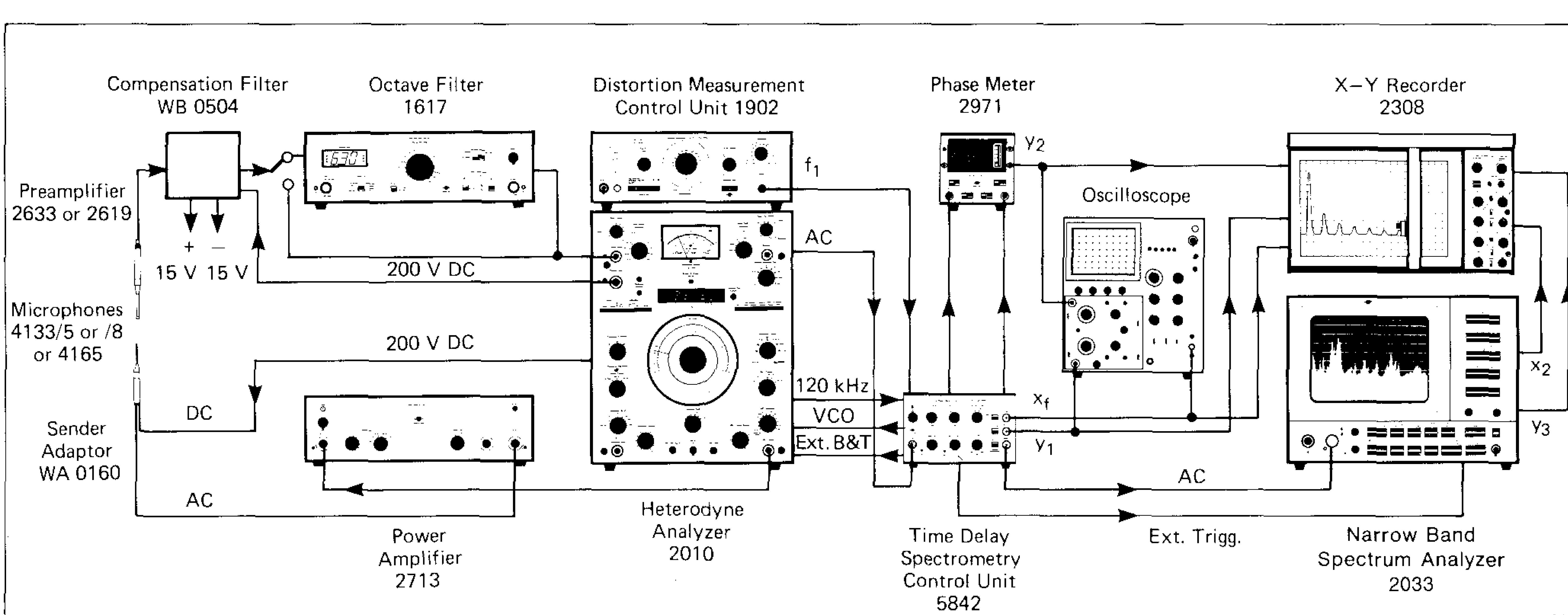


Fig. 11. Complete TDS System Type 9550 plus auxiliaries for scale model measurements

Conclusion

The Time Delay Spectrometry System Type 9550, enables the potential of condenser microphones used as sound sources to be utilised to the full.

The accuracy and stability of the set-up as well as the convenient presentation of result data, makes it possible to pin-point each individual reflection in a strucure as complex as a scale model of a concert hall.

The time selectivity of TDS enables the reflective properties of materials to be measured in an accurate and well-defined manner. The complex damping coefficient of a surface can be measured with only moderate requirements to the surroundings.

The TDS system is highly flexible system and may be used to measure electroacoustical, electrical or mechanical devices. TDS is in no way restricted to acoustical measurements. The only modification needed is the connection of the appropriate transducers (e.g. accelerometers, hydrophones, microphones) to the input and output terminals of the TDS system.

Acknowledgment
My thanks are due to Jens Holger Rindel of the Acoustics Laboratory, Technical University of Denmark for his assistance in preparing and effectuating the measurements.

References

1. V.L.Jordan, "Acoustical criteria for concert hall stages", Applied Acoustics, vol.15, 1982, pp.321-328.

2. V.L. Jordan "A group of objective acoustical criteria for concert halls", Applied Acoustics vol. 14, no.4, 1981, pp.253-266.

3. Leo L. Beranek "Acoustics", Mcgraw-Hill Book Company, 1954.

4. Richard C. Heyser, "Determination of Loudspeaker Signal Arrival Times" (in three parts). J.A.E.S. Vol.19, October-December 1971.

5. Richard C. Heyser "Concepts in the frequency and time domain response of loudspeakers", Monitor - Proc. IREE March 1976.

6. Cecil R. Cable and John K. Hilliard, "The practical application of Time-Delay Spectrometry in the field", Journal of AES vol. 28, no.5, pp.302-309.
F.J Fahy "Comments on *"The practical application of Time-Delay Spectrometry in the field"* and author's reply", Journal of AES vol. 29, no. 4, pp.262-264.

7. J.H. Rindel, "Lydrefleksion fra konvekse og konkave cylinderflader" (in Danish). Paper presented at the Scandinavian Acoustic Meeting, NAS-82, Stockholm, 1982.

8. Henrik Biering & Ole Zacho Pedersen "Free-field techniques. A comparison of time selective free-field techniques with respect to frequency resolution, measuring time, and noise-induced errors." Paper presented at 68th A.E.S. convention, Hamburg 1981. Available from B&K, Denmark.

9. "Time Delay Spectrometry Control Unit Type 5842", B&K System Development note.

10. Erling Frederiksen "Condenser microphones used as sound sources", B&K-Technical Review No.3, 1977.

11. P. Lehmann & H. Wilkens "Zusammenhang subjektiver Beurteilungen von Konzertsaelen mit raumakustischen Kriterien", Acustica vol.45, 1980.

12. A. Kulowski "Relationship between impulse response and other types of room acoustical responses", Applied Acoustics vol. 15, 1982, pp.4-10.

13. Jens Hee, "Use of the High resolution Sound Analyzer Type 2033 with a Desk-Top Calculator", B & K Application Note.

14. Don Davis and Chips Davis, "The LEDE^{TM} Concept for the control of Acoustic and Psychoacoustic Parameters in Recording Control Rooms". J.A.E.S. Vol.28, no.9 p.585.

15. Cecil R. Cable "Time Delay Spectrometry investigations of regenerative sound systems", Journal of AES vol. 26, no. 3, pp.114-119.

16. "TEF^{TM} and Acoustic Models", Syn-Aud-Con. Tech Topics, Vol.9, no.4, p.7.

Brüel & Kjær
DK-2850 NÆRUM, DENMARK
Telephone: + 45 2 80 05 00
TELEX: 37316 bruka dk

Printed in Denmark by Nærum Offset