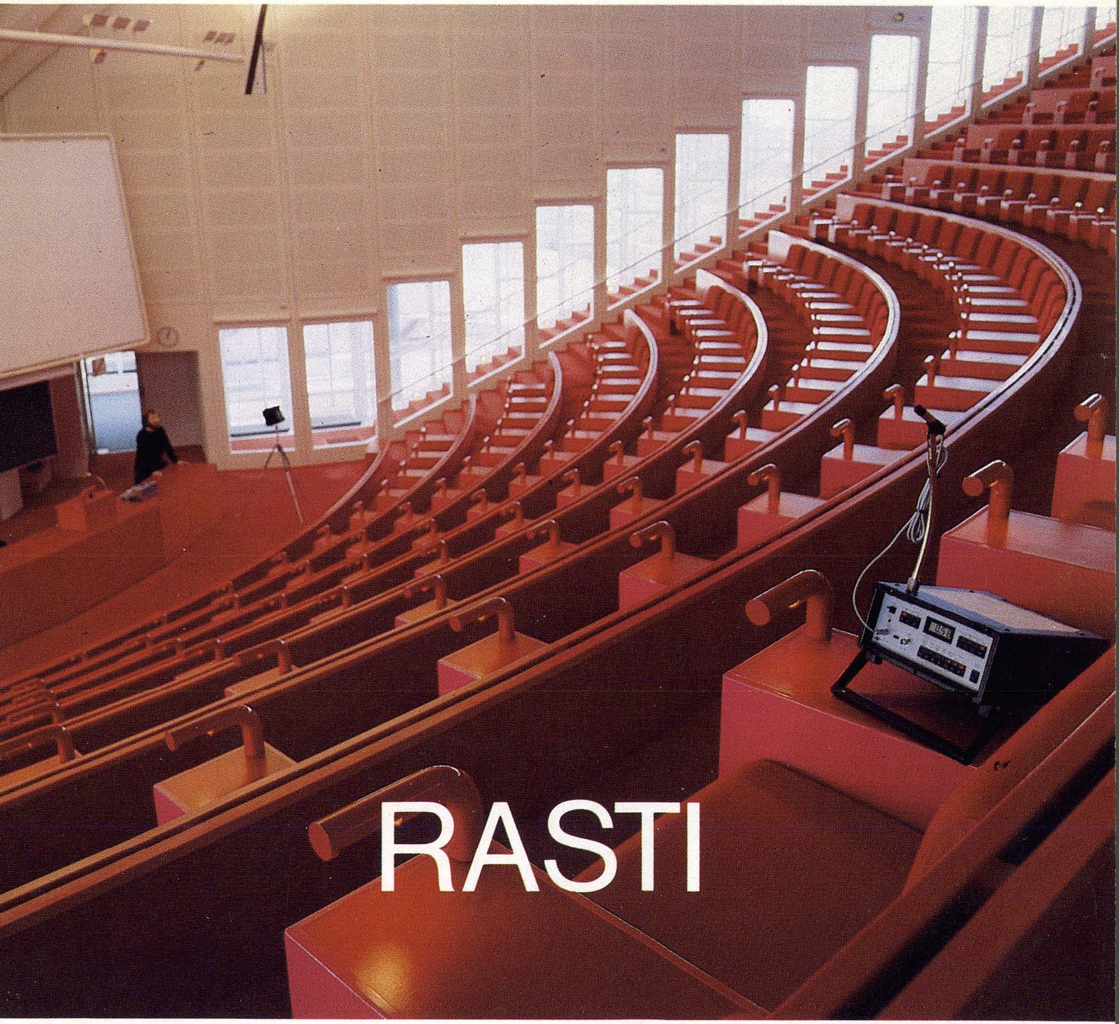
ISSN 007-2621

No. 3 — 1985

published quarterly

Technical Review

To Advance Techniques in Acoustical, Electrical and Mechanical Measurement



RASTI

Brüel & Kjær

PREVIOUSLY ISSUED NUMBERS OF
BRÜEL & KJÆR TECHNICAL REVIEW

2-1985 Heat Stress
A New Thermal Anemometer Probe for Indoor Air Velocity Measurements
1-1985 Local Thermal Discomfort
4-1984 Methods for the Calculation of Contrast
Proper Use of Weighting Functions for Impact Testing
Computer Data Acquisition from B & K Digital Frequency Analyzers 2131 / 2134 using their Memory as a Buffer
3-1984 The Hilbert Transform
Microphone System for Extremely Low Sound Levels
Averaging Times of Level Recorder 2317
2-1984 Dual Channel FFT Analysis (Part II)
1-1984 Dual Channel FFT Analysis (Part I)
4-1983 Sound Level Meters – The Atlantic Divide
Design principles for Integrating Sound Level Meters
3-1983 Fourier Analysis of Surface Roughness
2-1983 System Analysis and Time Delay Spectrometry (Part II)
1-1983 System Analysis and Time Delay Spectrometry (Part I)
4-1982 Sound Intensity (Part II Instrumentation and Applications)
Flutter Compensation of Tape Recorded Signals for Narrow Band Analysis
3-1982 Sound Intensity (Part I Theory).
2-1982 Thermal Comfort.
1-1982 Human Body Vibration Exposure and its Measurement.
4-1981 Low Frequency Calibration of Acoustical Measurement Systems.
Calibration and Standards. Vibration and Shock Measurements.
3-1981 Cepstrum Analysis.
2-1981 Acoustic Emission Source Location in Theory and in Practice.
1-1981 The Fundamentals of Industrial Balancing Machines and their Applications.
4-1980 Selection and Use of Microphones for Engine and Aircraft Noise Measurements.
3-1980 Power Based Measurements of Sound Insulation.
Acoustical Measurement of Auditory Tube Opening.
2-1980 Zoom-FFT.
1-1980 Luminance Contrast Measurement.
4-1979 Prepolarized Condenser Microphones for Measurement Purposes.
Impulse Analysis using a Real-Time Digital Filter Analyzer.

(Continued on cover page 3)

TECHNICAL REVIEW

No. 3 — 1985

Contents

THE MODULATION TRANSFER FUNCTION IN ROOM ACOUSTICS

by

*T. Houtgast**
and
*H.J.M. Steeneken**

ABSTRACT

In many cases, the sound transmission from a speaker to a listener is not perfect, which may result in reduced speech intelligibility. The performance of such a sound transmission system can be quantified by the Modulation Transfer Function: the extent to which the fluctuations in the original signal are preserved in the signal reaching the listener. An illustration is given of the way in which an MTF analysis can be performed and, additionally, how such data are converted into an index which quantifies the effect of a sound transmission system on speech intelligibility.

SOMMAIRE

Dans beaucoup de cas, la transmission du son depuis le locuteur jusqu'à l'auditeur n'est pas parfaite et il en résulte une réduction de l'intelligibilité. L'efficacité du système de transmission sonore peut être représentée par la Fonction de Transfert de Modulation, qui quantifie comment les fluctations du signal d'origine sont conservées dans celui atteignant l'auditeur. L'article illustre comment une analyse FTM peut être effectuée et comment les données obtenues peuvent être converties en un indice quantifiant les effets d'un système de transmission sonore sur l'intelligibilité.

ZUSAMMENFASSUNG

In vielen Fällen ist die Schallübertragung zwischen Sprecher und Zuhörer, und damit die Sprachverständlichkeit, beeinträchtigt. Die Qualität eines Sprachübertragungssystems läßt sich anhand der MTF (**M**odulation **T**ransfer **F**unction — Modulationsübertragungs-Funktion) beurteilen: Sie gibt an, in welchem Grad die Fluktuation des ursprünglichen Signals noch den Hörer erreicht. Die Abhandlung illustriert die Durchführung der MTF-Analyse und zeigt darüberhinaus, wie sich die gewonnenen Daten in einen Index überführen lassen, der die Sprachverständlichkeit eines Schallübertragungssytems beschreibt.

* Institute for Perception TNO; Soesterberg, The Netherlands.

Introduction

When a speaker addresses an audience – be it in a room, in an auditorium or outdoors with a public-address system – the speech signal reaching the listener will be distorted to some degree, which may lead to reduced speech intelligibility. Possible causes are echoes and reverberation, spectral deformation, ambient noise, etc. Thus, in terms of a sound transmission system with the original speech signal produced by the speaker as the input and the sound reaching the listener as the output, the performance of the system is not perfect: in general, the output is not a perfect copy of the input. It will be shown that the (im)perfectness of the input-output relationship of a sound transmission system can be described adequately by the Modulation Transfer Function (MTF), which can be used to quantify the effect on speech intelligibility.

In this paper some general aspects of the application of the MTF concept to sound transmission will be considered. The next paper will be concerned with its implementation in a measuring device and some practical examples.

The Modulation Transfer Function

The rationale underlying the application of the MTF concept in room acoustics has been described in various papers [1, 2, 3]. The MTF quantifies to what extent the modulations in the original signal are reduced, as a function of the modulation frequency. The modulations are defined by the **intensity envelope** of the signal: it is only in the intensity domain, that the interfering noise or reverberation will affect only the degree of modulations of a sine-wave shaped modulation **without** affecting the sine-wave shape. The scheme in Fig.1 illustrates how the MTF may be used to quantify the relation between the original speech signal at the input and the output signal (A or B). Since most disturbances may vary considerably as a function of carrier frequency, the analysis is octave-band specific. The example in Fig.1 considers one octave band only, i.e., the intensity envelopes in the octave band with centre frequency of 500 Hz. Two simple sound transmission systems are illustrated, one with reverberation only (case A; T = 2,5 s) and one with interfering noise only (case B; signal-to-noise ratio $S/N = 0\,\text{dB}$).

In general, the effect of reverberation or ambient noise is a reduction of the (relative) fluctuations in the envelope function, which can be quantified by an **envelope spectrum**. The envelope spectrum results from a 1/3-octave-band analysis of the envelope function (typically of a one-minute speech fragment), and reflects the spectral distribution of the

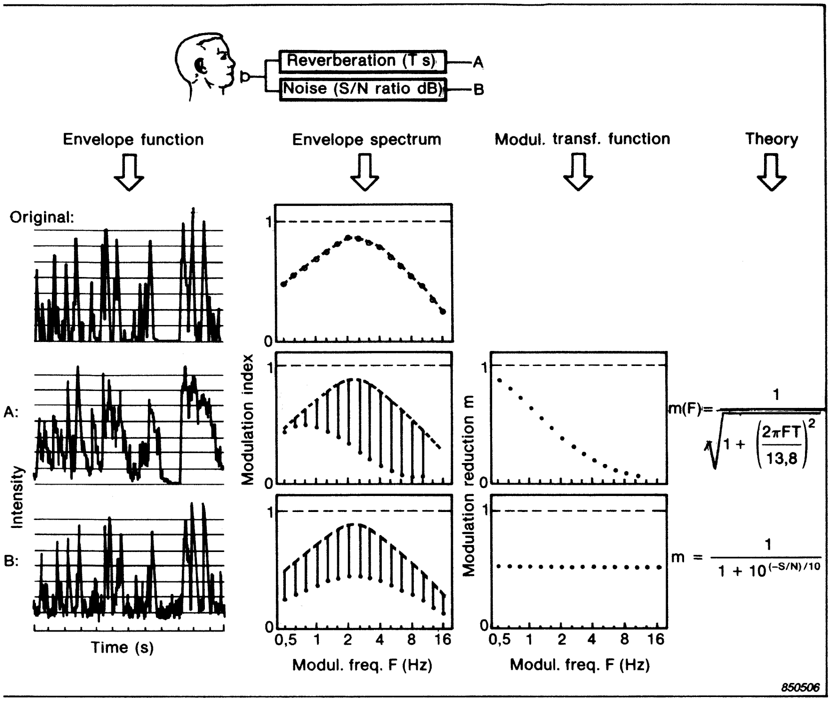


Fig. 1. The reduction of the fluctuations in the (octave-band specific) envelope of an output signal (A or B) relative to the original signal can be expressed by the Modulation Transfer Function. The two conditions considered (reverberation or noise interference), lead to characteristic MTFs, according to the theoretical expressions given at the right-hand side. See text for further explanation

envelope fluctuations relative to the mean intensity: the modulation index as a function of modulation frequency. The difference between the original and the resulting envelope spectrum reflects the reduction in the envelope fluctuations brought about by the sound transmission system. This leads to the MTF: the reduction factor of the modulation index as a function of modulation frequency.

Typically, as may be observed in Fig.1, in the case of reverberation the MTF has the form of a low-pass filter: the faster fluctuations are most sensitive to the effect of reverberation, as is to be expected. In the theoretical case of an ideal exponential reverberation process, the MTF is defined mathematically (see Fig.1), its low-pass character being determined by the product FT (F = modulation frequency, T = reverberation time). In case of noise interference, the MTF is defined by the *S/N* ratio and is independent of modulation frequency: the interfering noise results in an increased mean intensity and thus reduces the (relative) modulation index for all modulation frequencies by the same factor.

It is important to note that the (octave-band specific) MTF of a sound transmission system is **independent** of the input signal considered: it quantifies the modulation transfer for any input signal, be it speech, music or an artificial signal, provided that within that octave band these signals have the same mean intensity (because, for interfering noise, the *S/N* ratio is important).

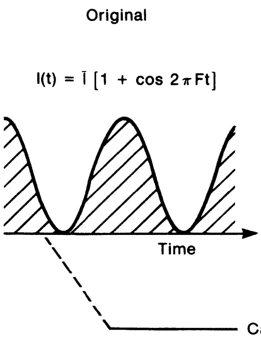
MTF Analysis

The MTF of a sound transmission system can be determined in various ways, the principle always being that the modulation reduction factor is derived from a comparison of the intensity modulations at the output and at the input of the system. One approach, illustrated in Fig.2, uses a specific test signal by which the modulation reduction factor is determined for each modulation frequency successively. This test signal, which is produced at the position of the speaker's mouth, consists of a noise carrier with 100% intensity modulation. The remaining modulation index at a listener's location directly reflects the modulation transfer function for that particular modulation frequency. The noise carrier is octave-band filtered, and the measurements are performed for different centre frequencies (typically from 125 Hz up to 8 kHz). As mentioned before, the mean intensity of the test signal is a critical parameter, and should be related to that of the speech as normally produced by a speaker at that position. As a rule, for each octave band considered, the L_{eq} of the test signal is to be adjusted to the L_{eq} of ongoing speech typical for the condition being tested.

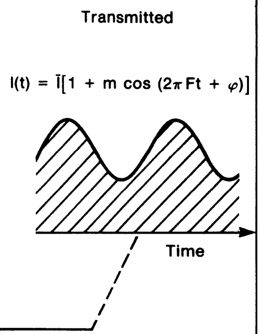
By this analysis, the performance of a sound transmission system is quantified by a family of curves, one curve for each octave band of the noise carrier, and each curve defined by 14 points on the modulation-frequency scale (F values from 0,63 up to 12,5 Hz in 1/3-octave intervals). The example in Fig.2 is a theoretical one, assuming an ideally exponential reverberation process, with T-values and *S/N* ratios as indicated for each of the octave bands. By these data the combined modulation-reduction factor $m(F)$ is determined mathematically by

$$m(F) = \frac{1}{\sqrt{1 + \left[2\pi F \dfrac{T}{13{,}8} \right]^2}} \cdot \frac{1}{1 + 10^{(-S/N)/10}} \qquad (1)$$

This being the product of the two factors for each of these two types of disturbances individually, as given in Fig.1, with T in seconds and S/N ratio in dB.



Oct. band (Hz)	T (s)	S/N (dB)
125	2,0	0
250	2,0	+5
500	2,0	+10
1k	1,4	+10
2k	1,0	+10
4k	0,7	+10
8k	0,5	+10



Carrier: octave-band filtered noise

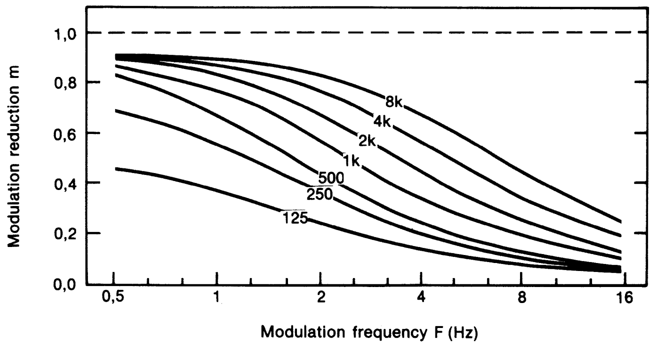


Fig. 2. Illustration of the way in which an MTF analysis can be performed, by using an octave-band filtered noise carrier, 100%-intensity modulated, for each modulation frequency successively. This leads to a family of MTF curves. As an illustration, each curve is derived for different theoretical values of T and S/N, as indicated

The Speech Transmission Index

The performance of a sound transmission system being quantified by a family of MTF curves, comprising $7 \times 14 = 98$ m values, the question remains of how to transform such a set of data into one single index representing the effect of that transmission system on speech intelligibility: the Speech Transmission Index (STI). The criterion for the relevance of such a transformation is, of course, that for a wide variety of transmission systems with different types of disturbances, the relationship between the STI values and the effect on speech intelligibility is unique, i.e., not system specific.

The algorithm for transforming a set of m values into a STI value, and the experimental verification on the basis of numerous intelligibility tests, is fully described elsewhere [4,5]. The most essential step in this transformation is a conversion of each of the 98 m values into an **apparent** signal-to-noise ratio $(S/N)_{app}$: irrespective of the actual type of disturbance causing the m value. It is interpreted as if it had been caused by interfering noise exclusively, $(S/N)_{app}$ being the signal-to-noise ratio which should have resulted in that m value. The conversion is defined mathematically by

$$(S/N)_{app} = 10 \log \frac{m}{1-m} \text{ dB} \qquad (2)$$

being the inverse of the expression given in Fig.1. A weighted average of the 98 apparent signal-to-noise ratios thus obtained results in the STI, after applying appropriate normalisation such that

$STI = 1{,}0$ when $(S/N)_{app} \geqslant 15\,\text{dB}$ for all 98 data points,

$STI = 0{,}0$ when $(S/N)_{app} \leqslant -15\,\text{dB}$ for all 98 data points.

By this calculation scheme each family of MTF curves can be transformed unambiguously into a STI value, by which the performance of that sound transmission system is quantified. Also, given the theoretical relations between $m(F)$ and the reverberation time T or the S/N ratio, the calculation scheme may be used for theoretical studies on the effect of reverberation and ambient noise in general. It has been shown that this provides a theoretical basis for a variety of empirical rules prevailing in auditorium acoustics [2]. As an example, one such familiar concept will be considered, i.e., the separation of the reverberant sound energy into a ɪseful (early) part and a detrimental (late) part (Fig.3). The upper left panel gives the theoretical relationship between *STI* and *S/N* ratio when noise is the only source of disturbance, with equal *S/N* ratio for all octave bands considered. Similarly, the upper right panel specifies the theoretical relationship between *STI* and *T* when (ideally exponential) reverberation is the only source of disturbance, with equal *T* values for all octave bands considered. By these relationships each *T* value can be converted into an equivalent *S/N* ratio (i.e., equivalent in terms of *STI*), leading to the curve in the lower panel of Fig.3.

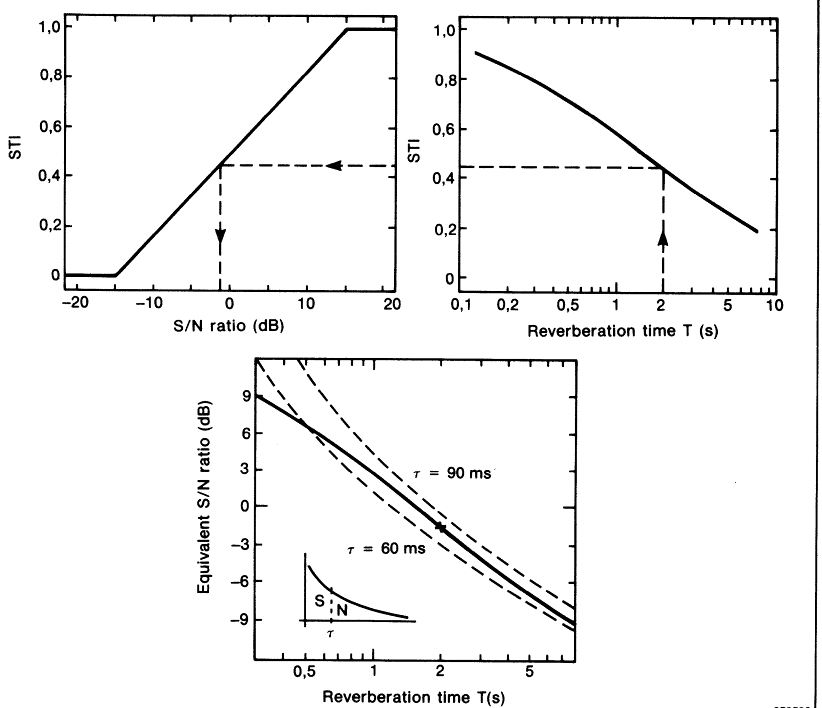


Fig. 3. The upper panels represent the theoretical relationships between the STI and S/N ratio, or STI and T. From this, each T value may be converted into an equivalent S/N ratio (solid curve in lower panel). The dashed curves in the lower panel represent an alternative and traditional approach in which the equivalent S/N ratio is defined by the ratio between the early and the late part of the echogram

The lower panel also represents an alternative way of interpreting the effect of reverberation in terms of *S/N* ratio, by subdividing the echogram into two parts: (1) an early useful part, to be regarded as "signal", and (2) a late detrimental part, to be regarded as "noise". For the theoretical case of purely exponential reverberation, the resulting *S/N* ratio as a function of *T* can be calculated, depending of course on the choice of the temporal boundary (τ) between the useful part and the detrimental part of the echogram. Two examples are indicated in Fig.3 for τ = 60 ms and τ = 90 ms. It can be seen that over a wide range of relevant reverberation times, the two approaches agree well for a τ value of 70 to 80 ms, a traditional value for separating useful and detrimental sound energy.

It has been shown that the STI calculation scheme can be used to predict the performance of an auditorium in the design stage, especially when modelling the sound field along the lines of geometrical acoustics, i.e., by ray-tracing [6].

STI and Speech Intelligibility

There exists a large body of experimental data on the relation between the STI and intelligibility scores obtained with speaker-listener panels [1, 4, 5, 7]. Typical relations are given in Fig.4. These relations are only illustrative since, besides the performance of the transmission system, intelligibility scores are affected by other factors also, such as the degree of training and skill of the speaker-listener panel and specific aspects of the speech material employed in the test (e.g., the use of a carrier phrase).

The qualification intervals (bad ... excellent) specified along the abscissa in Fig.4 are based on a large-scale study [7], involving various intelligibility tests and different languages. In this study the STI values were obtained by a specially designed measuring device (RASTI) which will be described in a separate contribution [8]. These intervals provide a background for the interpretation of actual STI measurements.

In the middle range, each qualification interval corresponds to an interval of 0,15 along the STI scale. This implies that differences of that magnitude are important: for two conditions with a STI difference of 0,15, the difference in speech intelligibility is significant and clearly noticeable. Accordingly, for an actual STI-measuring device one requires that the accuracy interval (e.g., the standard deviation for repeated measurements) is considerably smaller than 0,15. This may serve as a guideline for the implementation of the MTF concept in room acoustics along the lines presented in this contribution.

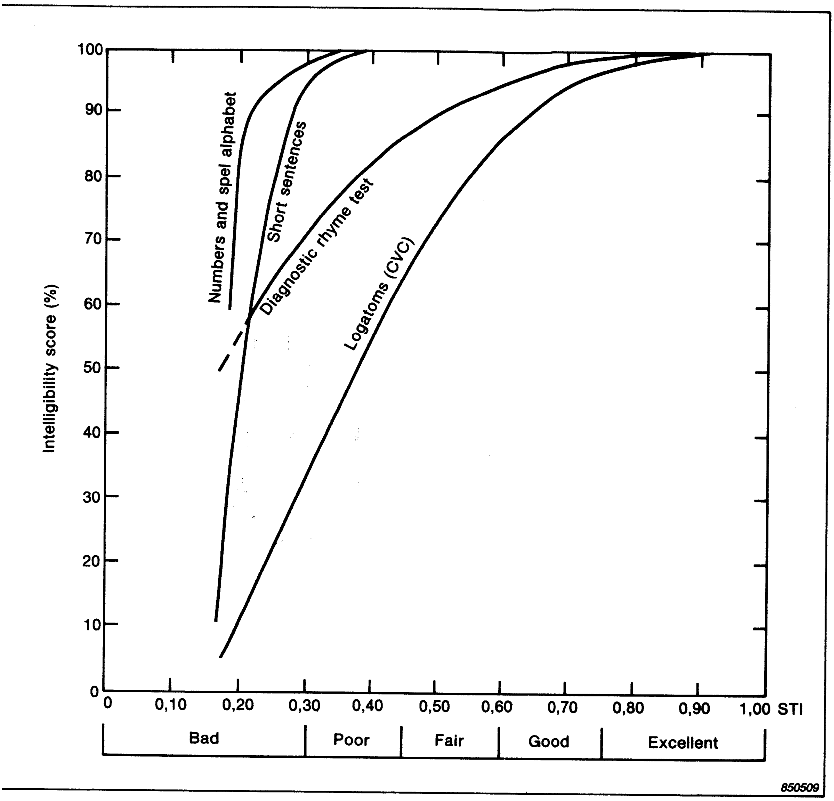


Fig. 4. Typical relations between the STI and intelligibility scores for various types of tests

References

[1] HOUTGAST, T & STEENEKEN, H.J.M. "The Modulation Transfer Function in Room Acoustics as a Predictor of Speech Intelligibility", *Acustica* 28 (1973), 66–73.

[2] HOUTGAST, T., STEENEKEN, H.J.M. & PLOMP, R. "Predicting Speech Intelligibility in Rooms from the Modulation Transfer Function. I. General Room Acoustics", *Acustica* 46 (1980), 60–72.

[3] HOUTGAST, T. and STEENEKEN, H.J.M. "A Review of the MTF Concept in Room Acoustics and Its Use for Estimating Speech Intelligibility in Auditoria", accepted for publication by the *J. Acoust. Soc. Amer.*77 (1985), 1060–1077.

[4] STEENEKEN, H.J.M. and HOUTGAST, T "A Physical Method for Measuring Speech-Transmission Quality", *J. Acoust. Soc. Amer.* 67 (1980), 318–326.

[5] HOUTGAST, T. & STEENEKEN, H.J.M "Experimental Verification of the STI", *Proceedings Fourth Internation Congress on Noise as a Public Health Problem* (Turin, 1983), 477–487.

[6] RIETSCHOTE, H.F. van & HOUTGAST, T. "Predicting Speech Intelligibility in Rooms from the Modulation Transfer Function. V. The Merits of the Ray-tracing Model Versus General Room Acoustics", *Acustica* 53 (1983), 72–78.

[7] HOUTGAST, T. and STEENEKEN, H.J.M. "A Multi-Language Evaluation of the RASTI-Method for Estimating Speech Intelligibility in Auditoria", *Acustica* 54 (1984), 185–199.

[8] STEENEKEN, H.J.M. & HOUTGAST, T. "RASTI": A Tool for Evaluating Auditoria", B & K *Technical Review (this issue)* (1985).

RASTI: A TOOL FOR EVALUATING AUDITORIA

by

*H.J.M. Steeneken**
and
*T. Houtgast**

ABSTRACT

The RASTI-method for the objective measurement of speech intelligibility in auditoria is described and applied in a number of practical conditions. The Modulation Transfer Function (underlying each RASTI measurement) can be used to obtain diagnostic information concerning the individual octave band contribution, in terms of the signal-to-noise ratio and the early-decay time at the measuring position. Some examples are given of the use of RASTI in mapping an audience area by iso-intelligibility contours, and in evaluating the effectiveness of a PA system. A computer program to analyze the measurement results is discussed and given in the Appendix.

SOMMAIRE

L'article décrit et donne des exemples d'applications concrètes de la méthode RASTI de mesure objective de l'intelligibilité de la parole dans un auditoire. La Fonction de Transfert de Modulation (qui est à la base de toutes les mesures RASTI) peut être utilisée pour obtenir des informations servant au diagnostic et concernant les contributions individuelles des bandes d'octave, du rapport signal /bruit et du temps de début de décroissnce de la réverbération, à l'endroit où l'on effectue la mesure. Des exemples sont donnés de l'utilisation de la méthode RASTI pour cartographier une zone d'audition par contours d'iso-intelligibilité et dans l'évaluation de l'efficacité d'un système amplificateur de puissance sonore. Des détails sont fournis sur un programme d'ordinateur analysant les résultats des mesures; le listage du programme est donné en appendice.

ZUSAMMENFASSUNG

In dieser Abhandlung wird die RASTI-Methode (**R**apid **S**peech **T**ransmission **I**ndex), ein Verfahren zur objektiven Messung der Sprachverständlickeit, vorgestellt und die praktische Anwendung erläutert. Die Modulation Transfer Function (MTF), die jeder RASTI-Messung zugrunde liegt, enthält Informationen über den Anteil der einzelnen Oktaven an der RASTI-Messung, den Fremdgeräuschabstand

* Institute for Perception TNO; Soesterberg, The Netherlands.

und die Anfangsnachhallzeit. Weiterhin wird an Beispielen gezeigt, wie sich mit der RASTI-Methode Iso-Sprachverständlichkeitskurven von Zuhörerräumen aufnehmen lassen und Sprachunterstützungsanlagen beurteilt werden können. Ein Computer-Programm zur Analyse der Meßergebnisse wird diskutiert und ist als Anhang gegeben.

Introduction

The use of the Modulation Transfer Function (MTF) in room acoustics for quantifying the effect of an enclosure on speech intelligibility has been

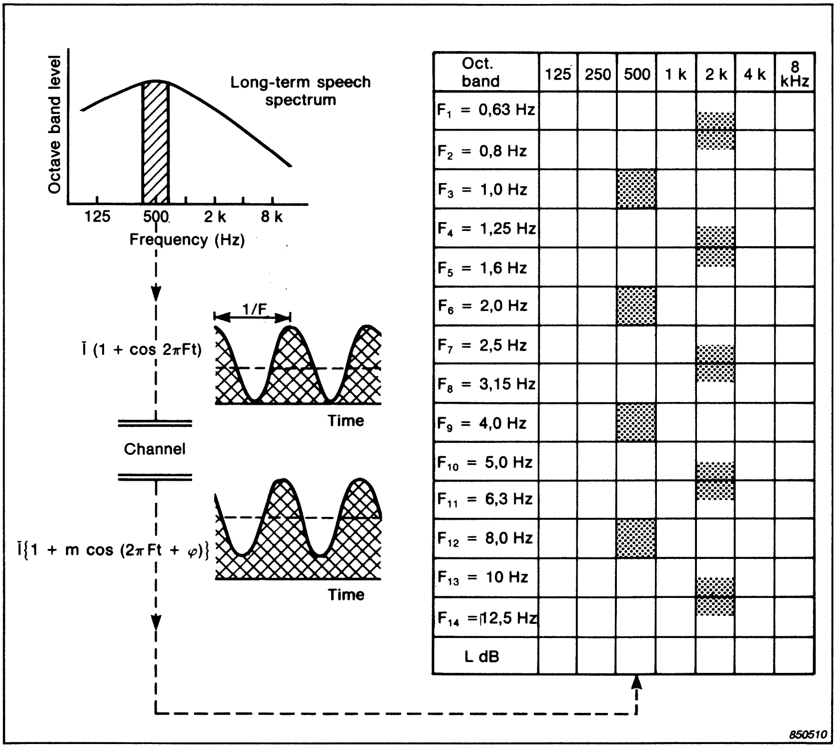


Fig. 1. For a complete analysis, the modulation reduction factor m is determined for 14 modulation frequencies and for 7 octave bands, resulting in a matrix of 98 data points. The octave band levels are measured as well. For the condensed procedure (RASTI) only nine modulation reduction factors are considered, marked by the hatched positions in the matrix

presented earlier [1, 2, 3]. The final index STI is based on a set of measurements as illustrated in Fig.1: for each of seven octave bands (centre frequencies from 125 Hz up to 8 kHz), 14 modulation frequencies are considered at 1/3-octave intervals ranging from 0,63 Hz up to 12,5 Hz. Thus, the final STI is based on a matrix comprising 98 data points, and is derived along the lines described in [4].

For the great majority of actual situations in auditoria, this set of 98 data points constitutes an unnecessarily detailed grid of analysis. Therefore, for a fast evaluation of auditorium conditions a more rapid measuring procedure was developed, based on a subset of the original 98 data points. The resulting index is called RASTI (Rapid STI), and is based on nine data points as indicated in Fig.1 (hatched cells in the matrix). Thus, the analysis is restricted to only two octave bands (centre frequencies 500 Hz and 2 kHz), and to four or five modulation frequencies for the two octave bands.

The RASTI system produced by Brüel & Kjær according to this concept gives a STI value within 8 s. The individual m values for the nine data points as well as the octave band levels are displayed by the system. This offers the possibility of obtaining diagnostic information on the nature of the (acoustic) degradation of the test signal during the transmission. Expected values of background noise can be fed in manually and the RASTI value measured. In this paper we will apply the RASTI method to extract the signal-to-noise ratio (*S/N*) and the early-decay time (EDT) from the MTF. A computer program for this purpose, written in PASCAL, is included in the Appendix.

Definition of the Rasti Method

Following the definition as described elsewhere [1, 3, 5] for the STI, the RASTI can be derived by the following steps as explained in the IEC-standard [6]:

a) Specify the A-weighted equivalent level ($L_{eq,A}$) of a running speech sample which can serve as a target for adjusting the level of the test signal. When the speech is replaced by the test signal, the long-term RMS level for the octave band with centre frequency 500 Hz should be 1 dB, and for the 2-kHz octave band 10 dB below this target. In this way the test signal spectrum meets the long term speech [1] spectrum for the two octave bands considered. The directivity pattern of the loudspeaker should reflect the directivity pattern of a natural speaker. (The B & K Speech Transmission Meter, Transmitter, Type 4225 is calibrated to a representative test-signal level given in the IEC-standard [6]).

b) Apply sinusoidal intensity modulation to a pink noise carrier, for the 500-Hz octave band at modulation frequencies of 1, 2, 4 and 8 Hz, and for the 2-kHz octave band at modulation frequencies of 0,7, 1,4, 2,8, 5,6 and 11,2 Hz.

c) Use an omni-directional microphone at the listener's position. Apply adequate octave band filtering and subsequent analysis for deriving the modulation index for each relevant modulation frequency in the intensity envelope.

d) Specify the modulation-reduction factor m (the ratio between the resulting modulation index and the initial modulation index of the test signal) for the relevant modulation frequencies in both bands.

e) Convert each of the nine m values into an apparent signal-to-noise ratio, according to:

$$(S/N)_{app} = 10 \log \left[\frac{m}{1-m} \right] \tag{1}$$

f) Truncate the obtained values of $(S/N)_{app}$ when exceeding the range of $\pm$ 15 dB.

g) Determine the mean of the nine values thus obtained: the mean apparent signal-to-noise ratio $(S/N)_{app}$.

h) Normalize to an index ranging from 0 to 1:

$$RASTI = \left[\overline{(S/N)}_{app} + 15 \right] / 30 \tag{2}$$

Implementation of the measuring procedure

Given the definition of RASTI, the underlying measurements can be performed in many ways. For the purpose of a fast screening device, a specific test-signal was developed, illustrated in Fig.2. Both octave bands are presented simultaneously, each with an intensity envelope in which the four or five relevant modulation frequencies are presented simultaneously too. Since the intensity envelope cannot drop below zero, the modulation index for each individual modulation frequency can only

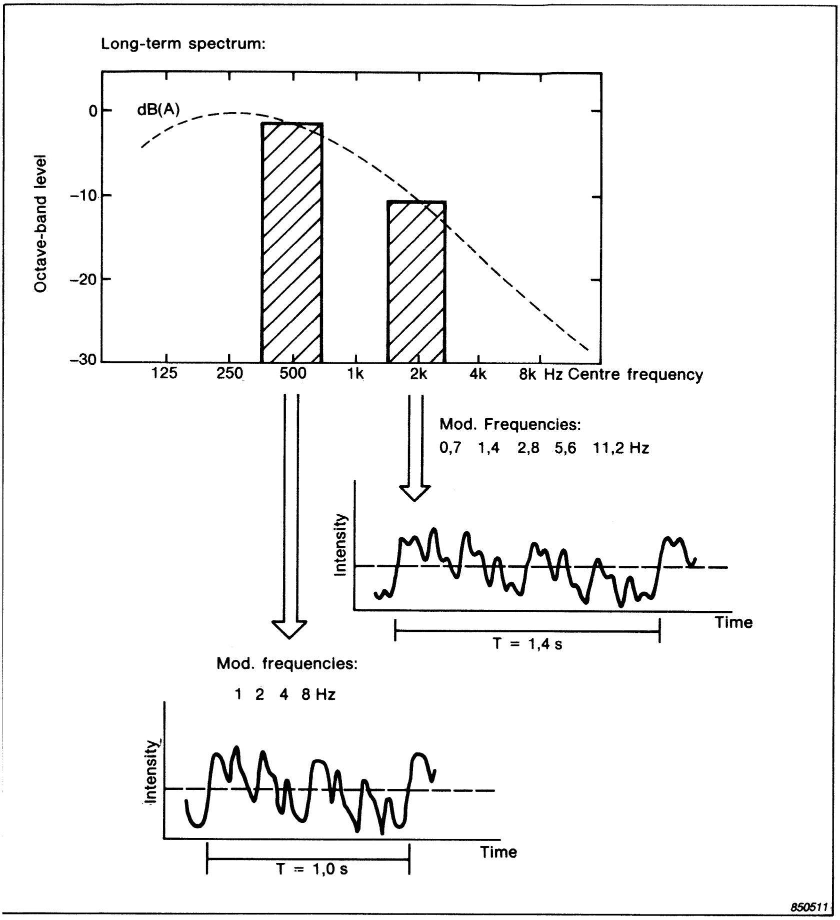


Fig. 2. Illustration of the RASTI test signal. Two octave bands of noise are presented simultaneously, with an intensity envelope comprising four or five simultaneous modulation frequencies, and with a modulation index of 0,4 and 0,32, respectively

be 0,4 and 0,32 for the octave bands with centre frequency 500 Hz and 2 kHz, respectively. Hence, in specifying the modulation-reduction factor m, the resulting modulation index is referred to this initial modulation index. As shown in Fig.2, the intensity envelope $I(t)$ is periodic; for the octave band with centre frequency 500 Hz ($T = 1\ s$) it is equal to:

$$I(t) = 1 + 0{,}4\,(\sin 2\pi \frac{t}{T} + \sin 2\pi \frac{2t}{T} + \sin 2\pi \frac{4t}{T} + \sin 2\pi \frac{8t}{T}) \qquad (3)$$

For the octave band with centre frequency 2 kHz ($T = 1{,}43\,s$) it is equal to:

$$I(t) = 1 + 0{,}32\,(\sin 2\pi \frac{t}{T} + \sin 2\pi \frac{2t}{T} + \sin 2\pi \frac{4t}{T} + \sin 2\pi \frac{8t}{T} + \sin 2\pi \frac{16t}{T}) \qquad (3)$$

Actually, the square root of these functions is used for the amplitude modulation of the noise carriers of the two octave bands which results in the desired intensity modulation with two repetition frequencies (1 Hz for the 500 Hz octave band and 0,71 Hz for the 2 kHz octave band).

Thus, rather than the original approach in which the test signal and the analysis are adapted to each data point successively (as shown in Fig.1, needing a time synchronisation between source and receiver), in the present approach this test signal is produced continuously, and allows a parallel analysis of the two octave bands and of the different modulation frequencies. Of course, the duration of the time interval during which the analysis of the test signal is performed, has great influence on the accuracy (reproducibility) of the resulting RASTI value. For the present purpose a basic measurement period of about 8 s was adopted.

Acoustical Properties of an Enclosure which can be derived from the MTF

The distortions affecting the intelligibility of a speech signal in an enclosure can be divided into two groups:

– signal-independent disturbing sounds, such as background noise introduced by air conditioners, traffic or by the public;

– signal-dependent disturbing sounds, such as reverberation and echoes.

Both types of disturbing signals have their specific effect on the modulation transfer function. The degradation introduced by background noise and the degradation introduced by reverberation can therefore be estimated individually from the MTF (see also Fig.1 of the preceding paper). In principle it is possible to obtain the delay time and the relative strength of an echo as well. However, for such an estimation the MTF has to be described with a high resolution in the modulation-frequency domain and cannot be obtained from a MTF described with only four or five modulation frequencies. Therefore, we will restrict ourselves to the contribution of background noise (expressed in the *S/N* ratio) and the contribution of the reverberation (expressed in the early decay time, EDT).

The effect of background noise

As indicated in the preceding paper [2], the modulation-reduction factor is given by:

$$m(F) = \frac{1}{\sqrt{1 + \left[2\pi F \dfrac{T}{13{,}8} \right]^2}} \cdot \frac{1}{1 + 10^{(-S/N)/10}} \qquad (4)$$

where F is the modulation frequency in Hz, T the reverberation time in seconds and S/N the signal-to-noise ratio in dB.

Since, for each octave band, four or five m values are obtained for different modulation frequencies F, the corresponding values for T and S/N which best fit these data can be derived. This is done by an iterative procedure, taking into account that the significance of the first term (reverberation) increases for higher F values. A first estimate of T and S/N is obtained by the following empirical procedure:

1. Define the cut-off frequency of the MTF which separates the horizontal low-frequency part caused by the noise and the sloping high-frequency part. According to the criterion of Fig.3 the decrease of the modulation-reduction factor above the cut-off frequency is more than 0,15 versus the mean modulation reduction for modulation frequencies up to and including the cut-off frequency. This is indicated by the horizontal dashed lines in Fig.3.

2. Calculate the S/N ratio according to the second item of (4) with a mean modulation-reduction factor m based on the horizontal part as defined above, which should include at least two modulation frequencies. When it includes only the lowest modulation frequency, it

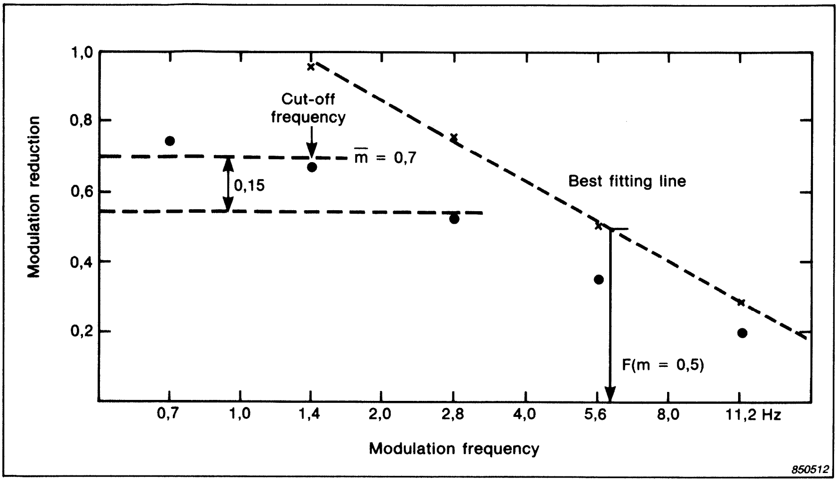


Fig. 3. Modulation transfer function as obtained for a combination of noise and reverberation. The horizontal part is described by the modulation-reduction factor for the modulation frequencies 0,7 and 1,4 Hz with a mean modulation reduction m = 0,7. The cut-off frequency is 1,4 Hz. The best-fitting line (based on 4 points marked x) describes the slope of the MTF and can be calculated after correction for the noise

cannot be excluded that this modulation-reduction is introduced by a combination of noise and reverberation (long reverberation times). This can be investigated, however, by a second measurement of the MTF with an increased signal level. In the Brüel & Kjær system a + 10 dB level can be selected.

3. Correct the MTF with a factor $1/m$, thus compensating for the effect of noise. The corrected MTF is now defined by reverberation only.

4. Calculate on least-square basis the best-fitting line for the modulation frequencies above and including the cut-off frequency, to estimate the slope of the MTF (Fig.3). This slope typically covers three octaves of the modulation-frequency scale. Since for RASTI, the modulation frequency increases in octave steps, this line can be described with a maximum of four modulation-reduction factors.

5. Derive from the line equation as found in step 4, the modulation frequency F_0 where the modulation-reduction factor is reduced to $m = 0,5$. Calculate T in s with the formula $T = 3,8 / F_0$, as can be obtained from (4) for $m = 0,5$.

The method described above is given in the Appendix as a PASCAL computer program. After the estimation of S/N and T this program calculates the theoretical MTF with the estimated S/N and T. The root-mean-square of the differences between the original and estimated MTFs is calculated as well, and considered as a measure of the fit. Typically this fit measure is below 0,1. As an example Fig.4 gives the estimated MTF (o) together with the original modulation-reduction factors (x) for a condition with a combination of reverberation and noise. As the calculation of T and S/N are based on an empirial algorithm we verified the T and S/N thus obtained by a systematic variation of T and S/N around the predicted value giving an optimal fit. The output of the computer program together with the systematic variation of T and S/N is given with Fig.4. For this particular example the estimates of T and S/N can be considered as the best fit.

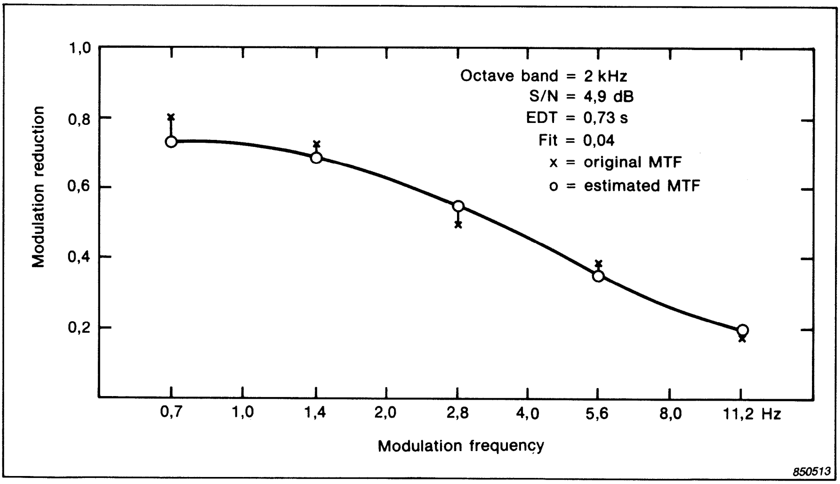


Fig. 4. MTF described by 5 modulation-reduction factors from a measurement in a practical situation (x). The estimated S/N ratio and EDT is obtained from this MTF and the theoretical MTF (o) is calculated with the estimated S/N ratio and EDT. The relation between the two MTFs is expressed in a goodness-of-fit measure (root-mean-square of differences)

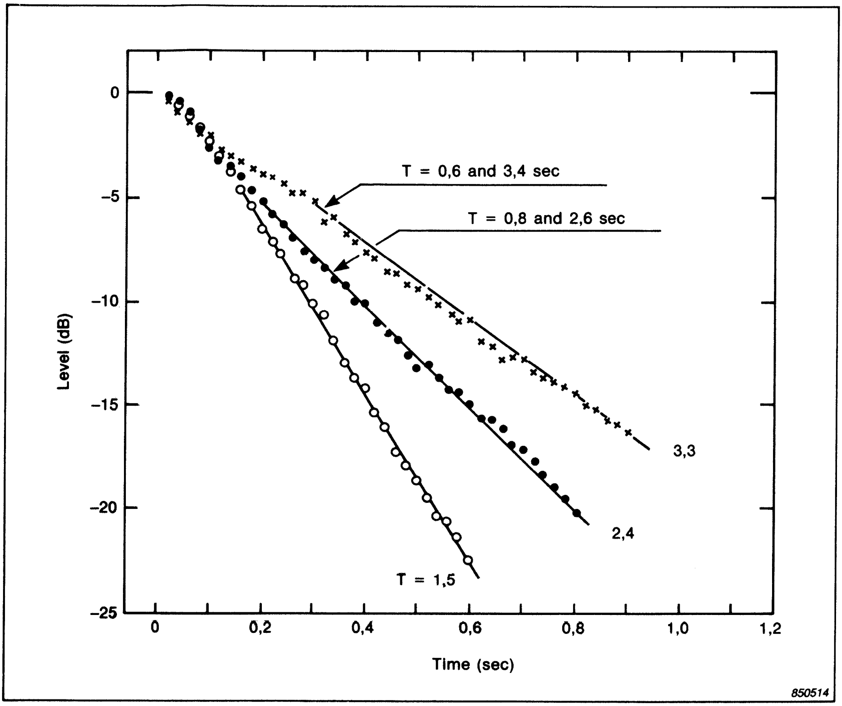


Fig. 5. Reverberation curves for three conditions (exponential decay for T = 1,5 s, and two combinations of combined decay curves resulting in T = 2,4 s and T = 3,3 s, respectively). The combined curves are results from recordings in rooms with different reverberation time. The effect on speech intelligibility is equal for all three conditions

The reverberation time *T* derived from an MTF according to the approach described above seems to be defined by the early-decay of the enclosure where the MTF was measured, rather than by the traditional reverberation time *T*. In Fig.5 three decay curves are given which were created from a laboratory experiment where conditions with different reverberation times were studied. The early decay time for these conditions was constant (EDT = 1,5 s), the traditional reverberation time, however, varied (1,5; 2,4; 3,3 s). Experimental results showed that the intelligibility scores as well as the MTFs for these conditions were essentially identical (Fig.6). This shows the significance of the early-decay for the transfer of fluctuating signals and for speech intelligibility.

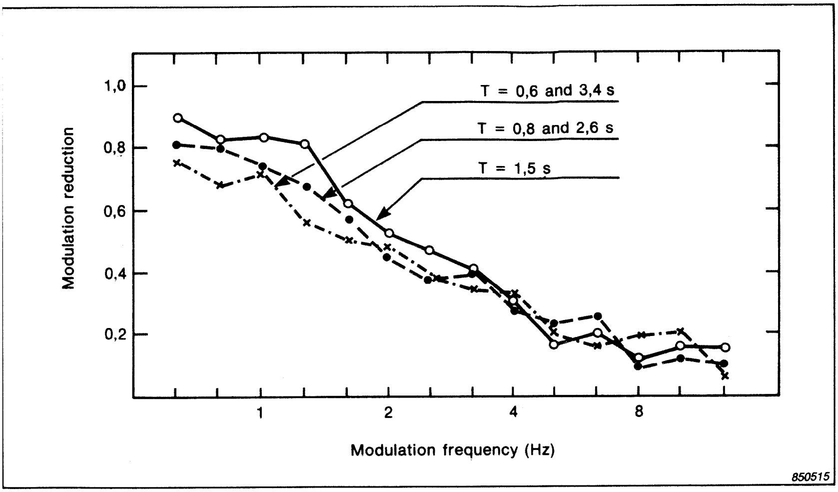


Fig. 6. MTF for the same conditions as given in Fig.5. The close agreement between the curves indicates the importance of the first part of the decay curve (Early Decay)

In the Brüel & Kjær instrument the equivalent *S/N*-ratio and the equivalent EDT is calculated for each measurement. The equivalent *S/N*-ratio is defined as the *S/N* ratio calculated from the RASTI value assuming the EDT value to be zero, and similary the equivalent EDT is the early decay time assuming the *S/N*-ratio to be better than 15 dB. This feature permits quick evaluation of speech intelligibility problems during the measurements.

Factors governing the reliability of a RASTI measurement

The measurement of the modulation-reduction factors is based on a Discrete Fourier Transform (DFT) of the envelope of the received signal for the two octave bands. The measuring time defines the selectivity of the DFT for each modulation frequency. The original modulation levels of the test signal together with the selectivity of the DFT analysis determine the systems accuracy and reproducibility for conditions with a fluctuating background noise rather than a constant background noise. For instance, voice babble comprises the entire range of modulation frequencies as this range is derived from the fluctuation rhythms of running speech [4, 7]. The effect of such fluctuating noises on the measurement results in situations with a poor transmission quality and the selectivity (for a short measurement time) can be very poor. An estimation of the reliability of a RASTI measurement can then be obtained by switching off the test signal and performing another RASTI measurement, which indicates the lower limit. The typical RASTI value thus obtained for voice babble and at the shortest measuring time is about RASTI = 0,4. The RASTI analysis algorithm in the Brüel & Kjær system has a built-in facility for the detection of random fluctuating signals and gives a warning.

The Application of the RASTI method in Auditoria

The evaluation of the intelligibility for an auditorium, conference room or a public-address system comprises several aspects such as:

- mapping the intelligibility throughout the auditorium;
- investigating the contribution of various background noise levels;
- evaluating the contribution of a public-address system in conditions with various background noise levels or different loudspeaker positions.

All these aspects can be studied with the RASTI method and, as was discussed before [8, 9], not only an estimate is obtained of the speech transmission quality but also of the nature of the transmission loss.

iso-RASTI Contours

The normal procedure for determining iso-RASTI contours is to measure RASTI values at a large number of positions evenly distributed throughout the audience area. In this way the RASTI can be mapped and used to construct iso-RASTI contours.

Depending on the gradient between successive RASTI values and on the resolution of the measuring grid, iso-RASTI contours can be drawn for 0,05, 0,1 or 0,2 STI intervals. In Fig.7 iso-RASTI contours, based on 29 measuring positions, are given for a lecture hall. In this example, for the empty hall and no background noise, the RASTI varies from 0,70 to 0,58 which implies an intelligibility rating between good and fair. Normally, the acoustics consultant starts with a measuring session in the absence of an audience, which may result in a non-representative absorbtion and noise level. The absence of a representative background noise can be compensated for by the application of an artificial noise source during the measurements or by correcting the RASTI for an imaginary back-

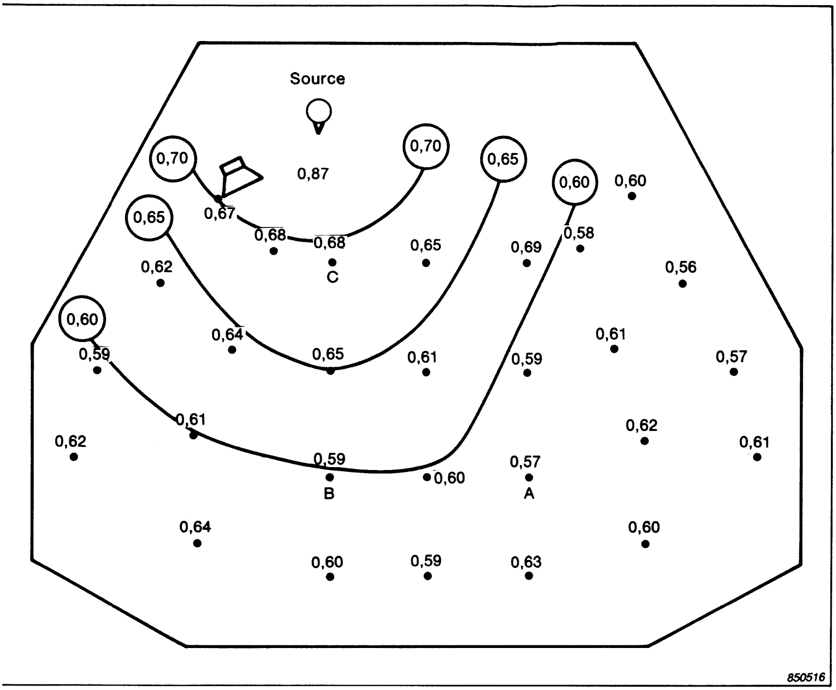


Fig. 7. iso-RASTI contours for an auditorium, without public, background noise and PA system. The original (29) data points where the contours estimated from, are given as well

ground noise in the Brüel & Kjær instrument. In the latter case we can also correct the MTFs for a certain noise level according to equation (4). The program given in the Appendix has an option to recalculate the RASTI for a given octave band level of background noises at 500 Hz and 2 kHz.

Fig.8 shows iso-RASTI contours obtained from the same measuring data as given in Fig.7, but corrected for an imaginary background noise with octave band levels (500 Hz and 2 kHz) of 40 dB.

Under the condition presented in Fig.7, with small differences of the RASTI, it is not very relevant to detect areas in the audience with a poor intelligibility, however for an imaginary background noise level of 40 dB for the octave bands 500 Hz and 2 kHz the iso-RASTI contours change dramatically. As given in Fig.8 the RASTI varies from 0,54–0,36 which means that areas with a poor intelligibility can be detected, caused by a low level of the direct sound far from the speaker or far from any reflecting surface.

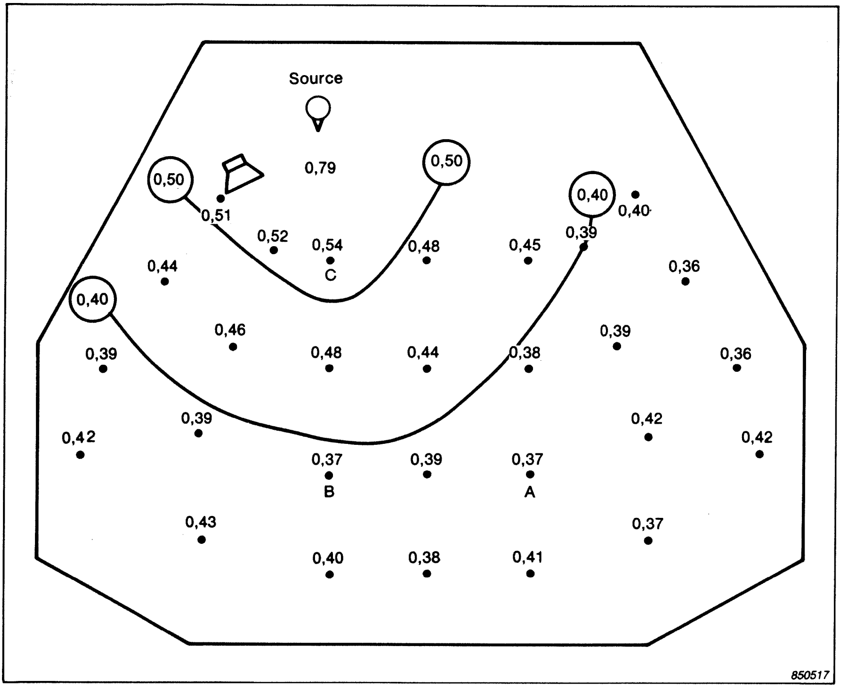


Fig. 8. iso-RASTI contours for the same data points as given in Fig.7 but corrected for an imaginary background noise with an octave band level of 40 dB

For three positions in the auditorium (marked A, B, C in Fig.7 and 8) the RASTI as a function of the background noise level is given in Fig.9 A, B, C (solid lines). These graphs indicate a low signal level at the positions A and B. Besides acoustical measures, such as a reflecting surface behind the speaker, a public-address system (PA) can be applied to increase the level of the (direct) sound.

The Evaluation of a Public-Address System with the RASTI

The application of a PA system in an auditorium increases the direct-sound level at the listener's position and hence the signal's resistance

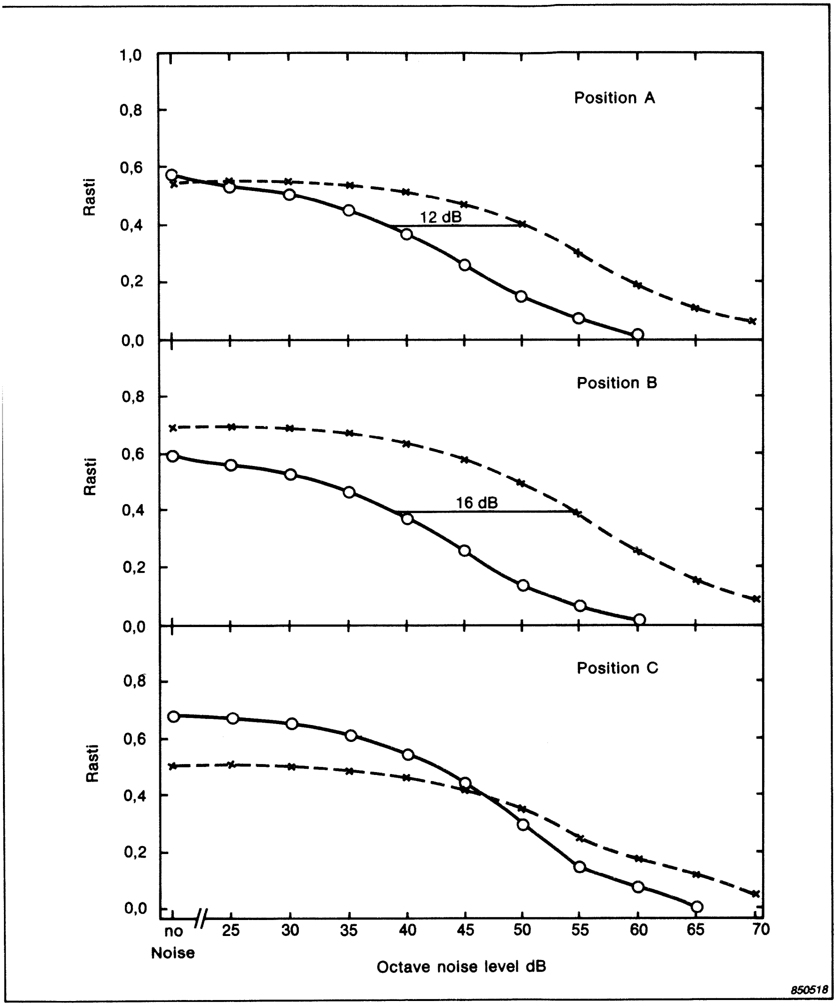


Fig. 9. RASTI value as a function of an imaginary background noise level for three positions (marked A, B, C in Fig.7) and without (o) and with (x) a public-address system. The position of the loudspeaker is also marked in Fig.7

against background noise. The signal level at the listener's position is defined by the system gain and by the position and directivity of the microphone and loudspeakers, and also by the acoustics of the room.

For a poorly designed system, however, with the loudspeakers not optimally directed to the (absorbing) audience, the reverberation field increases, which may result in a decrease of the intelligibility at low noise levels. An example of such a situation is given in Fig. 9C (dashed curve). For this condition a PA system in the auditorium as given in Fig.7 was applied. Only one loudspeaker at the marked position was used. The loudspeaker was placed above the audience, directed to position B.

The RASTI was measured at position A, B and C and the results, as a function of an imaginary background noise level, are given in Fig.9 (dotted lines). For position B the RASTI value is increased even for the conditions without background noise, which implies a better ratio between the direct and the indirect sound.

We can estimate the contribution of the PA system by the increase of the resistance against background noise for a given RASTI value. As shown in Fig.9B this "effective" gain is 16dB for a RASTI value of 0,4. For position A this effective gain (of the same PA system) is 11dB and at position C it is 0dB. With this method of validation, an optimal adjustment of the loudspeaker positioning and direction can be found.

In order to exclude the contribution of the transmitting room and the microphone and the re-transmission of the indirect sound by the system, the RASTI test signal can be connected electrically to the PA system. As the RASTI method does not account for non-linear distortion, an overload of the PA amplifier should be avoided.

Summary

The RASTI method applied in auditoria offers a fast method of evaluating the intelligibility and of detecting areas with a poor intelligibility or areas with a low resistance against background noise. This is illustrated by iso-RASTI contours. The RASTI method is standardized by IEC.

The contribution of a PA system can be expressed by the effective gain of the PA system at a central position, which reflects the increased resistance against background noise at a given intelligibility criterion.

For calculating the effect of an imaginary background noise on the RASTI value, this value can be introduced in the instrument or a special program given in the Appendix can be used. This program also estimates from the MTF as measured by RASTI, the signal-to-noise ratio and the early-decay time at the measuring position.

References

[1] HOUTGAST, T. & STEENEKEN, H.J.M. "The Modulation Transfer Function in Room Acoustics as a Predictor of Speech Intelligibility", *Acustica* 28, 66–73 (1973).

[2] HOUTGAST, T. & STEENEKEN, H.J.M. "The Modulation Transfer Function in room acoustics"; *B&K Tech. Rev (this issue)* (1985).

[3] HOUTGAST, T. & STEENEKEN, H.J.M. "The MTF concept in Room Acoustics and its use for estimating speech intelligibility in auditoria", *J. Acoust. Soc. Amer.*77, 1069–1077 (1985).

[4] STEENEKEN, H.J.M. & HOUTGAST, T. "A Physical Method for Measuring Speech-Transmission Quality", *J. Acoust. Soc. Amer.* 67, 318–326 (1980).

[5] HOUTGAST, T. & STEENEKEN, H.J.M. "A Multi-Language Evaluation of the RASTI-Method for Estimating Speech Intelligibility in Auditoria" *Acustica* 54, 185–199 (1984)

[6] Draft-IEC publication 268: Sound system equipment – Part 16. "Report on the RASTI method for the objective rating of speech intelligibility in auditoria".

[7] STEENEKEN, H.J.M. & HOUTGAST, T. "The Temporal Envelope Spectrum of Speech and its Significance in Room Acoustics" In: *Proceedings Eleventh International Congress on Acoustics* Vol.7, pp. 85–88, Paris (1983).

[8] STEENEKEN, H.J.M. & HOUTGAST, T. "Some Applications of the Speech Transmission Index (STI) in auditoria" *Acustica* 51, 229–234 (1982).

[9] STEENEKEN, H.J.M. & HOUTGAST, T. "Measuring iso-Intelligibility contours in auditoria" *Proc. F.A.S.E. third symp. on building acoustics,* Dubrovnik, (1979).

APPENDIX

he computer program "CONMTF", which extracts from the MTF an stimated S/N ratio and EDT, is written in the high-level language ASCAL. This implies a structural programming (according to an acous-ician's criterion!). The individual algorithms are programmed in separate rocedures together with separate input and output procedures.

he program runs on an IBM personal computer and uses the Microsoft ASCAL compiler. The program can be converted to any computer ystem (taking into account the system's restrictions).

he program is structured around a menu with the following options:

1 = input of octave levels and MTFs from keyboard

2 = input octave levels and MTFs from a file

3 = calculation of the RASTI, the octave contributions to the RASTI (MTF), the S/N ratio and the EDT

4 = calculation of the RASTI for a given noise level

5 = calculation of the RASTI as a function of the noise level from 25 dB to 70 dB

6 = initialization of the line-printer output

7 = stop

he procedures used for the menu items 3, 4 and 5 are the same and onsist of the algorithm for the RASTI calculation, diagnostics S/N EDT, alculation of the fit and output procedures. These procedures follow the quations described before.

Ve have tested the program extensively, but we cannot accept responsi-ility for errors in this program or its application.

```
  PROGRAM CONMTF (INPUT,OUTPUT);

CONST
  M0 = 'MENU';
  M1 = '1=Input from keyboard';
  M2 = '2=Input from file';
  M3 = '3=STI';
  M4 = '4=STI for Fixed Background Noise Level';
  M5 = '5=STI as function of Background Noise Level';
  M6 = '6=Printer Output Enable';
  M7 = '7=Stop';
  KEYB='Keyboard
  NAMELEN=15;
  SHIFT=15;
  RANGE=30;
  DELTA=0.15;
  F1=1.02;
  F2=0.73;
  TWPI=6.2831853;

TYPE
  OCTMOD = ARRAY[1..5] OF REAL;
  NAME = STRING(NAMELEN);

VAR
  MTFBUF,FITBUF:ARRAY [1..2] OF OCTMOD;
  L500,L2000:REAL;
  I500,I2000:REAL;
  LN500,LN2000:REAL;
  IN500,IN2000:REAL;
  SNROCT,EDTOCT,MTFCOR,SIGMA:ARRAY[1..2] OF REAL;
  MCOR500,MCOR2000:REAL;
  MTI500,MTI2000:REAL;
  RASTI:REAL;
  MMAX,MODE:INTEGER;
  NOISE,PRINTENABLE:BOOLEAN;
  DIAGN:ARRAY[1..2,1..2] OF BOOLEAN;
  INPUTFILE:NAME;
  INFILE,OUTFILE,PRINTER:TEXT;

  PROCEDURE MENU;

   BEGIN
     WRITELN;
     WRITELN (M0);
     WRITELN (M1);
     WRITELN (M2);
     WRITELN (M3);
     WRITELN (M4);
     WRITELN (M5);
     WRITELN (M6);
     WRITELN (M7);
   END;

  PROCEDURE INPUT;

  VAR
    I:INTEGER;

   BEGIN
     WRITE ('L500=');READLN (L500);
     WRITE ('L2000=');READLN (L2000);
     WRITELN ('Give 4 m values octave 500 Hz');
     FOR I:=1 TO 4 DO READLN (MTFBUF[1,I]);
     WRITELN ('Give 5 m values octave 2000Hz');
     FOR I:=1 TO 5 DO READLN (MTFBUF[2,I]);
   END;
```

```
PROCEDURE FILEREAD;

VAR
  I:INTEGER;

 BEGIN
   WRITE ('Filename = ');
   READLN (INPUTFILE);
   ASSIGN (INFILE,INPUTFILE);
   RESET (INFILE);
   READLN (INFILE,L500);
   READLN (INFILE,L2000);
   FOR I:=1 TO 4 DO READLN (INFILE,MTFBUF[1,I]);
   FOR I:=1 TO 5 DO READLN (INFILE,MTFBUF[2,I]);
   CLOSE (INFILE);
 END;

PROCEDURE OUTP;

VAR
  I:INTEGER;

 BEGIN
   IF PRINTENABLE
    THEN
     BEGIN
WRITELN(PRINTER);
WRITELN(PRINTER,'Filename=',INPUTFILE);
WRITELN(PRINTER);
WRITELN (PRINTER,'RASTI =',RASTI:8:2);
WRITELN (PRINTER,'Octave=      500 Hz    2000 Hz');
IF NOISE
 THEN
 WRITELN (PRINTER,'Lnoise=',LN500:8:2,'   ',LN2000:8:2,' dB');
WRITELN (PRINTER,'Lsign =',L500:8:2,'   ',L2000:8:2,' dB');
WRITELN (PRINTER,'MTI   =',MTI500:8:2,'   ',MTI2000:8:2);
WRITE (PRINTER,'SNR   =');
IF DIAGN[1,1]
 THEN
 WRITE (PRINTER,SNROCT[1]:8:2)
 ELSE WRITE(PRINTER,'     ****');
IF DIAGN[1,2]
 THEN
 WRITELN (PRINTER,'   ',SNROCT[2]:8:2,' dB')
 ELSE WRITELN(PRINTER,'        **** dB');
WRITE (PRINTER,'EDT   =');
IF DIAGN[2,1]
 THEN
 WRITE (PRINTER,EDTOCT[1]:8:2)
 ELSE WRITE (PRINTER,'     ****');
IF DIAGN[2,2]
 THEN
 WRITELN (PRINTER,'   ',EDTOCT[2]:8:2,' s')
 ELSE
 WRITELN (PRINTER,'        **** s');
WRITE (PRINTER,'Fit   =');
IF DIAGN[1,1] OR DIAGN[2,1]
 THEN
 WRITE (PRINTER,SIGMA[1]:8:2)
 ELSE
 WRITE (PRINTER,'     ****');
IF DIAGN[1,2] OR DIAGN[2,2]
 THEN
 WRITELN(PRINTER,'   ',SIGMA[2]:8:2)
 ELSE
 WRITELN (PRINTER,'        ****');
```

```
WRITELN (PRINTER,'Estimated MTF:');
IF DIAGN[1,1] OR DIAGN[2,1]
 THEN
  BEGIN
    WRITE (PRINTER,'MTF (500Hz)=');
    FOR I:=1 TO 4 DO WRITE(PRINTER,FITBUF[1,I]:5:2);
    WRITELN(PRINTER);
  END
 ELSE
 WRITELN (PRINTER,'No estimate for Octave 500Hz');
IF DIAGN[1,2] OR DIAGN[2,2]
 THEN
  BEGIN
    WRITE (PRINTER,'MTF  (2kHz)=');
    FOR I:=1 TO 5 DO WRITE(PRINTER,FITBUF[2,I]:5:2);
    WRITELN(PRINTER);
  END
 ELSE
 WRITELN(PRINTER,'No estimate for Octave 2kHz');
      END
     ELSE
      BEGIN
WRITELN;
WRITELN('Inputfile=',INPUTFILE);
WRITELN;
WRITELN ('RASTI =',RASTI:8:2);
WRITELN ('Octave=     500 Hz   2000 Hz');
IF NOISE
 THEN
 WRITELN ('Lnoise=',LN500:8:2,'  ',LN2000:8:2,' dB');
WRITELN ('Lsign =',L500:8:2,'  ',L2000:8:2,' dB');
WRITELN ('MTI   =',MTI500:8:2,'  ',MTI2000:8:2);
WRITE ('SNR   =');
IF DIAGN[1,1]
 THEN
 WRITE (SNROCT[1]:8:2)
 ELSE WRITE('    ****');
IF DIAGN[1,2]
 THEN
 WRITELN ('  ',SNROCT[2]:8:2,' dB')
 ELSE WRITELN('      **** dB');
WRITE ('EDT   =');
IF DIAGN[2,1]
 THEN
 WRITE (EDTOCT[1]:8:2)
 ELSE WRITE ('    ****');
IF DIAGN[2,2]
 THEN
 WRITELN ('  ',EDTOCT[2]:8:2,' s')
 ELSE WRITELN ('      **** s');
WRITE('Fit   =');
IF DIAGN[1,1] OR DIAGN[2,1]
 THEN
 WRITE (SIGMA[1]:8:2)
 ELSE
 WRITE('    ****');
IF DIAGN[1,2] OR DIAGN[2,2]
 THEN
 WRITELN ('  ',SIGMA[2]:8:2)
 ELSE
 WRITELN('      ****');
WRITELN ('Estimated MTF:');
IF DIAGN[1,1] OR DIAGN[2,1]
```

```
 THEN
  BEGIN
    WRITE ('MTF (500Hz)=');
    FOR I:=1 TO 4 DO WRITE(FITBUF[1,I]:5:2);
    WRITELN;
  END
 ELSE
 WRITELN('No estimate for Octave 500Hz');
IF DIAGN[1,2] OR DIAGN[2,2]
 THEN
  BEGIN
    WRITE ('MTF  (2kHz)=');
    FOR I:=1 TO 5 DO WRITE(FITBUF[2,I]:5:2);
    WRITELN;
  END
 ELSE
 WRITELN('No estimate for Octave 2kHz');
      END;
  END;

 PROCEDURE OUTPNOISE;

  BEGIN
    IF PRINTENABLE
     THEN
      BEGIN
WRITELN(PRINTER);
WRITELN(PRINTER,'RASTI =',RASTI:8:2);
WRITELN(PRINTER,'MTI    =',MTI500:8:2,'   ',MTI2000:8:2);
WRITELN(PRINTER,'Lnoise=',LN500:8:2,'   ',LN2000:8:2,'dB');
      END
     ELSE
      BEGIN
WRITELN;
WRITELN('RASTI =',RASTI:8:2);
WRITELN('MTI    =',MTI500:8:2,'   ',MTI2000:8:2);
WRITELN('Lnoise=',LN500:8:2,'   ',LN2000:8:2,'dB');
      END;
  END;

 PROCEDURE STIVALUE;
 VAR
   X,Y:REAL;
   I,J,IMAX,JMAX:INTEGER;

  BEGIN
    MTI500:=0;
    MTI2000:=0;
    IMAX:=4;JMAX:=5;
    FOR I:=1 TO IMAX DO
     BEGIN
IF MTFBUF[1,I]<=1
 THEN
 X:=MTFBUF[1,I]*MCOR500
 ELSE
 X:=1;
IF (X<=0)
 THEN
 X:=0.000001;
X:=(4.342944*LN(X/(1.000001-X))+SHIFT)/RANGE;
IF X<0
 THEN
 X:=0;
IF X>1
```

```
 THEN
 X:=1;
MTI500:=MTI500+X;
      END;
     FOR J:=1 TO JMAX DO
      BEGIN
IF MTFBUF[2,J]<=1
 THEN
 Y:=MTFBUF[2,J]*MCOR2000
 ELSE
 Y:=1;
IF (Y<=0)
 THEN
 Y:=0.000001;
Y:=(4.342944*LN(Y/(1.000001-Y))+SHIFT)/RANGE;
IF Y<0
 THEN
 Y:=0;
IF Y>1
 THEN
 Y:=1;
MTI2000:=MTI2000+Y;
      END;
   RASTI:=(MTI500+MTI2000)/(IMAX+JMAX);
   MTI500:=MTI500/IMAX;
   MTI2000:=MTI2000/JMAX;
 END;

PROCEDURE NOISECOR;

 BEGIN
   I500:=EXP((L500/10)*LN(10));
   I2000:=EXP((L2000/10)*LN(10));
   IN500:=EXP((LN500/10)*LN(10));
   IN2000:=EXP((LN2000/10)*LN(10));
   MCOR500:=I500/(I500+IN500);
   MCOR2000:=I2000/(I2000+IN2000);
 END;



  PROCEDURE SNR(OCT,MAX:INTEGER);

  VAR
    I:INTEGER;
    X,Y:REAL;
    READY:BOOLEAN;

   BEGIN
     Y:=0;
     READY:=FALSE;
     MMAX:=0;
      REPEAT
       MMAX:=MMAX+1;
       Y:=Y+MTFBUF[OCT,MMAX];
       X:=Y/MMAX;
       IF (MMAX=MAX)
THEN
READY:=TRUE
ELSE
IF ((X-MTFBUF[OCT,MMAX+1])>DELTA)
 THEN
 READY:=TRUE;
      UNTIL READY;
     IF X>1
```

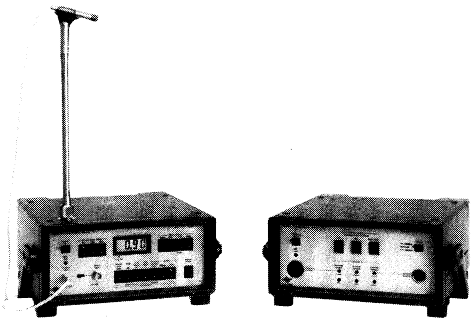
```
     THEN X:=1;
    IF MMAX>1
     THEN
      BEGIN
SNROCT[OCT]:=4.342944*LN(X/(1.000001-X));
MTFCOR[OCT]:=1/X;
      END
     ELSE
      BEGIN
DIAGN[1,OCT]:=FALSE;
MTFCOR[OCT]:=1;
      END;
  END;

 PROCEDURE EDT(OCT,MAX:INTEGER);
 VAR
   SUMX,SUMY,SUMXX,SUMXY:REAL;
   A,B,C:REAL;
   TOT:INTEGER;
   I:INTEGER;

  BEGIN
    SUMX:=0;SUMY:=0;
    SUMXX:=0;
    SUMXY:=0;
    TOT:=0;
    IF MMAX=1
     THEN
     MAX:=4;
    IF MMAX<MAX
     THEN
      BEGIN
FOR I:=MMAX TO MAX DO
 BEGIN
   C:=MTFBUF[OCT,I]*MTFCOR[OCT];
   SUMX:=SUMX+I;
   SUMY:=SUMY+C;
   SUMXX:=SUMXX+I*I;
   SUMXY:=SUMXY+I*C;
   TOT:=TOT+1;
 END;
B:=(SUMXY-SUMX*SUMY/TOT)/(SUMXX-SUMX*SUMX/TOT);
A:=(SUMY-B*SUMX)/TOT;
A:=(0.5-A)/B;
IF OCT=1
 THEN
 A:=F1*EXP((A-1)*LN(2))
 ELSE
 A:=F2*EXP((A-1)*LN(2));
EDTOCT[OCT]:=3.8/A;
IF (B>-0.15)
 THEN
 DIAGN[2,OCT]:=FALSE;
      END
     ELSE DIAGN[2,OCT]:=FALSE;
  END;

 PROCEDURE FIT(OCT,MAX:INTEGER);

 VAR
   I,J:INTEGER;
   X,XX,Y,F0,F:REAL;

  BEGIN
    IF OCT=1
```

```
       THEN
       F0:=F1
       ELSE
       F0:=F2;
      FOR I:=1 TO MAX DO
       BEGIN
F:=F0*EXP((I-1)*LN(2));
IF DIAGN[2,OCT]
 THEN
 FITBUF[OCT,I]:=1/SQRT(1+SQR(TWPI*F*EDTOCT[OCT]/13.8))
 ELSE
 FITBUF[OCT,I]:=1;
IF DIAGN[1,OCT]
 THEN
 FITBUF[OCT,I]:=FITBUF[OCT,I]/MTFCOR[OCT];
       END;
      X:=0;XX:=0;
      FOR I:=1 TO MAX DO
       BEGIN
Y:=MTFBUF[OCT,I]-FITBUF[OCT,I];
X:=X+Y;
XX:=XX+Y*Y;
       END;
      SIGMA[OCT]:=SQRT(XX/MAX);
    END;

   PROCEDURE DIAGNOSTICS;

    BEGIN
      DIAGN[1,1]:=TRUE;DIAGN[1,2]:=TRUE;
      DIAGN[2,1]:=TRUE;DIAGN[2,2]:=TRUE;
      SNR(1,4);
      EDT(1,4);
      FIT(1,4);
      SNR(2,5);
      EDT(2,5);
      FIT(2,5);
    END;


   PROCEDURE NONOISE;

    BEGIN
      NOISE:=FALSE;
      MCOR500:=1;
      MCOR2000:=1;
      STIVALUE;
      DIAGNOSTICS;
      OUTP;
    END;

   PROCEDURE FIXEDNOISE;
    BEGIN
      WRITE ('Give Noise Level Octaveband 500Hz ');
      READLN (LN500);
      WRITE ('Give Noise Level Octaveband 2000Hz ');
      READLN (LN2000);
      NONOISE;
      NOISE:=TRUE;
      NOISECOR;
      STIVALUE;
      OUTPNOISE;
    END;

   PROCEDURE VARNOISE;
   VAR
```

```
    I:INTEGER;
    LEV:REAL;

   BEGIN
     NONOISE;
     NOISE:=TRUE;
     FOR I:=1 TO 10  DO
      BEGIN
EV:=20+I*5;
N500:=LEV;
N2000:=LEV;
OISECOR;
TIVALUE;
UTPNOISE;
      END;
   END;

  PROCEDURE PRINTERON;
  VAR
    I:INTEGER;

   BEGIN
     IF PRINTENABLE
      THEN
      CLOSE(PRINTER);
     WRITE ('Printer on=1, off=0, choice = ' );
     READLN (I);
     IF (I=1)
      THEN PRINTENABLE:=TRUE
      ELSE PRINTENABLE:=FALSE;
     IF PRINTENABLE
      THEN
       BEGIN
ASSIGN(PRINTER,'LPT1:');
REWRITE(PRINTER);
       END
      ELSE
      INPUTFILE:=KEYB;
   END;

  {MAIN}
 BEGIN
   WRITELN ('   Program CONMTF version 1,1 26 dec 1984');
   WRITELN ('H.J.M. Steeneken Institute for Perception TNO');
   WRITELN ('          Soesterberg The Netherlands');
   PRINTENABLE:=FALSE;
   INPUTFILE:=KEYB;
   MENU;
    REPEAT
     WRITE ('Mode=');
     READLN (MODE);
     IF MODE IN [1..7]
      THEN
      CASE MODE OF
.:INPUT;
:FILEREAD;
:NONOISE;
:FIXEDNOISE;
:VARNOISE;
:PRINTERON;
':
       END
      ELSE
      MENU;
    UNTIL MODE=7;
 END.
```

News from the Factory

Speech Transmission Meter Type 3361



For objective assessment of speech intelligibility in auditoria, theatres, schools and industry, Brüel & Kjær has developed a speech transmission meter based on measurement of the Speech Transmission Index according to the RASTI method.

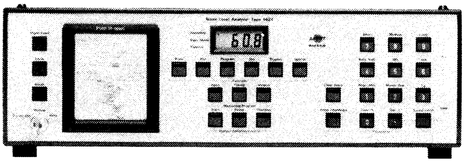
Applications include optimizing of speech reinforcement systems, assessment of public address systems and investigation of acoustical privacy.

The Brüel & Kjær Speech Transmission Meter Type 3361 allows the measurement of an index of speech transmission to be made in less than 10 s. The index derived is the Rapid Speech Transmission Index (RASTI). The RASTI method is being standardised by the IEC (IEC draft pub. 268 part 16).

The Speech Transmission Meter Type 3361 consists of two instruments: Transmitter Type 4225 and Receiver Type 4419. To make a measurement the Transmitter is placed at the speaker's position and emits a special acoustic test signal from the built-in loudspeaker. This test signal conists of a pink noise carrier signal (octave bands centred at 500 Hz and kHz) which is intensity modulated by a sum of low frequency sine vaves. The Receiver is placed at the listening position of interest and analyses the incoming signal, measuring the reduction in modulation lepth for each of the modulating frequencies. This measured reduction n signal modulation is converted to an index of speech intelligibility. The esulting index – RASTI – varies between 0 and 1 and has been found to orrelate well with the results of traditional subjective methods which ise teams of speakers and listeners.

n addition to the RASTI-value, the Type 3361 provides the following nformation: a speech transmission index for each octave band of the arrier signal, the modulation reduction factor for each modulation requency (there are 9 modulation frequencies between 0,7 Hz and 11,2 Hz) and the estimated values of S/N ratio and reverberation time early decay time) which alone would have resulted in the measured eduction in signal modulation.

Noise Level Analyzer Type 4427



A new self-contained portable Noise Level Analyzer introduced by Brüel & Kjær offers a wide range of features for accurate on-site analysis of ommunity noise, airport and traffic noise or any other acoustical event equiring accurate measurements and extensive statistical analysis of ollected data.

The Brüel & Kjær Type 4427 Noise Level Analyzer represents an innovaive design concept, complying with the relevant sections of IEC 651 and ANSI S 1,4 (1983) Sound Level Meter Specification Type 0. It permits ast, user-friendly dialogue selection of instrument settings and provides lata collection, storage, level analysis and print-out in one compact unit. Time-saving menu-driven procedures allow easy interactive instrument set-up, reducing the need for instruction manuals. Sophisticated dataprocessing facilities incorporated in the 4427 allow comprehensive frontend processing of signal data.

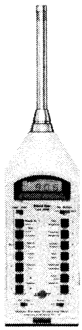
The detector circuit provides F, S, I and Peak plus 3 s and 5 s Takt-Maximalpegel responses in parallel with True Linear 1 s L_{eq} responses. A built-in IEC/IEEE or optional RS-232 C communication interface port provides for remote set-up and control with the same ease as operating the frontpanel keypad.

The LMS detector dynamic range of 110 dB ensures that no information from the input signal is lost, and a wide range of levels can be measured with extreme accuracy.

A built-in graphic printer allows fully annotated permanent records to be made on metallised paper.

Powered by batteries, the Noise Level Analyzer offers this unique combination of features in a compact unit ideally suited for field operation.

Modular Precision Sound Level Meter Type 2231



The Brüel & Kjær Modular Precision Sound Level Meter Type 2231 meeting Type 1 accuracy specifications, sets new standards for versatility and convenience. The measurement applications are numerous, ranging from industrial noise and community noise to architectural acoustics and research and development.

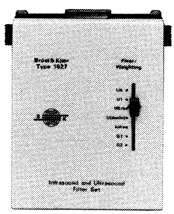
The outstanding feature of the 2231 is its modular construction. The Sound Level Meter derives its many measurement capabilities from a series of interchangeable Application Modules. Whenever a different module is loaded the instrument software is reformatted, enabling it to perform the necessary measurement functions. Three Application Modules are currently available. As standard the 2231 is supplied with Application Module BZ 7100, which is an Integrating Sound Level Meter

Module. Application Module BZ 7101 is Statistical Analysis module allowing measurement of L_N, Cumulative Distribution, and Probability Distribution. Application Module BZ 7102 is a "Taktmaximal" module allowing measurements according to the German "TA-Lärm".

A selectable polarization voltage allows the use of almost any microphone in the B & K range, and further increases the measurement possibilities. The standard microphone allows measurements in the range from 24 dB to 130 dB in seven overlapping 60 dB ranges. Results are displayed on an advanced Liquid Crystal Display which includes a quasi-analogue scale and alphanumerics to be displayed clearly. The soft-touch controls give full tactile feedback, but ensure almost silent operation.

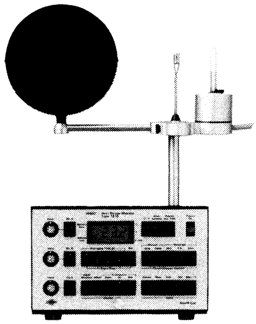
1/1 octave and 1/1 and 1/3 octave band analysis respectively are possible. With Filter Sets Type 1624 and 1625. A new Filter Set Type 1627 takes advantage of the extended frequency response of the 2231, and allows measurements in the infrasound and ultrasound ranges. AC and DC Output sockets allow chart or tape recording of the sound signal.

Infrasound and Ultrasound Filter Set Type 1627



The new Infrasound and Ultrasound Filter Set Type 1627 from Brüel and Kjær is intended primarily for use with the new Modular Precision Sound Level Meter Type 2231. Its six filter networks include G1 and G2 which allow infrasound measurements in accordance with ISO/DIS 7196. Measurements of audible sound in the presence of ultrasonic noise are possible with filter network U (Audio) which complies with IEC TC 29-169/WG 16. The other filter networks are: U1, a proposed weighting which approximates the sensory perception of ultrasound; Ultra, a 12,5 kHz high pass filter; Infra, a 20 Hz low pass filter. The Filter Set also has a Linear setting. Included with the Filter Set is a special microphone adaptor which extends the useful low frequency range of the standard 1/2 inch microphone to below 1 Hz.

WBGT–Heat Stress Monitor Type 1219



Brüel & Kjær WBGT–Heat Stress Monitor Type 1219 is a handy, easy-to-operate, portable instrument for measurement of the WBGT-index (Wet Bulb Globe Temperature) in accordance with ISO 7243. The WBGT-index has long been used as a guide to the level of heat stress on working man in hot environments such as steel works, bakeries etc. Type 1219 comes equipped with a WBGT Transducer which contains sensors for measurement of Wet Bulb-, Air- and Globe Temperatures. The Monitor has a built-in non-volatile memory which stores up to 60 values of each individual parameter, plus mean, maximum, minimum values.

In addition, Type 1219 calculates the "maximum 1 hour mean WBGT" value. Recorded data is automatically measured at equal intervals throughout the selected recording period, and a choice of four (1, 2, 4 and 8 hours) recording periods is available. Type 1219 also indicates the start time at which the "maximum 1 hour mean WBGT" value occurred. Recorded data can be replayed manually on the display or output to an X-Y recorder.

Display and output of data in either degrees Centigrade or degrees Fahrenheit are possible. Type 1219 also calculates the modified WBGT specified by ISO 7243 for situations where there is a high solar load. A weighted WBGT can also be determined by the Heat Stress Monitor when three WBGT Transducers have been connected to the instrument. Alternatively, the three inputs can be used to simultaneously obtain three complete sets of data from different locations.

The WBGT–Heat Stress Monitor can be ordered as a separate unit together with basic accessories and one WBGT Transducer MM 0030. Alternatively, a complete set – WBGT–Heat Stress Monitor Set Type 3531 – may be obtained.

PREVIOUSLY ISSUED NUMBERS OF BRÜEL & KJÆR TECHNICAL REVIEW

'Continued from cover page 2)

3-1979 The Rationale of Dynamic Balancing by Vibration Measurements.
Interfacing Level Recorder Type 2306 to a Digital Computer.
2-1979 Acoustic Emission.
1-1979 The Discrete Fourier Transform and FFT Analyzers.
4-1978 Reverberation Process at Low Frequencies.
3-1978 The Enigma of Sound Power Measurements at Low Frequencies.
2-1978 The Application of the Narrow Band Spectrum Analyzer Type 2031 to the Analysis of Transient and Cyclic Phenomena.
Measurement of Effective Bandwidth of Filters.
1-1978 Digital Filters and FFT Technique in Real-time Analysis.
4-1977 General Accuracy of Sound Level Meter Measurements.
Low Impedance Microphone Calibrator and its Advantages.
3-1977 Condenser Microphones used as Sound Sources.
2-1977 Automated Measurements of Reverberation Time using the Digital Frequency Analyzer Type 2131.
Measurement of Elastic Modulus and Loss Factor of PVC at High Frequencies.
1-1977 Digital Filters in Acoustic Analysis Systems.
An Objective Comparison of Analog and Digital Methods of Real Time Frequency Analysis.

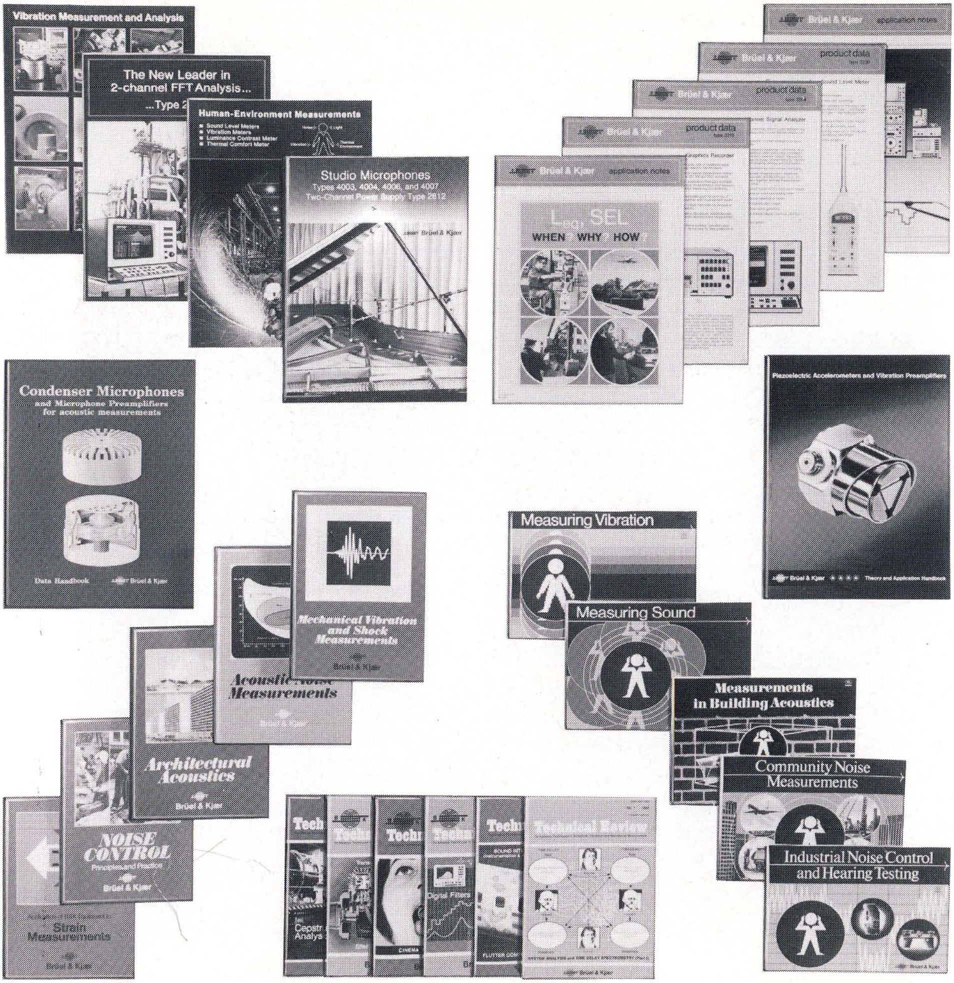
SPECIAL TECHNICAL LITERATURE

As shown on the back cover page, Brüel & Kjær publish a variety of technical literature which can be obtained from your local B & K representative.
The following literature is presently available:

- Mechanical Vibration and Shock Measurements (English), 2nd edition
- Acoustic Noise Measurements (English), 3rd edition
- Architectural Acoustics (English)
- Strain Measurements (English, German)
- Frequency Analysis (English)
- Electroacoustic Measurements (English, German, French, Spanish)
- Catalogs (several languages)
- Product Data Sheets (English, German, French, Russian)

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

Printed in Denmark by Nærum Offset



BV 0019-11

Brüel & Kjær

DK-2850 NÆRUM, DENMARK · Telephone: + 45 2 80 05 00 · Telex: 37316 bruka dk