Proceedings of the Institute of Acoustics

OBJECTIVE SPEECH INTELLIGIBILITY TESTING OF SOUND SYSTEMS

Peter Mapp

Peter Mapp Associates, Colchester, C03 4JZ. [E mail : petermapp@btinternet.com]

The paper examines and reviews the most commonly used, acoustically based, Indirect speech intelligibility measures such as Articulation Index, % Alcons, STI and Rasti as well as the lesser used early energy fractions such as C7 and C50. Their uses and more particularly their limitations are discussed. Measurement accuracy and common error mechanisms are highlighted, as are the effects of non-linear electronic and acoustic environments. Erroneous effects that can be caused by modem signal processing and ways of avoiding such problems are also presented. The implications for producing workable and effective standards for sound system intelligibility measurement are discussed.

1 INTRODUCTION

The practical problems associated with carrying out accurate and statistically meaningful word tests when commissioning and accepting sound systems are well known. For many years, the advantages of being able to carry out some form of objective electro-acoustic test to measure the potential speech intelligibility of a sound system have long been recognised. The advent in recent years of the computer based analyser has opened up the possibilities of carrying out highly complex analyses hitherto unavailable outside the research laboratory. Early energy ratios and Direct to Reverberant acoustic energy ratio assessments have become viable and commonplace 'on site' as have Modulation Transfer Function measurements and Speech Transmission Index assessments. In the USA the Percentage Loss of Consonants (% Alcons) tends to be the most popular method of assessment whilst in Europe either Rasti or its more complex parent - the full STI method, is far more widely used. In addition, more straightforward Direct to Reverberant ratio measurements such as C50, C35 and C7 also exist. Each of the methods has its own distinct limitations but these are rarely discussed. In Europe, standards now exist which require the intelligibility of sound systems, used for emergency and Life Safety purposes, to be measured and rigorous pre-set criteria achieved. The economic cost of failure can be very high - particularly if the non compliance of the sound system delays the opening of a venue or a building. The potential for liquidated damages and litigation is immense. Yet the assessment methods are by no means immune to operator error or 'manipulation'. In the author's experience it is quite possible to manipulate the measurement procedure or data such as to convincingly either improve or degrade an apparent result. Equally it is all too easy to inadvertently or unintentionally corrupt the measurement by not understanding the potential pitfalls of these apparently automatic measurements [1].

2 BACKGROUND TO ASSESSMENT METHODS

It is worth remembering the following two basic facts as they are fundamental to our understanding of speech and its potential measurement.

Speech is a complex auditory stimulus varying in a complex way as a function of both frequency and time.

No one single aspect of the speech waveform is essential for speech perception.

It is surprising but even today, the way in which speech is processed by the ear-brain mechanism is far from being fully understood. It is clear however, that the brain treats speech in a very different way to other acoustic signals. Therefore other perceptual models related to the way we analyse sound for its loudness, duration, rate of change, spectral and temporal content for example may not directly apply to speech and even less so to the quality of speech termed 'intelligibility'. That all the above do affect speech perception to some degree or other is well known and indeed it is possible to devise psycho-acoustic tests to probe the effect of each. However, the exact way in which each factor affects another when in combination, is still very much open to debate. However, receiving only limited information can, depending upon the exact combination of the circumstances, still lead to the correct answer being deduced. Speech intelligibility has been likened to "a loss of information". If too much information or certain key elements are lost, then the intelligibility will be affected but there is no one factor which determines the perceived intelligibility. As there is considerable redundancy built into normal speech, it is quite possible to loose considerable amounts of information before intelligibility is lost - however it is this exact combination of potential losses of highly interactive elements and components, that make predicting (and hence measuring) the resultant intelligibility an almost impossible task. Whereas at first it would seem almost impossible to create an indirect objective measurement system, in practice surprisingly good results can be achieved - even with relatively simple methods. However, there are limitations and each of the common methods only works within a fairly narrow tolerance window - a fact which unfortunately has either been forgotten or not understood by many. In order to devise (or understand) a method for measuring the potential intelligibility of a sound system, it is important to have a basic knowledge of the main factors that can and do affect intelligibility. From this basis it is then possible to see under which conditions the various methods available may operate effectively.

PRIMARY FACTORS AFFECTING SPEECH INTELLIGIBILITY

BANDWIDTH
FREQUENCY RESPONSE
LOUDNESS
SIGNAL TO NOISE RATIO
REVERBERATION TIME (DIRECT TO REVERBERANT RATIO)
LISTENER ACCUITY
TALKER ANNUNCIATION & RATE OF DELIVERY
TALKER VOICE TYPE MALE, FEMALE

SECONDARY FACTORS

DISTORTION (eg THD)
SYSTEM NON LINEARITIES & COMPRESSION
SYSTEM EQUALISATION
UNIFORMITY OF COVERAGE ECHOES
REFLECTIONS & REFLECTION DIRECTION
DIRECTION OF SOURCE (RELATIVE TO LISTENER)
DIRECTION OF INTERFERING NOISE
VOCABULARY
CONTEXT

In order to accurately predict the intelligibility of speech it is necessary for a measurement system not only to take account of each of the above factors but also to deal with them in combination and not as a series of individual parameters. Current electronic measurement systems fall well short of these goals.

3 DIRECT MEASUREMENT METHODS

The only truly accurate and direct method of measuring speech intelligibility is to carry out an objective (scored) word test. A number of methods are currently used ranging from the relatively simple MRT Modified Rhyming Tests to Phonemically balanced (PB) word tests. Apart from factors related to the talkers, listeners and a given sound system, there are two factors common to all subject based intelligibility tests that have a significant influence on the scores obtained from a test evaluation. The two factors are : (1) the speech material employed and (2) for a given type of speech material, the total number of alternative members of that material the listeners expect to be presented with during a test. Test formats may either be in the form of Large sets perhaps comprising 1000 meaningful test phonemically balanced words or 650 Logatoms (monosyllabic or polysyllabic test speech sounds that have no meaning to the listeners). Usually, open test lists or pseudo open test lists are employed whereby the listener writes on an answer sheet each test item that the listener believes was presented. Alternatively, a small closed set (eg M RT test) may be used. MRT tests are very much simpler to carry out than PB word tests and in comparison require minimal participant training. They also offer good test - retest reliability. However, as with all subject based testing, it is essential that the listening panel is audiologically screened and the tests are conducted in the participant's first language. Other basic precautionary measures such as ensuring listening fatigue does not set in and randomised presentation of the test sequence also need to be taken to ensure statistically valid results. The results can be recorded in a number of ways but it is common practice to employ multiple choice test sheets or write down word lists, where the test word embedded in a carrier sentence is selected from a group of limited choice options. Using a carrier sentence has a number of advantages as for example it enables the reverberant field to be excited by the preceding words and the effects of room reverberation therefore to be more accurately accounted for. The sentence structure also provides the talker with a means of enunciating the words in a natural manner with a controlled and measurable level of effort.

The "steady" stream of test sounds also allows dynamic or temporally based devices such as automatic gain controls or compressors etc to operate in their normal manner. It is essential however that the test word is correctly presented in terms of its sound level (amplitude) with respect to the carrier sentence. Training of both the listeners and test talkers is essential and should not be underestimated. For example it is generally anticipated that for small closed set word tests, 5 to IO minutes training of the listeners and talkers is usually sufficient. However, with large open set tests, talker training may take up to an hour and listener training several hours. The useful test-life of a trained panellist is also limited. The need and advantages of an automatic, electronically based measurement system are clearly obvious.

4 INDIRECT TEST METHODS

A number of indirect test methods have been developed to assess the effects of noise and reverberation on speech. With these methods, by measuring an appropriate parameter or set of parameters the likely potential intelligibility of the sound system operating under those given conditions is then predicted. The correlation between the signal to noise ratio or direct to reverberant ratio and the likely intelligibility, for example, being based on previously established criteria.

4.1 ARTICULATION INDEX (AI)

In many communication systems, noise is the primary speech intelligibility degradation factor. (eg radio and telephone communication systems or PA systems in noisy environments.) One of the first standardised methods established for assessing intelligibility under noisy conditions was the Articulation Index. This was based on the original 1947 work of French & Steinberg and developed by Kryter, Baranek and others and published as an Ansi Standard in 1969. (S 3.5 1969). The AI concept is that speech intelligibility is proportional to the average difference in dB between the masking noise level and the long term average speech level (plus 12 dB to allow for speech peaks), taken at either 1/3 or 1/1 octave centre frequencies. The resultant SIN ratios are then weighted and combined to form the Articulation Index. Scores range from 0 to 1 with 0.3 and below being rated as unsatisfactory, between 0.3 and 0.5 as satisfactory, 0.5 to 0.7 as good and greater than 0.7 as very good to excellent.

One of the enduring factors that emerged from French and Steinberg's early work was the isolation of the relative contributions of each frequency band to intelligibility [2]. The importance of the higher frequency consonants immediately became apparent with the 2 kHz octave band for example contributing over 30 % to the total score (see figure1). Although a number of modifications to the basic Articulation Index procedure have been developed to take account of factors such as reverberation, these have not proved particularly successful.

4.2 SPEECH INTERFERENCE LEVEL (SIL)

This is another noise based method but employing a simpler procedure than the Articulation Index. Here the noise level in the 500, 1k, 2k and 4k octave bands is measured and the arithmetic mean taken and related to a table of maximum satisfactory communication distances.

4.3 PERCENTAGE LOSS OF CONSONANTS (% ALCONS)

The concept of the Percentage Loss of Consonants was first proposed by Peutz over 28 years ago in 1971 [3]. It was not until 1986 however that a method of making a measurement that directly correlated with % Alcons was formulated. This employed the Techron TEF machine. Essentially the Direct to Reverberant ratio of the sound system transmitted acoustic signal is measured together with the Early Decay Time. From these parameters the TEF computes the % Alcons score which is based on a set of correlations three different acoustic environments with a total listening panel size of almost 100 [4].

Figure 2 shows a typical TEF % Alcons plot. The Direct Sound component, towards the left hand side of the graph, clearly dominates the situation being some 20 dB higher than the later reflections and reverberation. A visual examination of the graph suggests that this system will be highly intelligible. However when the early decay time is evaluated (fig 2b) this is found to be 2.6 seconds, a fairly high value and one that could readily degrade intelligibility. However evaluating the Direct to Reverberant ratio for the signal produces a value of + 8.7 dB which results in an overall Percentage Loss of Consonants score of 4.2 % Alcons. This is equivalent to 0.68 Rasti and would be rated as good to very good.

Figure 3 shows a similar plot taken under exactly the same circumstances in the same building but this time with a less directional source loudspeaker. The direct sound component is only just discernible from the reflected and reverberant field. A visual inspection suggests questionable intelligibility. Evaluation of the % Alcons shows - the Direct to Reverberant ratio to have now dropped to minus 4.2 dB with an equivalent % Alcons score of 13 % which is marginally acceptable. (equivalent to 0.48 Rasti or 'poor' subjective rating). Whereas the TEF % Alcons method has the advantage of enabling the impulse response [Energy Time Curve (ETC)] to be seen and hence reflections to be evaluated, it does suffer from a number of drawbacks. (1) Although in the current TEF implementation the process is semi automated, the algorithm used can be readily caught out and misleading results produced which might not be readily identifiable to the inexperienced user. A skilled operator is therefore required. (2) The Alcons measurement is only based on the 1/3 octave frequency band centred at 2 kHz (ie in practice from around 1800 to 2200 Hz). The behaviour at other frequencies is not evaluated. Whereas large format CD horns and equivalent devices that exhibit a Q factor and coverage characteristics which may not vary significantly with frequency over the upper speech frequency range (eg 1,000 to 8,000 Hz) many other common loudspeaker devices do and carrying out a measurement at just one frequency can give rise to misleading results, which are generally tend to be over optimistic in nature. (3) The method does not take into account the following factors : Background noise and S/N ratio or frequency spectrum of background noise, effects of distortion or system non linearities, system frequency response, bandwidth and equalisation, late discrete (isolated) reflections and echoes, direction of sound and reflections.

Furthermore with distributed sound systems it is difficult to establish the correct division point for the Direct & Reverberant components, making reliable & repeatable measurements difficult [3,4].

4.4 OTHER DIRECT TO REVERBERANT RATIO MEASURES

A number of simpler 'Direct to Reverberant' ratio measures are also in use, particularly for the assessment of natural speech and music in auditoria. Of these the most popular is C50, whereby the energy ratio of the first 50 mS of 'direct' sound to the overall reverberant sound is taken as a measure of speech clarity. Ratios of at least 0 dB should be aimed for and + 4dB upwards required for good intelligibility. Whereas it has been shown that there is good correlation between the C50 index and speech intelligibility, a well defined scale has not been agreed. Measurements are usually confined to the 1kHz octave but there is no reason why other bands should not be used and some researchers use an averaging technique to produce a single combined result. Because the transition point between the Direct and Reverberant components is defined, measurement reliability and repeatability are generally better than for % Alcons measurements.

C7 is another measure sometimes used - particularly in Germany. For adequate intelligibility a value > -15 dB is required. C35 is also another variation of this measure. For the reverberant church situation shown in figures 2 & 3 the following 'C' values were obtained.

	% Alcons	**C50**	**C35**	**C7**
High Q source	4.2 %	9.9 dB	8.9 dB	-2.1 dB
Low Q source	13 %	-3.6 dB	-7.0 dB	-19.0 dB

As with % Alcons, the above measures require or assume a high signal to noise ratio and are purely assessments of the effect of reverberation on intelligibility but based on a single frequency band. Whereas this approach is of use when working with speech transmission in natural acoustic environment, the potentially non uniform frequency response, frequency dependent Q and nonlinearity of a sound systems in practice, reduces the usefulness and universality of such measures.

Recently, a rather more sophisticated approach has proposed by Marshall [5] termed ELR or Early to Late Sound Energy Ratio. This is based on the C50 concept but uses a speech weighted scale using a modified version of the French & Steinberg factors. Measurements are made at the main speech frequencies of 500, 1kHz, 2 kHz & 4 kHz and their contributions weighted respectively by 15, 25, 35 and 25 %. The method overcomes the problem of the single frequency band measurement but does not take into account background noise or other sound system non linearities.

4.5 SPEECH TRANSMISSION INDEX (STI)

Also dating back to 1971, [6] the Speech Transmission Index did not really receive any recognition until 1985 when Bruel & Kjaer brought out their Rasti meter. This, for the first time, promised to be the magic box that would enable simple speech intelligibility measurements to be made without the need for a high degree of operator skill or complex equipment. The Rasti (Rapid Speech Transmission Index) scale was developed as at that time portable computational power was limited. Instead of carrying out the full 98 point measurement matrix based on 14 modulation frequencies over the 7 carrier bands of 125 Hz to 8 kHz, a total of just 9 modulation frequencies are employed spread over the 500 Hz and 2 kHz bands.

The Rasti / STI concept is an elegant one and employs a test signal with many speech-like qualities. The concept is based upon the fact that speech is effectively a modulated waveform with modulation frequencies in the range from approximately 0.5 Hz to 15 Hz modulating the normal carrier frequency range of 125 Hz to 8 kHz. Houtgast and Steeneken found that there was a good correlation between the reduction in modulation depth and speech intelligibility. Again, a measurement of a loss of information.

Figures 4 and 5 illustrate the STI / Rasti concept. STI has a number of advantages over the other methods so far described as it is able to take account of both noise and reverberation effects. A further advantage of the STI method is that the process is totally defined and does not rely on an operator to position cursors or make other acoustic judgements.

Although the method does not inherently produce a plot of the impulse response or ETC, many implementations enable this function to be obtained if required. Apart from producing the single number index, the STI matrix provides much useful information and allows the experienced operator to carry out a number of diagnostic measures enabling, for example, indications to be obtained as to the nature of the potential intelligibility reduction eg due to noise or reverberation. The Rasti method, as implemented by B & K, provides a very portable system that allows rapid intelligibility checks to be made both directly within an auditorium or via a sound system. Being readily available before the full STI implementation, Rasti has been adopted by a number of standards requiring the potential intelligibility of a sound system to be specified and verified. The first of these was the European Standard IEC 849 published in 1989 specifying 'Sound Systems for Emergency Purposes'. (Revised 1998). A number of european national and city authorities and consultants have also adopted the Rasti rating method as for the first time it apparently allowed the intelligibility of a sound system to be unambiguously (?) specified and then verified. The method has also been adopted for specifying and verifying the performance of Aircraft Passenger Announcement systems bringing about a significant improvement in the performance of the average system.

5 MEASUREMENT SYSTEM SHORTFALLS

5.1 RASTI

Being a foreshortened method, the Rasti approach does contain a number of inherent compromises and experience over the past 10 years or so does show it not to be the infallible guide we all thought it was going to be [7]. Although Rasti performs measurements at both 500 Hz and 2 kHz it assumes that the sound system response extends from around 200 Hz to at least 6 kHz and is linear. Now whereas a large, high quality system in a theatre, concert hall or church for example would undoubtedly do this, a transportation or industrial installation system based on re-entrant horns would not. So in the latter case a higher value than the true reading could well be obtained. (Typically in the author's experience by at least 10 % or more). Hence a system could be measuring 0.45 Rasti (15 % Alcons) and yet in reality be only 0.40 Rasti (20 % Alcons) a very noticeable difference and one that could take a system from being acceptable to completely unacceptable. By way of illustration of this limitation, the author produced the now infamous 'twin' peaked graph shown in figure 7 [8]. Although the majority of the information is missing, the peaks at 500 and 2 kHz are sufficient to obtain a perfect Rasti score ! [It should be pointed out that the same could be said of the current % Alcons implementation which relies solely on the 2 kHz value].

The Rasti signal is very speech like in that it has a crest factor of around 11 to 12 dB which compares well with typical speech with values around 12 to15 dB. However the signal and process can be affected by non linear elements within a sound system such as compression or limiting for example. This tends to reduce the modulation depth and hence result in a lower overall Rasti value than should be expected. (The actual reduction is highly dependent upon the compressor / limiter setting parameters eg attack and release times and thresholds but apparent reductions of up to 0.1 5 have been noted by the author). [Interestingly, it has also been shown [9 & 10] that appropriately applied, compression can actually improve intelligibility]. Distortion within a system is also not taken into account.

The full STI measurement procedure overcomes many of the above problems and now that modern portable computing power make it a viable option, it is likely to replace Rasti in certain situations - particularly where limited response systems are involved. However until a direct readout STI meter is developed, nothing can beat the portability and immediacy of a dedicated Rasti meter.

5.2 STI

The greater bandwidth capability of the full STI system and improved resolving power of the 98 point matrix produces an inherently more accurate measurement approach. Most commercial implementations employ stimulus signals which are inherently less prone to compression effects than the original Rasti signal and certain equipment should allow distortion to be taken into account - at least to some degree. Although seen by many as approaching the ultimate goal of an operator immune measurement system, It is still possible to readily fool an STI measurement and sound system parameters such as equalisation are not fully accounted for - particularly in the reverberation only case [11]. As with % Alcons and Rasti, strong, late reflections or echoes are not properly accounted for with respect to their effect on system intelligibility [12].

For example a strong reflection arriving at approximately 60 mS can completely null out the true modulation reduction at 8 Hz whereas a reflection at 40 mS will affect the 11.6 Hz modulation. (see figure 8). Figure 9 shows the typical impulse response of a system exhibiting strong, late arriving reflections. The resulting STI matrix is noticeably distorted. AS with most of the measures discussed, an STI measurement is highly dependent upon the sample size (impulse response length) acquired during a measurement. It is possibly to radically change apparent STI values by adjusting the sample acquisition length. [13].

In order to provide an undisputed measurement, procedures within the measurement hardware / software need to be in place to ensure that adequate time information is always included. Although the STI method can accurately take account of background noise by evaluating the individual octave band signal to noise ratios, this is only the case for steady state noise without undue fluctuation. In many situations eg crowd / spectator noise in stadia and arenas, the noise is far from steady in level but a highly significant contributor to potential intelligibility reduction (see figure 10). Although a statistical analysis can be carried out eg calculating the L10 percentile, there is little evidence to relate speech intelligibility and signal to noise ratios determined in this way.

An omnidirectional microphone is normally employed when making intelligibility type measurements. No account is therefore made of the direction of the speech signal or interfering noise. However, recent research has shown directionality to be a significant factor. The directivity of the ear itself may also be a significant factor. For example, Figure 11 presents a measurement made via an omnidirectional measurement microphone and via a human ear. The reduction in the reverberant energy picked up by the ear is clearly visible. This raises the question as to whether an omnidirectional microphone is completely appropriate.

There is considerable evidence to show that male and female voices can produce different intelligibility scores under the same conditions. At present, a single STI speech spectrum is used. A correction factor for male / female voices would seem to be appropriate. Figures 12 (a) & (b) present typical male and female voice spectra whilst figure 13 presents the current STI reference spectrum.

The differences between the three sets of curves are immediately obvious with the female spectra peaking at 250 Hz and exhibiting no content in the 125 Hz octave band. Similarly, the effect that equalisation can have on perceived system intelligibility is quite considerable, yet again the effect that this significant parameter can have is not appropriately accounted for.

This highlights the problem in that the STI method can, under a number of circumstances, give a misleading answer. However to know that the result is incorrect requires considerable experience and skill. Ironically, it is often only when the intelligibility of a sound system is brought into doubt that an objective test will be carried out. Indeed it can be argued that the primary range for sound system testing will lie between approximately 0.4 and 0.6 STI (20 % to 6 % Alcons). Outside these ranges the intelligibility is almost of only academic interest either being totally unintelligible or unquestionably good.

6 SUMMARY OF INDIRECT INTELLIGIBILITY MEASUREMENT ERROR MECHANISMS

At the present time, the following parameters are identified as causing either all or some of the measurement methods a problem : Sample truncation, Non Linear Frequency Response, Non Linear Acoustic environment, (not STI), System non linearities (compression / limiting), Non linear time processing (eg Phase shifters), Distortion, Equalisation, Echoes & Discrete reflections, Tonal Noise, Time varying noise, Absolute sound level of Speech Signal, Male / Female Talkers *, Rate of Speech & Vocal effort 1 annunciation.

[* Recent modifications to the STI procedure take account of male / female talker differences.]

Whilst it is easy to criticise the STI and the other indirect electronic / acoustic test methods, when used within its correct parameters, the STI method can give a very good assessment of potential sound system intelligibility. Care however must be taken to ensure that factors such as signal processing, system non linearities and sample truncation have not affected the result. Whilst work is under way on using real speech as the test source signal and powerful speech recognition programs as the measurement system, these too are easily fooled and are a long way from practical commercial use in evaluating sound system performance. The use of complex stimuli and perceptual models using error plane techniques however does look to be a promising avenue of research that perhaps might lead to a more 'intelligent' and sophisticated measurement approach [14].

7 REFERENCES

[1] P Mapp, Speech Intelligibility Measurements - What was the Answer you wanted ? Institute of Acoustics Conference Reproduced Sound 6, 1990. (Unpublished paper).

[2] French & Steinberg, Factors governing the intelligibility of speech sounds. JASA vol 19 No1 1947.

[3] VMA Peutz, Articulation Loss of consonants as a criterion for speech Transmission in a room. JAES vol No 11 1971.

[4] P Mapp & P Doany, Speech Intelligibility Analysis and Measurement for a Sound- System in a Reverberant Environment. AES 87th Convention New York 1989.

[5] G Marshall, An analysis procedure for room acoustics and sound amplification systems based on the Early to late energy ratio. JAES vol 44 No5 May 1996.

[6] T Houtgast and HJM Steeneken, The Modulation Transfer Function in Room Acoustics as a Measure of Speech Intelligibility. Acustica 28, 1973.

[7] P Mapp, Is Rasti a reliable indicator of sound performance ? AES 96th convention Amsterdam 1994. system

[8] P Mapp, Paper presented at Syn Aud Con Intelligibility 2, Bloomington University 1990.

[9] N Dvorko, K Nizamova, The quality of speech transmission in background of music and effects in audio-visual programs. AES 103rd Convention, Munich,1997.

[10] P Barnett, Implications of Amplitude Compression on Rasti Performance, proc. IOA vol 19 pt 6 (1997).

[11] P Mapp, Some Effects of Equalisation & Spectral Distortion on Sound System Intelligibility. proc. IOA vol 19 pt 6 (1997).

[12] P Mapp, The importance of the Modulation Transfer Matrix in sound system assessment. AES 96th convention Amsterdam 1994.

[13] P Mapp, A comparison between STI and Rasti Intelligibility Measurement Systems. AES 100th convention Copenhagen 1996.

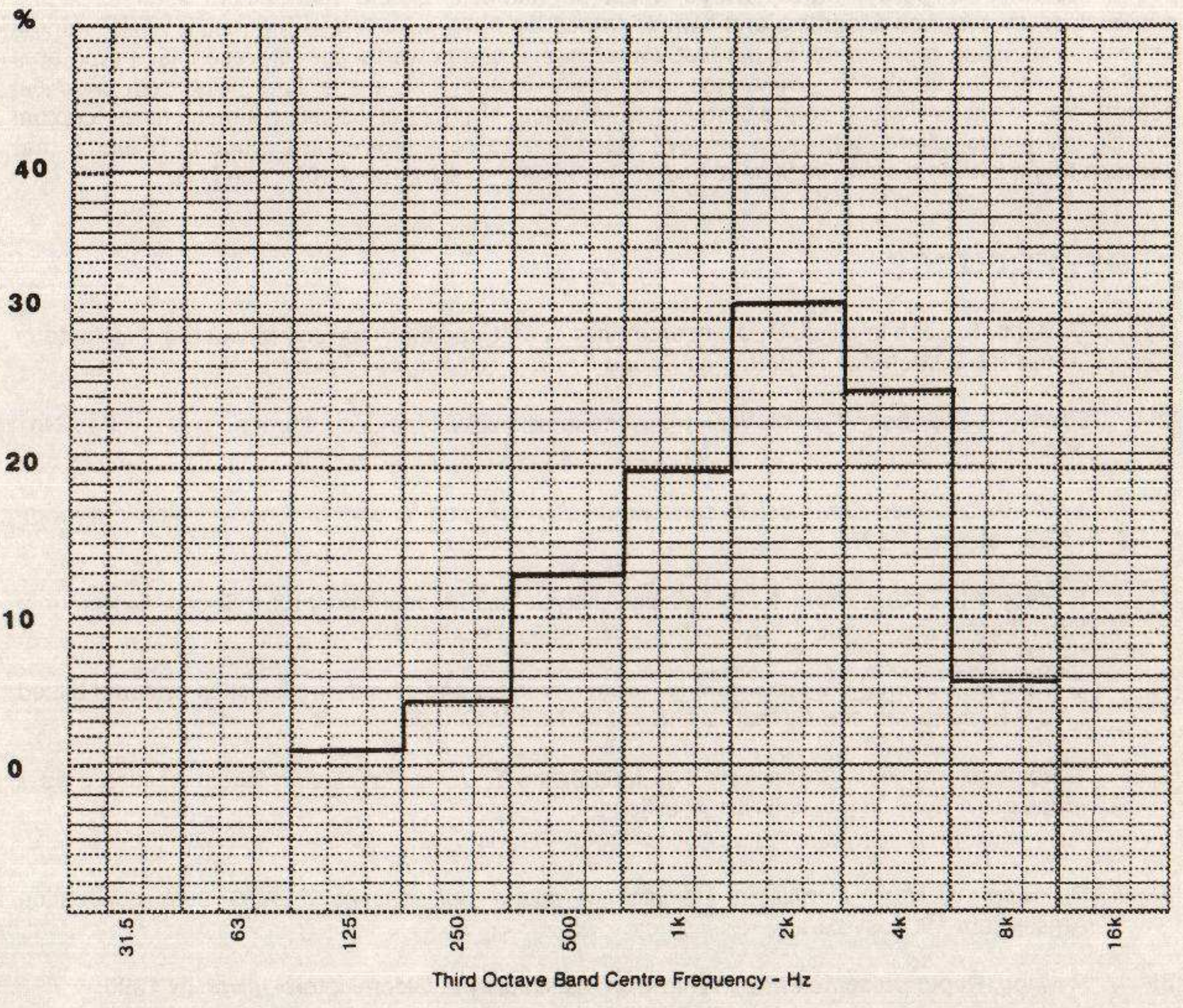


FIGURE 1 OCTAVE BAND CONTRIBUTIONS TO SPEECH INTELLIGIBITY

Peter Mapp

·er Name:
·te:
·cation: St Raymonds Church; Joliet, IL

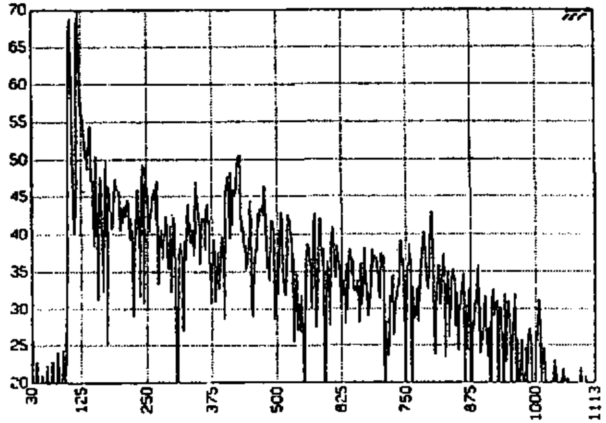


FIGURE 2 (A) DIRECT & REVERBERANT SOUND COMPONENTS

HIGH Q LOUDSPEAKER IN REVERBERANT CHURCH

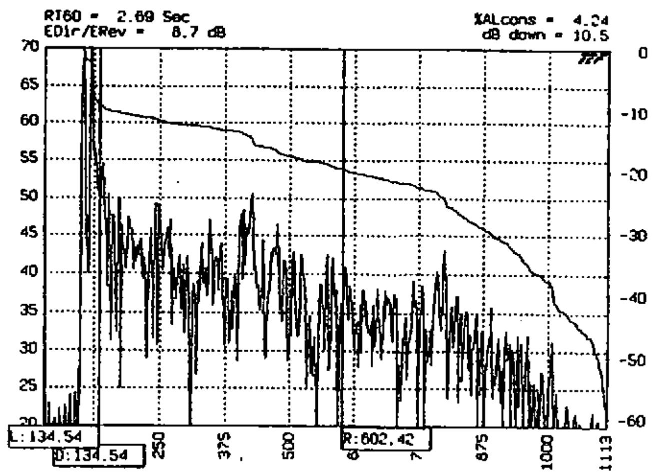


FIGURE 2 (B) % ALCONS ASSESSMENT WITH HIGH Q LOUSPEAKER

Test Title: DES, 2KHz, ETC, BACK

User Name:
Date:
Location: St Raymonds Church; Joliet, IL

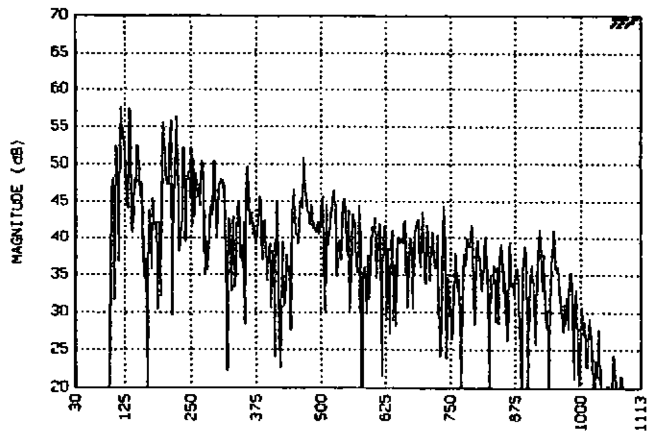
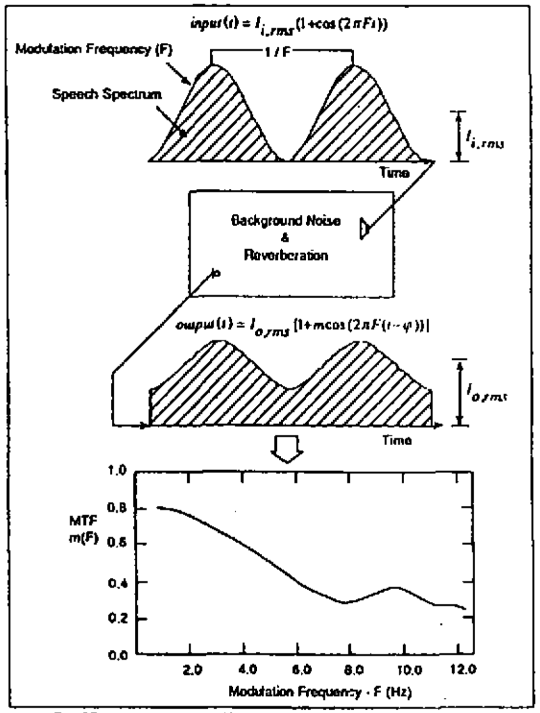


FIGURE 3 DIRECT & REVERBERANT SOUND COMPONENTS

LOW Q LOUDSPEAKER IN REVERBERANT CHURCH



The STI method uses an artificial speech signal modeled after the behavior of actual speech. This signal is an amplitude-modulated speech spectrum. In this figure, the blurring effect background noise and reverberation have on the input waveform is shown. The modulation Index (m) is a measure of the preservation of the original modulation at the input of the system, and the Modulation Transfer Function (MTF) is the modulation index as a function of modulation frequency (F). (Figure attributable to Houtgast and Steeneken

FIGURE 4 STI CONCEPT

Peter Mapp

ACOUSTIC + AUDIO ANALYSIS

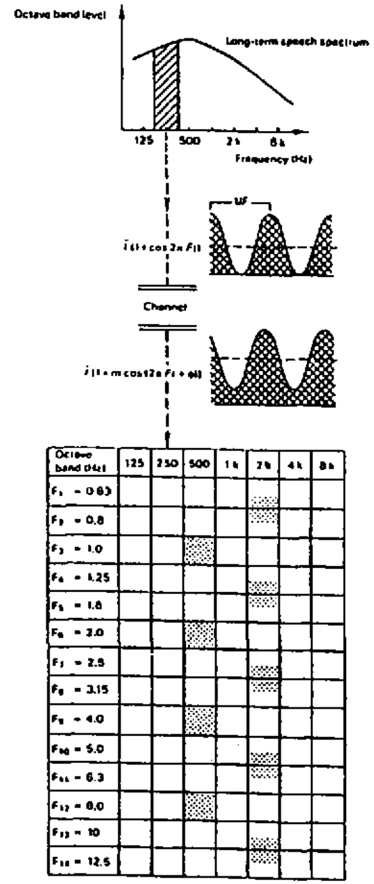


FIGURE 5 FREQUENCIES FOR STI & RASTI METHODS

Proceedings of the Institute of Acoustics

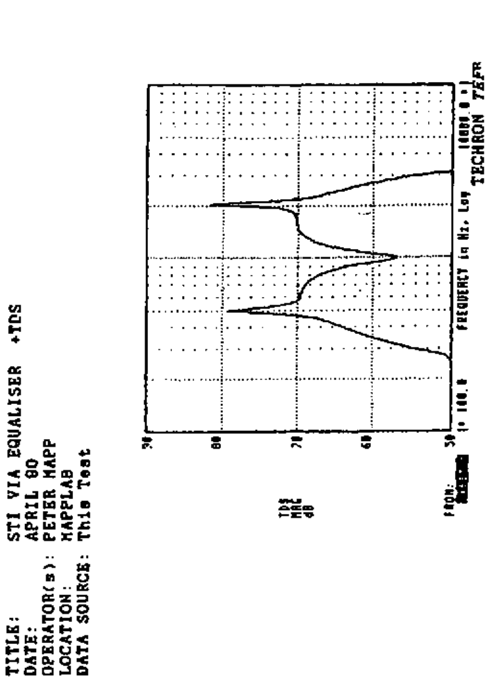


FIGURE 7 ARTIFICIAL SYSTEM RESPONSE DEMONSTRATING HOW TO FOOL RASTI MEASUREMENT

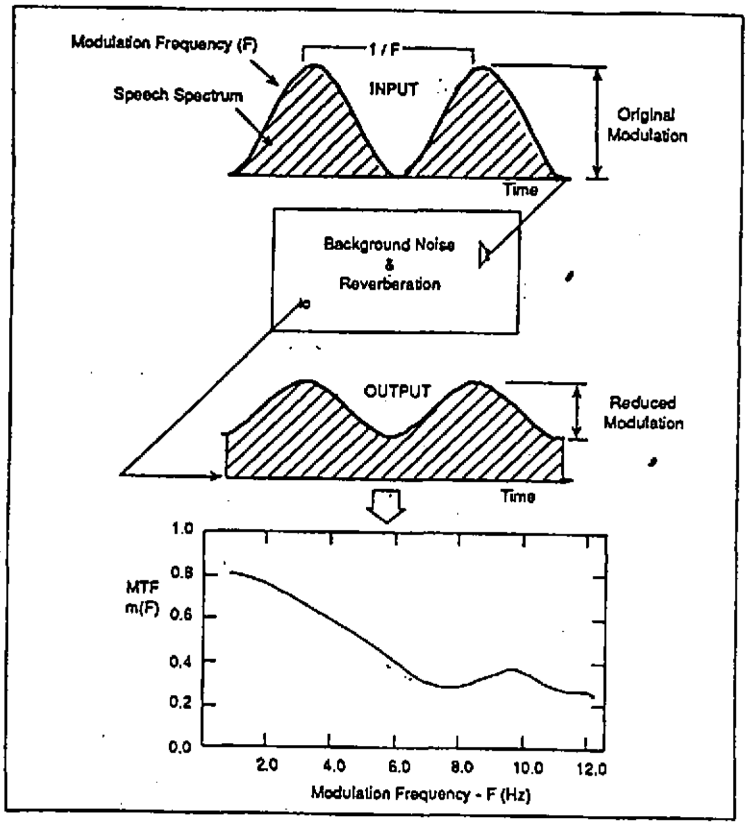
MTF Matrix

Frequency-Hz	125	250	500	1000	2000	4000	8000
level dB-SPL	47.5	56.6	68.8	75.4	75.8	68.0	51.0
m-correction	1.000	1.000	1.000	1.000	1.000	0.998	0.985
0.63	0.059	0.765	0.815	0.857	0.855	0.794	0.463
0.80	0.086	0.695	0.746	0.806	0.815	0.763	0.448
1.00	0.076	0.613	0.670	0.747	0.766	0.726	0.422
1.25	0.016	0.530	0.600	0.690	0.712	0.684	0.382
1.60	0.042	0.434	0.515	0.624	0.643	0.619	0.339
2.00	0.115	0.347	0.426	0.566	0.577	0.564	0.306
2.50	0.040	0.250	0.357	0.527	0.540	0.549	0.289
3.15	0.036	0.204	0.351	0.529	0.575	0.597	0.339
4.00	0.078	0.183	0.425	0.570	0.662	0.688	0.409
5.00	0.041	0.177	0.460	0.577	0.687	0.709	0.414
6.30	0.086	0.088	0.399	0.494	0.560	0.583	0.309
8.00	0.094	0.202	0.379	0.490	0.567	0.590	0.330
10.00	0.072	0.221	0.339	0.530	0.632	0.662	0.369
12.50	0.017	0.144	0.252	0.414	0.466	0.505	0.272
octave TI	0.097	0.391	0.491	0.566	0.594	0.590	0.418

STI value= 0.461 (0.557 modified) ALcons= 14.0% Rating= FAIR

ROW X1 SEAT 99 MID STAND LOCATION

FIGURE 6 TYPICAL STI MEASUREMENT MATRIX



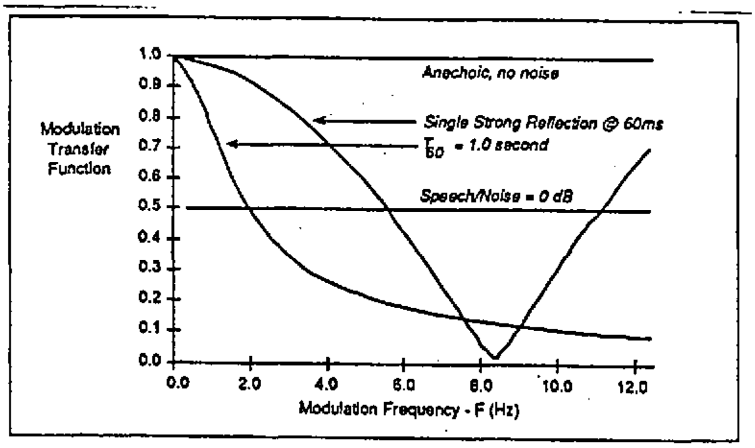
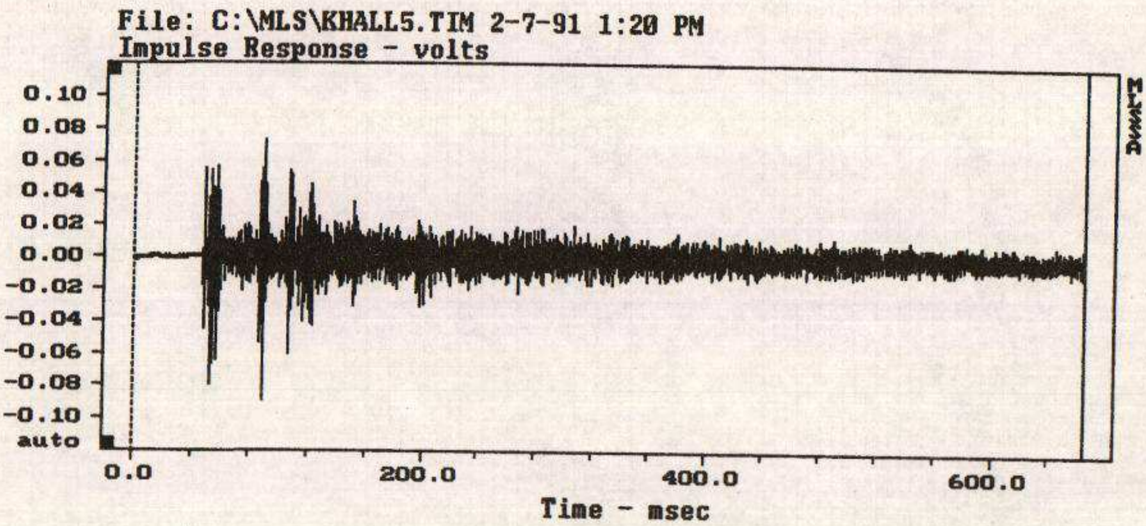


FIGURE 8 EFFECT OF DISCRETE & LATE REFLECTIONS ON STI



Comment: MIC IN CENTRAL AREA

RASTI = 0.319

FIGURE 9 TYPICAL IMPULSE RESPONSE SHOWING STRONG LATE REFLECTIONS

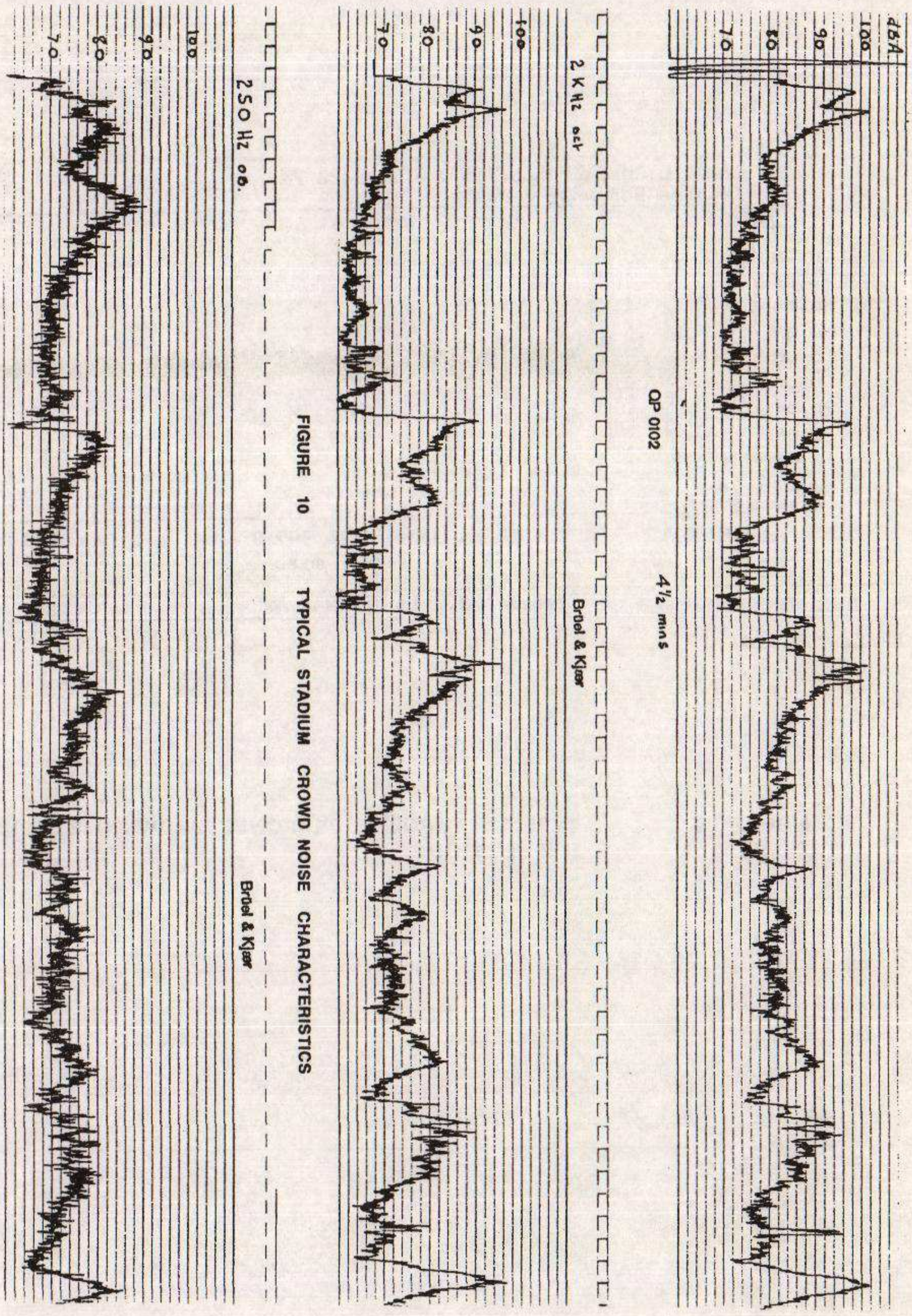


FIGURE 10 TYPICAL STADIUM CROWD NOISE CHARACTERISTICS

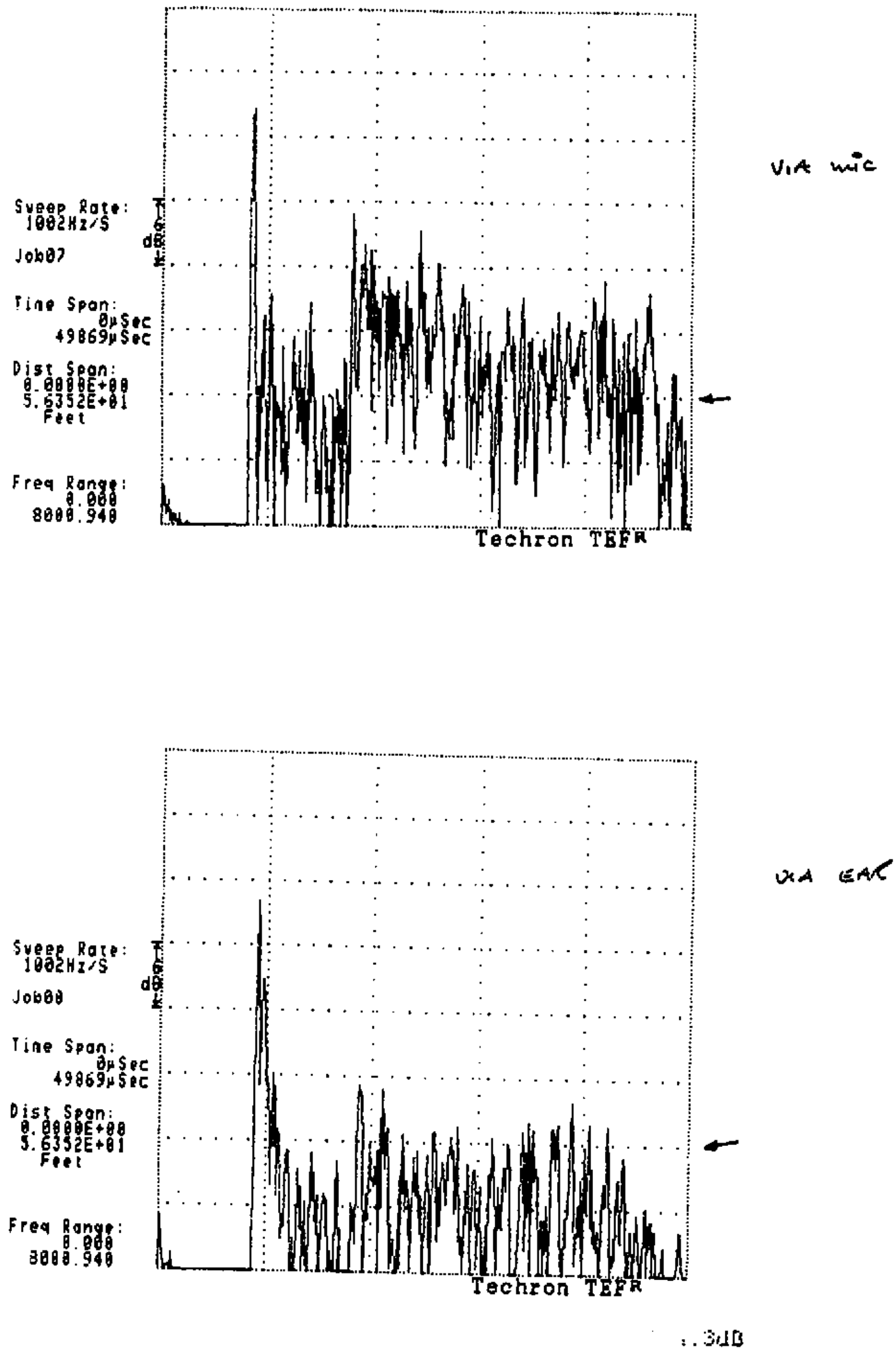


FIGURE 11 ETC IMPULSE RESPONSE MEASUREMENTS MADE VIA AN OMNIDIRECTIONAL MICROPHONE AND VIA THE EAR

Peter Mapp

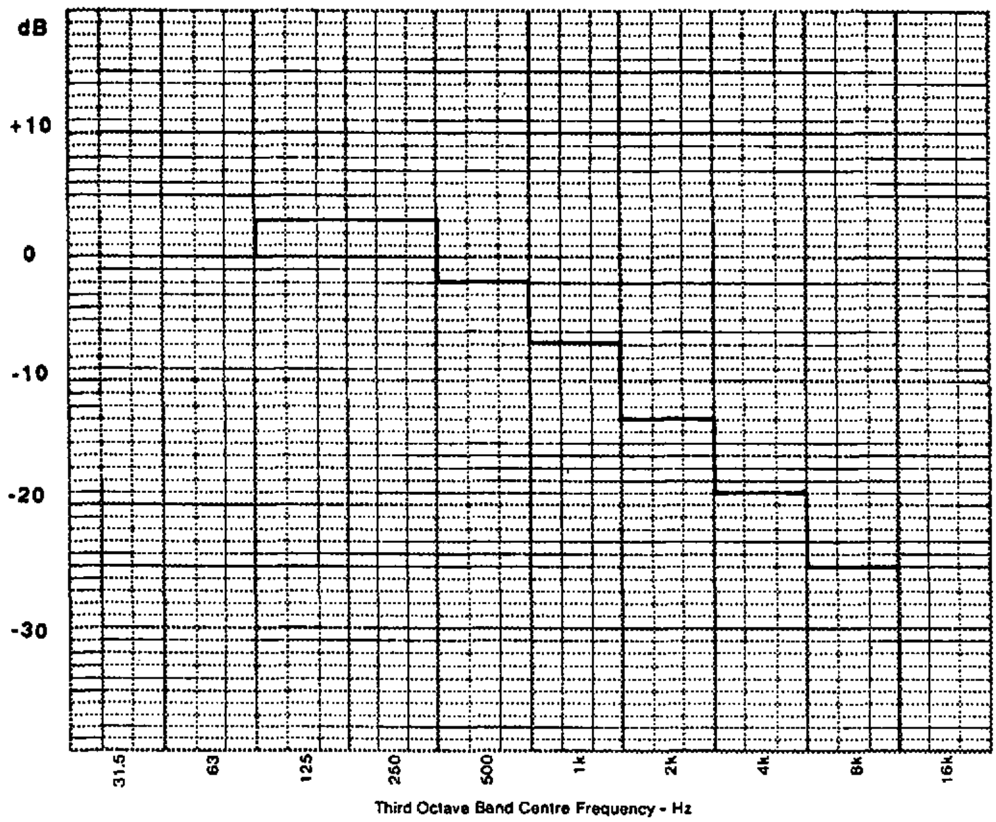


FIGURE 12 (A) TYPICAL MALE SPEECH SPECTRUM

Peter Mapp

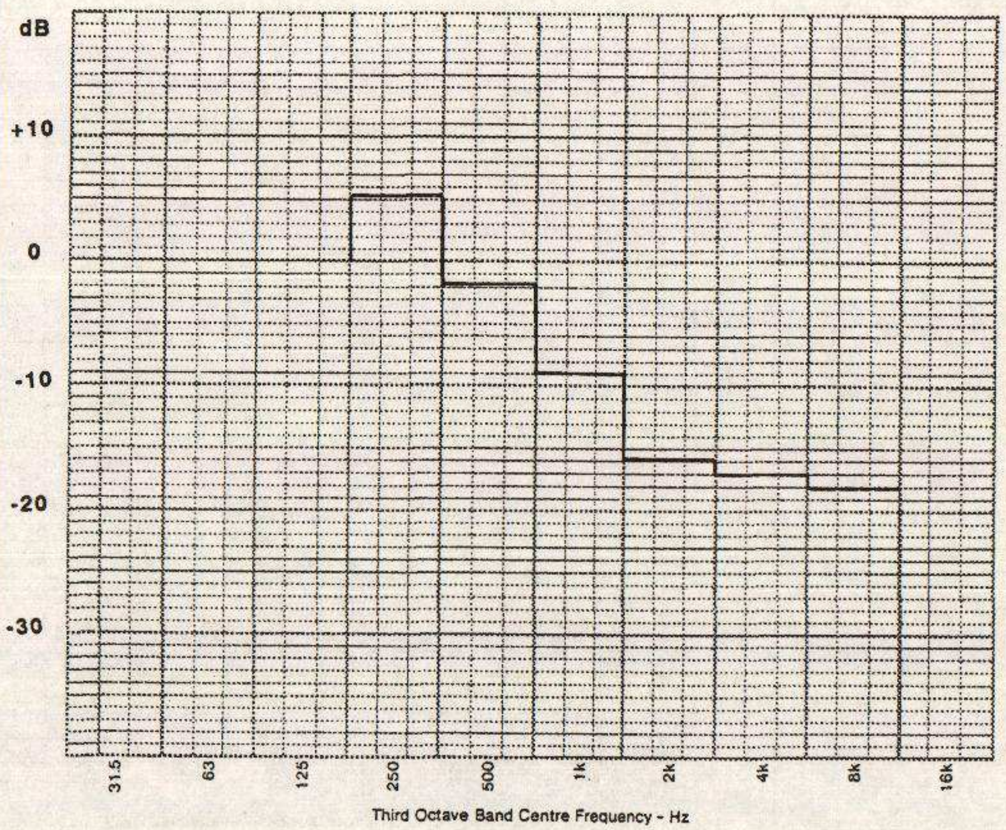


FIGURE 12 (B) TYPICAL FEMALE SPEECH SPECTRUM

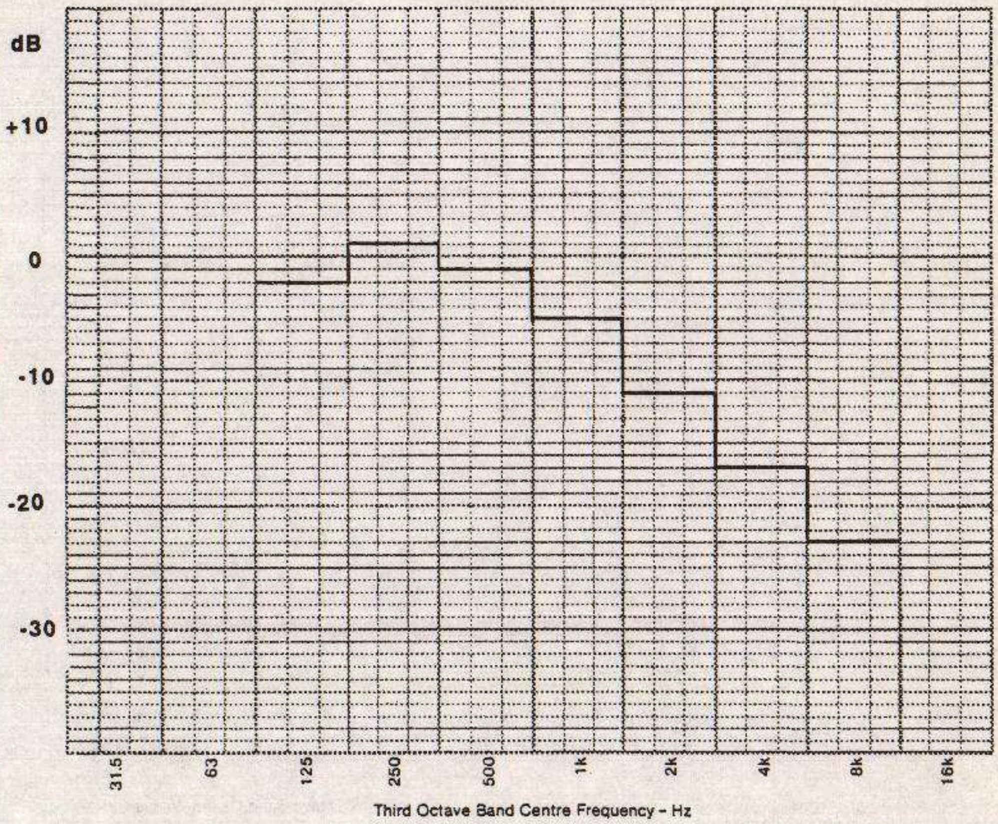


FIGURE 13 CURRENT STI REFERENCE SPECTRUM