TRANSIENT SOUND INTENSITY MEASUREMENTS FOR EVALUATING THE SPATIAL INFORMATION OF SOUND FIELDS IN REVERBERANT ENCLOSURES

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ABSTRACT

TRANSIENT SOUND INTENSITY MEASUREMENTS FOR EVALUATING THE SPATIAL INFORMATION OF SOUND FIELDS IN REVERBERANT ENCLOSURES

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Over the last twenty years, new subjectively relevant objective room-acoustic indicators for evaluating the acoustical quality of an enclosure have been introduced. While these indicators give new insight into the acoustical "Goodness" of a listener position, in order to *design* halls, *assess* or to *correct* an acoustical defect in an existing enclosure, there is a need to understand to what extent they are influenced by the physical design features of the enclosure. To meet such a need, information about the directional characteristics of sound is required. The spatial distribution of sound energy is usually not considered due to lack of an efficient, accurate and easy to perform measurement method.

The main objectives of the present study are, *first;* to review known and speculative room-acoustic indicators for use in assessing reverberant spaces such as concert halls, opera houses, multi-purpose halls and churches. *Second,* to introduce an easy to perform measurement method for directional sensing in sound fields. *Third,* to develop a simple and inexpensive PC-based instrument primarily for the measurement of sound fields directional characteristics as well as contemporary room-acoustic indicators. *Fourth,* to propose new room-acoustic

indicators which have relevance to directional information.

This study introduces a three-dimensional sound intensity measurement technique for obtaining spatial information of sound fields in an enclosure. The technique has been validated and its accuracy investigated. The method gives results that provide valuable information regarding the directional behaviour of sound in enclosures. Subsequently both the system and the measurement method were applied to known spaces as example applications in order to assess sound quality, to detect the effect of the surrounding interior features of the space, and to assess potential diagnostic capability with respect to interior physical changes.

The study has validated the measurement procedure as well as the importance and potential of visualizing the directional characteristics of sound fields at the listener position employing 3-D transient sound intensity impulses.

In order to utilize the now-available directional information, existing temporal and spatial sound diffusion indices and techniques have been reviewed and additional prospective quantifiers are proposed. The study allows the possibility of developing an appreciation between cause and effect in the matter of interior architectural features design and by providing a better judgment base, removes much of the guess work to achieve cost effective remedial treatments. It also exposes a new dimensional perspective to workers developing objective indicators of subjective response.

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..... To, My Parents, My Wife and My Son

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NOMENCLATURE

AL_{CONS}	= Articulation loss of consonants, %.
$a(t)$	= Weighting function.
a	= Correction factor, ≈ 0% for good listener, varies between 1.5% and 12.5%.
C_{50}, C_{80}	= Clarity, dB.
D_{50}	= Definition or Distinctness coefficient, ratio.
D	= Source-Listener distance, m.
D_C	= Critical distance, metre.
d	= Spacing between the microphone pair, m.
D_θ	= Angular distribution indicator.
EDT	= Early decay time, sec.
EDT_p	= Pressure-based early decay time, sec.
E_{BL}	= Background noise energy.
E_{SL}	= Speech signal energy.
$EDT(f)$	= Early decay time as a function of frequency (f), sec.
F	= Modulation frequency, Hz.
G	= Relative level, or strength, dB.
G_{pu}	= The cross spectrum of the pressure, $p(t)$ and particle velocity, $v(t)$.
G_{pp} & G_{uu}	= The auto-spectrum of $p(t)$ and $v(t)$ respectively.
H	= Hilbert transformation.
$IACC$	= Inter-aural cross-correlation function, ratio.
$<\vec{I}>$	= Average intensity, w/m^2.
$\vec{I}$	= Incoming intensity, w/m^2.
$I(t)$	= Sound energy (Sound intensity), w/m^2.
j	= Imaginary root.
k	= Wave number (i.e. $2\pi f/c$, c = speed of sound, m/s)
LEF	= Lateral Energy Fraction, Ratio.
ms, or *msec*	= millisecond.
m	= metre.
$m(F)$	= Complex modulation transfer function.
N	= The total number of received reflections in the chosen time window.

$P_o(t)$	=	Pressure impulse response measured by omnidirectional microphone.
$P_o(t)_{free\ field}$	=	Pressure impulse response measured in a free field 10 metres from the sound source.
$P_L(t)$	=	Pressure impulse response measured by a unidirectional microphone (e.g. figure of eight microphone).
$P_r(t)$, $P_l(t)$	=	Right and left pressure impulse response measured inside or near the ears of a dummy head.
p_1	=	Sound pressure of channel 1, Pa.
p_2	=	Sound pressure of channel 2, Pa.
$p(t)$	=	Analytic pressure, Pa.
Q	=	The sound source directivity factor, ratio.
RT	=	Reverberation time, sec, (e.g. RT_{20}, RT_{25}, RT_{30}).
RT_I	=	Intensity-based reverberation time, sec.
RT_p	=	Pressure-based reverberation time, sec.
R	=	Running liveness, or reverberant-to-early sound ratio, dB.
RE	=	Real part of complex function.
$RASTI$	=	Rapid speech transmission index, ratio.
R_h	=	Reverberation distance, m.
sec	=	Second.
SPL	=	Sound pressure level, dB.
SWL	=	Sound power level, dB.
SNR_{95}	=	Weighted signal-to-noise ratio, dB.
STI	=	Speech transmission index, ratio.
S/N	=	Signal-to-noise ratio.
$(S/N)_{app}$	=	Apparent signal-to-noise ratio.
ST_{te}	=	The support indicator, dB.
STD	=	Standard Deviation.
t_e	=	denote integral time limit.
T_S	=	Centre time , sec.
T_{1400}	=	RT at 1400 Hz, sec.
T_{st}	=	Statistical reverberation time, sec.
U_{te}	=	Useful-to-detrimental sound ratio, dB.
V	=	Enclosure volume, cubic metre.

$v(t)$	=	Analytic velocity, m/s.
$\phi_{rl}(\tau)$	=	Cross-correlation of left and right signals.
$\phi(\tau)$	=	Cross-correlation function.
τ	=	Time interval, or time shift.
—	=	Average value.
ρ_o	=	Air density, kg/m^3.
*	=	Complex conjugate.
φ	=	Microphone phase mismatch, radians.
Δr	=	Microphone spacing, m.
$\theta_m(n)$	=	The n^{th} measured angle of the intensity vector re. the direct sound.
$\theta_t(n)$	=	The n^{th} theoretical angle of the intensity vector re. the direct sound and is equal to $n.(\theta_T/N)$, θ_T being the total angle of assessment.
Ω	=	Solid angle of interest.
μ_o	=	μ measured in free field with the same microphone.
Δt	=	Duration of time window.
DD	=	Directional diffusion, ratio.
DD_x, DD_y and DD_z	=	The cartesian components of the directional diffusion, ratio.
SD	=	Spatial directional diffusion, ratio.
Δ	=	Temporal diffusion, ratio.
Θ	=	Diffusion index, ratio.
Ψ	=	Pressure correlation coefficient.
γ^2	=	Coherence function.
ψ	=	Deflection angle of the probe minimum sensitivity.

CHAPTER 1

INTRODUCTION

1.1 BACKGROUND

The title of the present research work "***Transient Sound Intensity Measurements for Evaluating the Spatial Information of Sound Fields in Reverberant Enclosures***" is related to architectural or room acoustics. "***Room Acoustics***" is mainly concerned with understanding the behaviour of sound within an enclosed space with a view to obtaining the optimum acoustic effect on the occupants of that space; the space may be small, as in a classroom, or large as found in an opera house and the effects which the space may impose on the subjective properties of sound have to be considered.

The acoustical quality of sound in an enclosure depends not only on the psycho-physiological characteristics of the hearing mechanism but also on its type, for example, speech or music, the properties of secondary sources and the non-human emitters and receivers of sounds i.e. loudspeakers or microphones. A consideration of the physical aspects of the sound processes is not enough to judge the acoustical characteristics of the enclosure, therefore a subjectively relevant objective evaluation of these properties is needed.

This study deals with contemporary and new "*Room-Acoustic Indicators*" that is those parameters needed to understand, quantify and judge the quality or goodness of sound for both music and oratory purposes in enclosures; attention will be confined primarily to those indicators applicable in large reverberant enclosures; such spaces are characterized mainly by large volumes and long reverberation times i.e. 1.2 seconds or more. Theoretically an enclosure is considered large if its shortest dimension is at least one wavelength (preferably two wavelengths) longer than that of the lowest frequency component of interest. Enclosures, and particulary those which are used as large auditoria have linear dimensions which are significantly larger even at low sound frequencies (e.g. 63 Hz has a wave length of 5.44 metres). Sound radiated in such enclosures typically persists and decays over a longer time than in small rooms. The enclosure shape, geometrical characteristics and the interior design features can have a major influence on the received sound and hence upon perceived sound quality. Moreover, the quality of sound in such enclosures may greatly vary from one position to another. These "***Reverberant Enclosures***" are typical of many auditoriums, concert halls, opera houses, theatres, arenas, and churches.

Enclosures may be divided into two basic groups for acoustic purposes; those for direct listening (e.g. lecture halls, theatres, concert halls and churches) and those for sound transmission via an electro-acoustic system or radio signals (e.g. film studios and broadcasting studios).

In enclosures of the first category, both the emission of the natural sound signals and reception processes take place in the same enclosure. In this case the modifying effect imposed upon the source signal by the enclosure characteristics is very important as are the influence of any augmenting sound amplification systems. A listener will usually experience sound directly from the source and sound which has been influenced by the enclosure properties; however the interaction and relative strength of these components will subsequently dictate the subjective impression of quality.

In the second category, the primary sound source is separated from the listener and the connection between them is accomplished by means of an electroacoustic system. Two enclosures are used, the primary which contains the original sound sources and the microphone, and the secondary enclosure in which the loudspeakers are placed and the listeners are seated.

The present study confines attention to the first category i.e. those enclosures employed for direct listening. Both categories of enclosure can also be divided based on their function for example speech and/or music. The variety of enclosures intended for different uses shows that particular acoustical requirements must be satisfied in each case. These requirements can be formulated by detailed studies of the sound propagation in the space together with the physical variations which may influence acoustical conditions and correlating these measures to subjective assessments.

Over the last two decades, many subjectively relevant objective room-acoustic quantities or indicators for evaluating the acoustical quality of an enclosure have been introduced. Some of these indicators relate to reverberance and musical clarity, others to spaciousness or spatial impression, some to speech intelligibility and others to acoustic conditions at performer locations. These indicators are typically "Reverberation Time" (*RT*), "Early Decay Time" (*EDT*), "Definition" (D_{50}), Clarity" (C_{50},C_{80}), "Centre Time" (T_S), "Lateral Energy Fraction" (*LEF*), "Inter-Aural Correlation Coefficient" (*IACC*), "Relative Strength" (*G*), "Speech Transmission Index" (*STI*), "Rapid Speech Transmission Index" (*RASTI*), "Articulation Loss of Consonants" (AL_{cons}), "Support" (*ST*), "Modulation Transfer Function" (*MTF*), and "Useful-to-Detrimental Sound Ratios" (U_{50}, U_{80}, and SNR_{95}).

Most of the newer indicators are calculated from the room impulse response time-dependent sound energy ratios and except for *LEF* and *IACC* do not involve sound directional characteristics. Numbers of them are found to be inter-related and to some extent considerable overlap is exhibited; this implies that knowing a few will allow others to be deduced. They have also been correlated to overall geometric variables such as room width, height and wall angles through empirical models based on statistical analyses. Techniques for precise measurement of these indicators have become common however some indicators are location dependent and more sensitive to the early reflection sequence, so that they are dependent on both the overall geometrical characteristics of the space and architectural details in the vicinity of the measurement position.

Whilst such indicators are valuable, in order to *design* halls, or to *correct* an acoustical defect in an existing enclosure, there is still a need to quantitatively understand the extent to which they are influenced by the various physical design features of the enclosure. In order to develop this understanding, indicator values must be obtained together with directional characteristics information; thus a need exists for suitable comprehensive instrument development. The purpose is to gain knowledge which would allow one to attribute the measured indicators to the interior physical features and to assess diagnostic capabilities with respect to the effect and its cause.

The main objectives of the present study are, *first*; to examine accepted and speculative room-acoustics indicators for use in assessing sound quality in reverberant spaces. *Second*; to develop a comprehensive PC-based instrument for the measurement of such indicators. *Third*; to introduce a new measurement method which gives the directional behaviour of sound at a listener location. *Fourth*; to propose new room-acoustic indicators which have relevance to directional information. Subsequently the developed system will be applied to known spaces i.e. target application in order to obtain the indicators, link the measurements to enclosure physical features and assess diagnostic capability. The study aims to develop an appreciation between cause and effect in the matter of interior architectural features design with particular respect to identifying specific remedial treatments, as well as offering new insight into parameters influencing subjective response.

1.2 OBJECTIVES OF THE RESEARCH WORK

The objectives of the present study can be defined more specifically as follows:

1. ***To examine*** known and speculative objective Room-Acoustic indicators for use in assessing reverberant enclosures for goodness of both musical and speech quality. Speculative indicators will primarily be concerned with direction-based indices hitherto impossible to measure.

2. ***To develop*** a comprehensive PC-based instrument or system to measure objective room-acoustic indicators and to obtain additional valuable information with respect to the directional behaviour of sound at a point in reverberant enclosures.

3. ***To introduce*** a new measurement method which utilizes the potential of transient sound intensity measurements. The purpose is to obtain comprehensive sound directional characteristics in a routine manner with the least hardware and cost. Sound Intensity measurement has the potential for this use and will be examined in detail.

4. ***To apply*** the developed measurement system, and new measures to known enclosures for the purpose of linking the subjectively relevant objective indictors to the interior architectural features of the enclosure

under study and to assess diagnostic capabilities. Additional signal analysis, information presentation and interpretation procedure will also be assessed in order to directly yield an in-situ appreciation between cause and effect with respect to interior architectural features' design and acoustical quality at a point.

5. ***To develop*** simple quantifiers of sound direction distribution and diffuseness from known directional characteristics of the sound field at a listener location.

1.3 IMPORTANCE OF THE RESEARCH TOPIC

An extensive review of research related to contemporary room-acoustic indicators has revealed various measurable aspects having an influence on acoustic perception in an enclosure. Common practice is to quantify and rate the quality of sound in a hall, on spatial averages, compared to others. However reverberant enclosures which might have, on spatial average, the same indicator values can have subjectively different response as to acoustic goodness. Thus space-averaged values are deceiving if used for comparison purpose unless individual position measurements are considered and seat to seat variations thoroughly investigated. Also, most contemporary room-acoustic indictors are based on the capture of room impulse response and subsequent analysis of its time-dependent energy ratios, in consequence they have been developed without regard to sound directional sensing. However subjective criteria must be influenced by the spatial distribution of sound energy which hitherto has not been measured. The spatial distribution of sound energy is usually not considered due to the lack of an efficient, accurate and easy to perform measurement procedure.

Last but not least, as expressed by Gade[30] " *We need to know how the objective parameters are governed by the various physical design variables* ", and " *We need to know how much the design may or should be changed before substantial changes in the objective parameters appear* " In some other cases it has been reported that interior architectural modifications resulted in inaudible sound quality changes or improvements, contrary to all reasonable expectations - Why is this ...?.

Obtaining and visualizing the spatial directional information of sound energy received at a listener location on a time base would clearly be valuable in the following five areas :

1- Such knowledge will help interpret and understand the existing single number criteria in a better and more reliable way particulary when acoustical abnormalities or deficiencies are found. This can lead to effective remedial treatments and possibly improved direction dictated quality indictors.

2- It will allow the measurement of new objective indicators of known laboratory-established values but hitherto impossible to be measured in the field; for example Front/Back energy ratio, which is now known to be a contributor to spatial impression or spaciousness.

3- Many of the newer measures are sensitive to changes in the early reflection sequence, therefore visualizing the sound energy directionally in the early time period will help identify the most dominant influencing interior architectural features at the position under investigation (e.g. influence of proscenium, cantilevered or recessed balconies and facia in the case of a concert hall or vaults, arches and pillars in a church). These influences might be observed and identified from the early part of captured time-based impulse response by looking at delay times. However at best this procedure is difficult, and involves elaborate guessing and generally only applies to seats near the stage or the sound source location where a few multiple strong reflections have occurred with well defined time resolution. Such guessing exercises are impossible for listener locations in the further rear audience seats in a hall or for seats with particular locations such as those under a balcony or in a recessed lounge .....etc.

4- It is known that sound diffusion is required for good acoustical quality[2,52]. Developments of some single number criteria are based on the assumption of diffuse field theory, but measurements in existing halls show that the sound field is not diffuse. Techniques for quantifying sound diffuseness have been proposed a long time ago but due to the lack of an applicable and efficient measurement technique are not considered in many room-acoustics evaluations. Sound diffuseness is *energy*, *time*, *direction* and *frequency* dependent and to be comprehensively assessed all parameters must be obtained. Having obtained sound directional information on an energy directional basis versus time, known and prospective descriptors of sound diffuseness may be tested in conjunction with investigating their relation to existing room-acoustic indicators.

1.4 RESEARCH APPROACH AND METHODOLOGY

The research will consist of both an experimental and theoretical approach. To achieve the defined research objectives, research work may be broken down into several tasks related to each objective as below:

Objective 1 : will require a comprehensive literature survey of room-acoustic indicators, indicators' developments, formulations and rationale. Indicators such as those mentioned on page 1.4 will be examined. Particular attention will be paid to their optimum value range generally accepted in the literature and applicable to the reverberant characteristics of large enclosures. Speculative indicators will primarily be concerned with direction-based or sound diffuseness indices; existing ones will also be examined.

Objective 2 : will involve a literature review related to both available commercial and non-commercial computer-based room acoustics measurement systems and instrumentation in use for room-acoustics evaluations. Their merits and drawbacks will be identified. The system realization will involve software developments for data processing and analysis algorithms, excitation signals generation, data acquisition control and synchronization coding together with hardware interfacing. The primary task is the capture of transient signals (i.e. room impulse responses) and their subsequent processing. Post processing and analysis may involve digital filtering, integration, correlation, intensity evaluation and spectral analysis. Both the system operation and the processing modules have essentially to be validated.

Objective 3 : will involve examination of the measuring techniques for the purpose of obtaining sound field directional information . Particular attention will be given to the direct measurements of sound energy flow for determination of source location and energy path. Vector intensity measurements have been applied to many problems in areas such as noise source localization and diagnosis, diffraction, structural vibration, sound power measurements , transmission loss and damping .... etc, but are only now being utilized in room-acoustics evaluations.

Objective 4 : will involve the selection of suitable measurement sites, for example the **"LOYOLA"** concert hall and chapel, and **"ST. LOUIS DE FRANCE"** church. This task will require the design of the measurements' procedure with a view to link measurements' results, and the influence of interior architectural features.

Objective 5 : will require theoretical formulation of simple quantifiers of sound diffuseness at a point in an enclosure considering different views of defining sound diffusion.

Fig. 1.1 depicts a block diagram for the methodology of the research work and its main components.

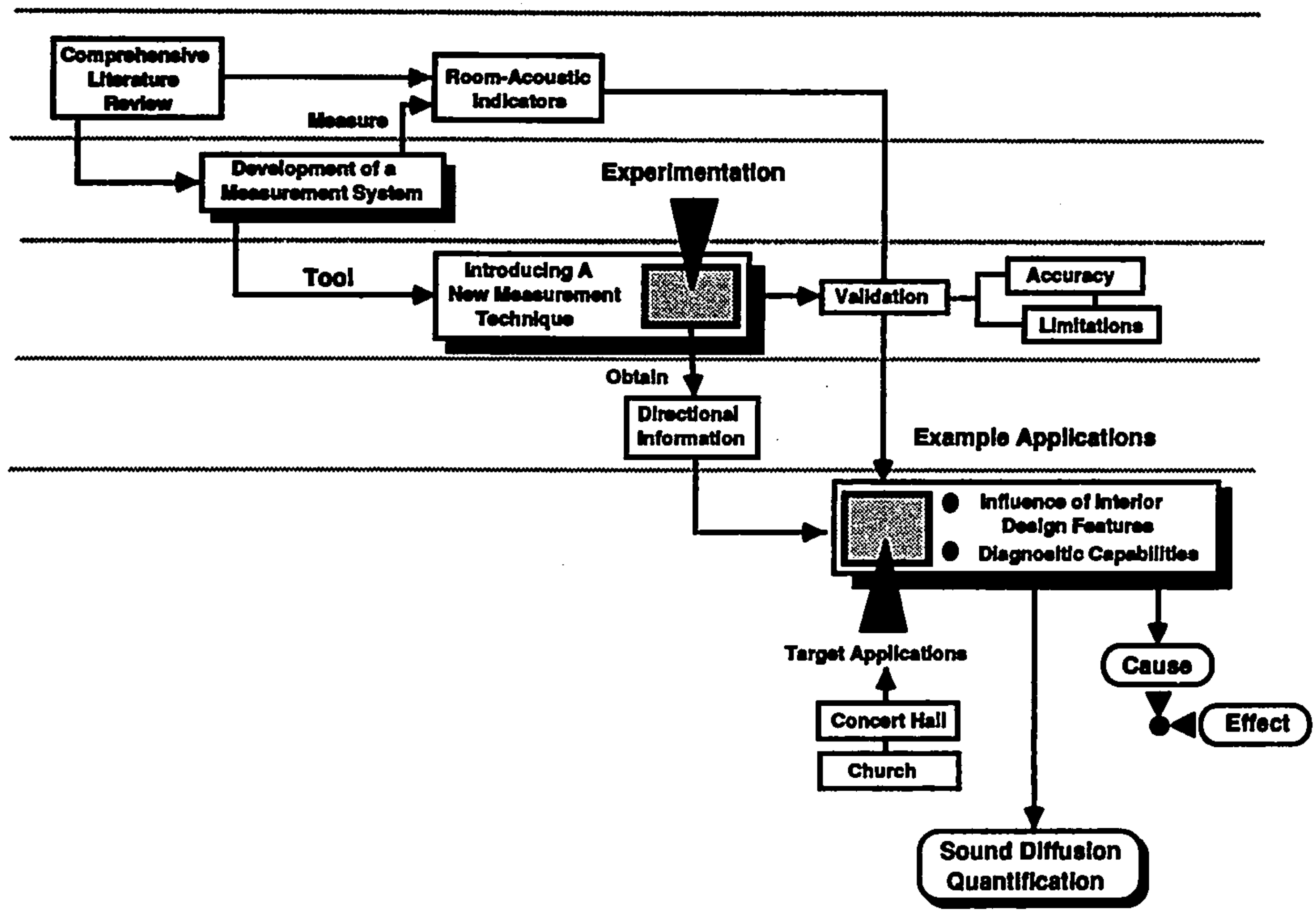


Fig. 1.1 Block diagram of the research work methodology and main components.

CHAPTER 2

LITERATURE REVIEW

2.1 INTRODUCTION

This chapter reports the state of the art with respect to conventional and contemporary objective room-acoustic indicators in use for evaluating sound quality in enclosures. The literature review is classified according to the subjective acoustical attributes widely accepted and considered essential for acoustical goodness; typically they are reverberance, clarity and blend, loudness, tonal colour, spaciousness, auditory source width and speech intelligibility. The corresponding related objective indicators are also grouped under their general subjective attributes. The purpose here is to detail their physical meaning, concepts, basic assumptions, limitations of application and optimal range values, if any.

This chapter also reports the state of the art concerning other important related aspects such as multi-dimensional subjective evaluation studies, impulse response measurement techniques and current computer-based room acoustic measurement systems and instrumentations. The main objective is to develop an understanding of contemporary research work in the area of room-acoustics; this being essential in identifying issues of current research concern, shortcomings and in detailing the present research scope and objectives.

2.2 KNOWN AND SPECULATIVE ROOM-ACOUSTIC INDICATORS

2.2.1 General Background

Wallace C. Sabine's pioneering work[1] developed a fundamental understanding of reverberation and led to the well known reverberation formula which relates an enclosure's primary acoustic and architectural parameters. The reverberation time as a function of frequency was for many years considered the only necessary parameter for characterizing the quality of room-acoustical conditions. Gradually, as a result of measurements in existing halls it became evident that different rooms with almost the same reverberation time were subjectively judged to be acoustically different. Therefore, a need for additional or supplementary room-acoustic criteria was established.

Studies conducted in the fifties and early sixties confirmed that the arrival time of early reflected sounds at a listener's position is critical to the subjective appreciation of quality for both musical clarity and speech intelligibility. Subsequent research work in the late sixties and early seventies, established that not only the temporal distribution of early reflections as perceived by the listener are important but also the spatial aspects or more precisely, the directional distribution of these early reflections is of equal importance.

During the seventies and eighties, the field of objective and subjective room-acoustics, has exhibited intense research activity. These activities resulted in a general consensus on two major aspects; *the first* is the identification of the important aspects in the listener's judgment of room-acoustic quality; *the second* is knowing how these subjective aspects can be objectively measured. The primary developments in architectural acoustics since the beginning of the 19th century are discussed in books by Sabine, Beranek[2], Cremer and Muller[3], Ando[4], and Kuttruff[5]. A significant study of the acoustical characteristics of well known concert halls and auditoria was undertaken by Beranek[2] and published in 1962 in which he provided valuable information concerning the architectural details of many well known halls. Moreover, and most importantly he introduced a number of subjective criteria along with their objective and/or their physical correlates, shown in **Table 2.1**[2,6], derived from his own, musicians' and the critical listening experience of others in auditoria. Ando[4] concluded from subjective tests in an anechoic chamber with samples of music, that there are four orthogonal design-related objectives: (1) The Initial Time-Delay Gap, (2) Reverberation Time, (3) Loudness Level, and (4) a measure of the diffusion of sound at a listener's position termed the "Inter-Aural Correlation Coefficient".

Recent multi-dimensional subjective studies of existing halls especially, concert halls in both Europe and North America have resulted in the introduction of a set of new objective parameters which are considered subjectively more relevant and informative than the classical Sabine reverberation time. These widely accepted indicators together with their symbols, units, definitions, or mathematical expressions, corresponding subjective attributes and suggested tolerance range

TABLE 2.1 : Beranek's criteria for concert hall acoustics. (inferred from references[2,6])

Subjective, Independent Positive Attribute	Objective Expression, or Physical Correlates	Ideal Value Based on a Developed Numerical Rating Scale
Intimacy	Initial Time Delay Gap between receipt of direct and first reflected sound	10-20, msec
Liveness	RT_{mid} = Average of reverberation time, RT, at 500 and 1000 Hz for fully occupied hall	1.9, sec
Warmth	Ratio of low- to mid-frequency reverberation, RT $(RT_{125} + RT_{250})/(2\ RT_{mid})$	1.20 - 1.25
Loudness of Direct Sound	Distance to centre of main floor seats	20 metre (~ 60 ft)
Loudness of Reverberant Sound	Ratio of mid-frequency reverberant time, RT to volume of hall, $(RT_{500} + RT_{1000})/2)/ V * 10^6$, V in ft^3	3.0 sec/ft^3
Diffusion	Long reverberation time + Wall and Ceiling irregularities	"Adequate" (Non - Some - Adequate)
Balance and Blend Sectional Balance in Orchestra	Design of performers end of hall + Response of hall as perceived by performers.	"Good" (Poor - Fair - Good)
Ensemble Ease of on-stage hearing	Performers' ability to hear each other	"Easy" (Difficult - Intermediate - Easy)

Note: Other negative independent attributes such as Echo, Noise and Tonal Distortion were also identified.

values are shown in **Table 2.2**. The indicators' Symbols shown in **Table 2.2** will subsequently be used to refer to the corresponding indicators in the text. For example "Reverberance" *RT* or *EDT*, "Definition" D_{50}, "Clarity" C_{50} or C_{80}, "Spatial Impression" *SI*, and "Loudness" *G*.

Beranek[7] summarized the status of concert hall design with emphasis on those elements of architecture that must be controlled if satisfactory results are to be obtained. Reverberation time, intimacy based on the initial time-delay gap, sound energy that follows the direct sound in the first 50 ms, diffusion, tonal texture together with other factors were demonstrated. Jordan[8] suggested a group of criteria that may be generally sufficient to characterize the acoustics of a particular hall. A certain classification of halls may be achieved when "Early Decay Time", "Lateral Efficiency" and "Clarity" are used as the informative criteria. Numerical values of the proposed criteria in various halls were also reported.

Bradley[9] comprehensively reviewed the development of new types of auditorium acoustic measures for both speech and music. The importance of assessing halls in terms of these new objective parameters was stressed. A list of important subjective acoustical parameters was also provided. Earlier research work[10,12-13] confirmed that a number of these newer indicators are found to be inter-related and to some extent a considerable overlap is exhibited among them. Jullien[12] showed from measurement results in ninety hall configurations, that most of the objective parameters are highly correlated and therefore, they can be calculated

TABLE 2.2 Room-Acoustic Indicators and Their Subjective Attributes.

Room-Acoustic Indicator	Symbol, Units	Definition, or Mathematical Expression	Proposed By	Subjective Attribute	Tolerance Limits
Early Decay Time	EDT, sec	Slope of best fit straight line to sound level decay curve from 0 to -10 dB, extrapolated to -60 dB	Jordan 1975 after Atal *et al.* 1965	Reverberance or Liveness	1.8 → 2.6, sec
Reverberation Time	RT RT_{20}, or RT_{25}, or RT_{30} sec	Slope of best fit straight line to sound level decay curve from -5 to -25 or -30 or -35 dB, extrapolated to -60 dB	Sabine 1923	Reverberance or Liveness	1.4 → 2.8, sec
Running Liveness, or Reverberant-to-Early Sound Ratio	R, dB	$R = 10 . LOG \left(\frac{\int_{50\,ms}^{\infty} P_o^2(t)\,dt}{\int_{0}^{50\,ms} P_o^2(t)\,dt} \right)$ $R = 10 . LOG\,(1/D_{50}-1)$	Beranek & Schultz 1965	Reverberance or Liveness	≈ 4.0 → 6.0 dB

TABLE 2.2 Continued

Room-Acoustic Indicator	Symbol, Units	Definition, or Mathematical Expression	Proposed By	Subjective Attribute	Tolerance Limits
Definition or Distinctness Coefficient	D_{50}, ratio	$D_{50} = \left(\dfrac{\int_0^{50\,ms} P_o^2(t)\,dt}{\int_0^{\infty} P_o^2(t)\,dt} \right)$	Thiele 1953	Speech Intelligibility & Sound Definition	0.4 → 0.6
Clarity	C_{50}, C_{80}, dB	$C_{t_e} = 10.LOG \left(\dfrac{\int_0^{t_e} P_o^2(t)\,dt}{\int_{t_e}^{\infty} P_o^2(t)\,dt} \right)$	Reichardt 1975	Music Clarity & Blend	Ideally = 0 dB or ± 2 dB, -3.0 dB is also tolerable
Centre Time	T_S, sec	$T_S = \left(\dfrac{\int_0^{\infty} t.P_o^2(t)\,dt}{\int_0^{\infty} P_o^2(t)\,dt} \right)$	Cremer 1982 after Kürer 1969	Music Clarity & Blend	< 140-144 ms

TABLE 2.2 Continued

Room-Acoustic Indicator	Symbol, Units	Definition, or Mathematical Expression	Proposed By	Subjective Attribute	Tolerance Limits
Relative Level or Strength	G, dB	$G = 10.LOG\left(\frac{\int_0^{\infty} P_o^2(t)\,dt}{\int_0^{\Delta t} (P_o^2(t)_{freefield}\,dt)_{10m}}\right)$ $or,\ G = SPL - SWL$	Lehmann 1976	Loudness	+ 2 to + 6 dB
Lateral Energy Fraction	LEF, ratio	$LEF = \left(\frac{\int_{t_s}^{80ms} [\,P_L^2(t)\,dt\,]_{Lateral}}{\int_0^{80ms} [\,P_o^2(t)\,dt\,]_{Total}}\right)$ $where,\ t_s = 0\ or\ 5\ ms$	Jordan 1980 ; Barron 1981	Spaciousness or Spatial Impression or Envelopment	0.20 - 0.30
Inter-Aural Cross-Correlation Coefficient	IACC, ratio	$\Phi_{rl}(\tau) = \frac{\int_0^{t_e} P_r(t)\,.\,P_l(t+\tau)\,dt}{\left(\int_0^{t_e} P_r^2(t)\,dt \int_0^{t_e} P_l^2(t)\,dt\right)^{1/2}}$ $IACC = max.\ of\ \lvert \Phi_{rl}(\tau) \rvert,\quad \lvert\tau\rvert \le 1\ ms$ $where,\quad t_e = 80\ or\ 100\ ms$	Gottlob 1973 after Keet 1969; & Damaske 1967/68	Spaciousness or Apparent Source Width	< 0.35 - 0.4

TABLE 2.2 Continued

Room-Acoustic Indicator	Symbol, Units	Definition, or Mathematical Expression	Proposed By	Subjective Attribute	Tolerance Limits
Weighted Signal-to-Noise Ratio	SNR_{95}, dB	$SNR_{95} = \dfrac{\int_0^{95\,ms} a(t)\,.\,P_o^2(t)\,dt}{\int_{95\,ms}^{\infty} P_o^2(t)}$ where, $a(t) = \begin{cases} 1, & 0 \le t \le 35\,ms \\ -\frac{1}{60}(t-95), & 35 < t \le 95\,ms \\ 0, & t > 95\,ms \end{cases}$	Lochner & Burger 1964	Speech Intelligibility	-4 → +4 dB
Articulation Loss of Consonants	AL_{cons}, %	$AL_{cons} = \left(\frac{200 D^2 T^2}{V} + a\right)$ % for $D \le D_c$ $AL_{cons} = (9T + a)$ % for $D > D_c$ where, $D_c = 0.2\sqrt{V/T}$, m T = RT at 1400 Hz , sec D = Source-listener distance , m a = correction factor , % or, $AL_{cons} = 170.5405\,.\,e^{(-5.419\,.\,RASTI)}$	Peutz 1971	Speech Intelligibility	< 10% = Very Good > 10% < 15% = Good > 15% = Difficult

TABLE 2.2 Continued

Room-Acoustic Indicator	Symbol, Units	Definition, or Mathematical Expression	Proposed By	Subjective Attribute	Tolerance Limits
Speech Transmission Index & Rapid Speech Transmission Index	STI, ratio RASTI, ratio	Based on: $m(F) = \left(\dfrac{\int_0^\infty P_o^2(t) \cdot e^{-j(2\pi F)} dt}{\int_0^\infty P_o^2(t)} \right)$ *where, F = modulation frequency, Hz* For diffuse sound conditions: $m(F) = \dfrac{1}{\sqrt{1+(2\pi F.(T/13.8)^2}} \cdot \dfrac{1}{1+10^{(-S/N/10}}$ $(S/N)_{app} = 10.LOG\left(\dfrac{m(F)}{1-m(F)}\right)$, dB $(-15 \leq (S/N)_{app} \leq +15)$ $RASTI = [(\overline{S/N})_{app} + 5] / 30$	Steeneken & Houtgast 1980	Speech Intelligibility	0 → 1, A Rating Scale is also established
Useful-to-Detrimental Sound Ratio	U_{te}, dB	$U_{t_e} = \dfrac{C_{t_e}}{[1+(C_{t_e}+1)*(E_{RL}/E_{SL})]}$ *where,* $t_e = 50$, or 80 ms	Bradley 1986	Speech Intelligibility	-3 → +3 dB

TABLE 2.2 Continued

Room-Acoustic Indicator	Symbol, Units	Definition, or Mathematical Expression	Proposed By	Subjective Attribute	Tolerance Limits
Tonal Color, Timbre	EDT(f), (1) sec/octave, (2),(3) ratio	(1) $EDT(f) = (1/3).(EDT_{2000Hz} - EDT_{250Hz})$ (2) $EDT(f) = \frac{EDT_{250Hz} + EDT_{500Hz}}{EDT_{1000Hz} + EDT_{2000Hz}}$ (3) $Bass\ Ratio = \frac{(EDT_{125Hz} + EDT_{250Hz})}{(EDT_{500Hz} + EDT_{1000Hz})}$ $Treble\ Ratio = \frac{(EDT_{2000Hz} + EDT_{4000Hz})}{(EDT_{500Hz} + EDT_{1000Hz})}$	(1) Jordan ; (2) Gade ; (3) Bradley	Tone Color, & Balance Bass & Treble Balance	NOT firmly established
Support	ST_{100}, ST_{200}, dB	$ST_{t_e} = 10.LOG\left(\frac{\int_{20ms}^{t_e} p^2(t)\,dt}{\int_{0}^{10ms} p^2(t)\,dt}\right)_{1m-source}$ where, t_e = 100 or 200 ms	Gade 1989	Impression of Support & Ease of Ensemble	-12 dB±1 dB -8 → -12 dB

from a knowledge of just a very few independent parameters. However no correlation could be found between "Lateral Energy Fraction", *LEF* and "Inter-Aural Correlation Coefficient", *IACC*. Recently, Tachibana and Yamasaki[13] investigated the relationships among six objective room acoustic indices by statistical analysis. The highest correlation was found between C_{80} and T_S, D_{50} and C_{80}, therefore, it was suggested to consolidate them. T_S seemed the best to the authors as representing transient room characteristics. Although *RT* and *EDT* are similar quantities, they were found to have low correlation.

Barron and Lee[14] conducted extensive objective acoustic measurements in fifteen unoccupied concert halls and two multi-purpose halls used for music. Results were compared with traditional predicted values and a revised theory of sound decay in concert halls was proposed and validated.

Several authors[10-13,15-19] had reported measurements of room-acoustic indicators in well known concert halls. Schroeder *et al.*[15] evaluated the acoustics of the "*Philharmonic* " concert hall in New York by a new method utilizing digital techniques. Several quantities were measured and analyzed to yield possible acoustic deficiencies. A large number of measurements[10] had been undertaken in a variety of halls to improve familiarity with new room-acoustic parameters and to provide both more and a broader range of data than previously known. Details of a procedure for efficiently and accurately calculating various auditoria acoustic measures from pistol shots has also been described[11], and confirmed that various measures are strongly inter-correlated.

The acoustic characteristics of concert halls for different styles of music have been investigated by Giménez and Marin[16]. The subjective preferences were classified according to three criteria; energy, time and space. An assessment approach was applied to three different halls to yield optimal values, then an assessment scale for concert halls that offers an evaluation for different styles of music was presented. Nagata[17] summarized the listening impressions of five concert halls in *Tokyo* and compared them with a design criteria; architectural and acoustical characteristics were also considered. He concluded with the essential aspects to be considered in room acoustic design such as the temporal and spatial control of early sound reflections in addition to the control of reverberant sound.

Marshall[18] discussed experience with the Early/Reverberant sound ratios. Measured data were examined with respect to the variables: reverberation time, frequency, source and receiver locations. He concluded that energy ratios have proved useful in quantifying and predicting intelligibility and music acoustics.

In an attempt to better define the acoustical conditions required of a good concert hall, Bradley[19] comprehensively analyzed measured values of the new objective room-acoustics quantities in three well known classical concert halls. The variations of the quantities with both source and receiver position were elaborately examined. The objective characteristics of the same three halls were also compared to those of a multi-purpose hall[20]. The usefulness of employing the newer acoustic measures for objective comparisons of concert halls was also demonstrated[21].

Beranek[22] more recently emphasized five acoustical factors that contribute in a primary way to the acoustical quality of a concert hall at a listener's position: These are (a) The initial-time -delay gap; (b) Reverberation time (for occupied halls) at mid-frequencies; (c) Loudness of the sound; (d) The sound field diffusion; and (e) the ratio of the reverberation times at low-frequencies to those at mid-frequencies. The author's judgment as to the relative importance of the five principal factors were also discussed and optimum values suggested.

Barron[23] conducted an acoustic survey of British auditoria involving both objective acoustic measurements and subjective listening tests and concluded that both offer much improved understanding of the acoustic behaviour of individual halls. The acoustic problems associated with large concert halls were identified and the need for sufficient early reflections was stressed.

Haan and Fricke[24] reviewed the usefulness and the guidelines available to designers with the objective of detailing an approach to acoustic design in the early design stage. This approach was based on a statistical analysis of the form of existing auditoria. Re-evaluation of exiting rules of thumb with respect to room volume, layout of audience seating, and shape of auditorium was also attempted.

Pelorson *et al.*[25] attempted to evaluate the accuracy expected from acoustic parameter measurements and the influence of each element of a measuring system was evaluated. C_{80} and *LEF* were found to depend strongly on the sound source (i.e. the loudspeaker system). The effect of small spatial variations on the acoustic parameters was evaluated and compared to reproducibility and within hall variations. The critical problem of hall to hall comparisons was also considered. The study showed that *EDT*, C_{80}, and *G* are suitable for comparing one hall to another and *RT* and *LEF* generally are unable to describe spatial variations comparisons between halls but they were found to be useful for comparing positions within a given hall.

2.2.2 Reverberance

Reverberance can be expressed by either "Reverberation Time", *RT* or "Early Decay Time", *EDT*. *RT* is defined as the time it takes for sound to decay by 60 dB after the source is stopped. It is usually determined by extrapolating the slope of the first part of a sound pressure level decay curve or the Schroeder[26] integrated impulse decay curve between -5 and -35 dB. Reverberance appears to be responsible for the sensation of being in a room as well as providing a sensation of distance from the sound source. The *RT* as a function of frequency (typically in octave bands) was for a long time considered the only crucial parameter for describing room-acoustic conditions or for specifying design criterion. Later, analysis of measurements results conducted in well known halls, showed that different rooms with apparently similar *RT* 's were subjectively judged to be different[10]. Today, reverberation time is considered a rather crude objective measure of acoustical quality in enclosures, especially in large reverberant spaces. However, it is still a useful indicator of the mean properties of the room, besides it is easy to estimate from a knowledge of the room materials' absorption characteristics and geometry; in addition being a single value, it is considered appropriate for general acoustical characterization of an enclosure.

There are already several developed formulae to express optimum values of reverberation time[27]. Experiments had shown that optimum *RT* values depend on the use for which the enclosure is intended, its geometry, the sound frequencies, the nature of the sound source and the type of musical performance. Recommended values of *RT* 's are provided in references[3]. It is accepted[22] that for optimum listening conditions (*RT*) values must be in the range of about 1.6 to 2.0 seconds at mid frequencies (i.e. average of *RT* at 500 Hz and 1 kHz one octave bands). Optimal values[16,28] for *RT* also depend on the style of the music played e.g. 1.7-1.9 s for classical music, 1.5-1.7 s for Baroque and 2.2-2.4 s for Romantic music. For *EDT*, the range of optimal values is similar to *RT* but decreases by about 10%. However as the music tastes of individuals differ, optimum values of reverberation time is typically defined as those values acceptable to a large group of listeners. To give a sense of the preferred ranges of *RT* 's at mid-frequencies for a variety of activities, a bar graph[28] has been developed (Fig. 2.1).

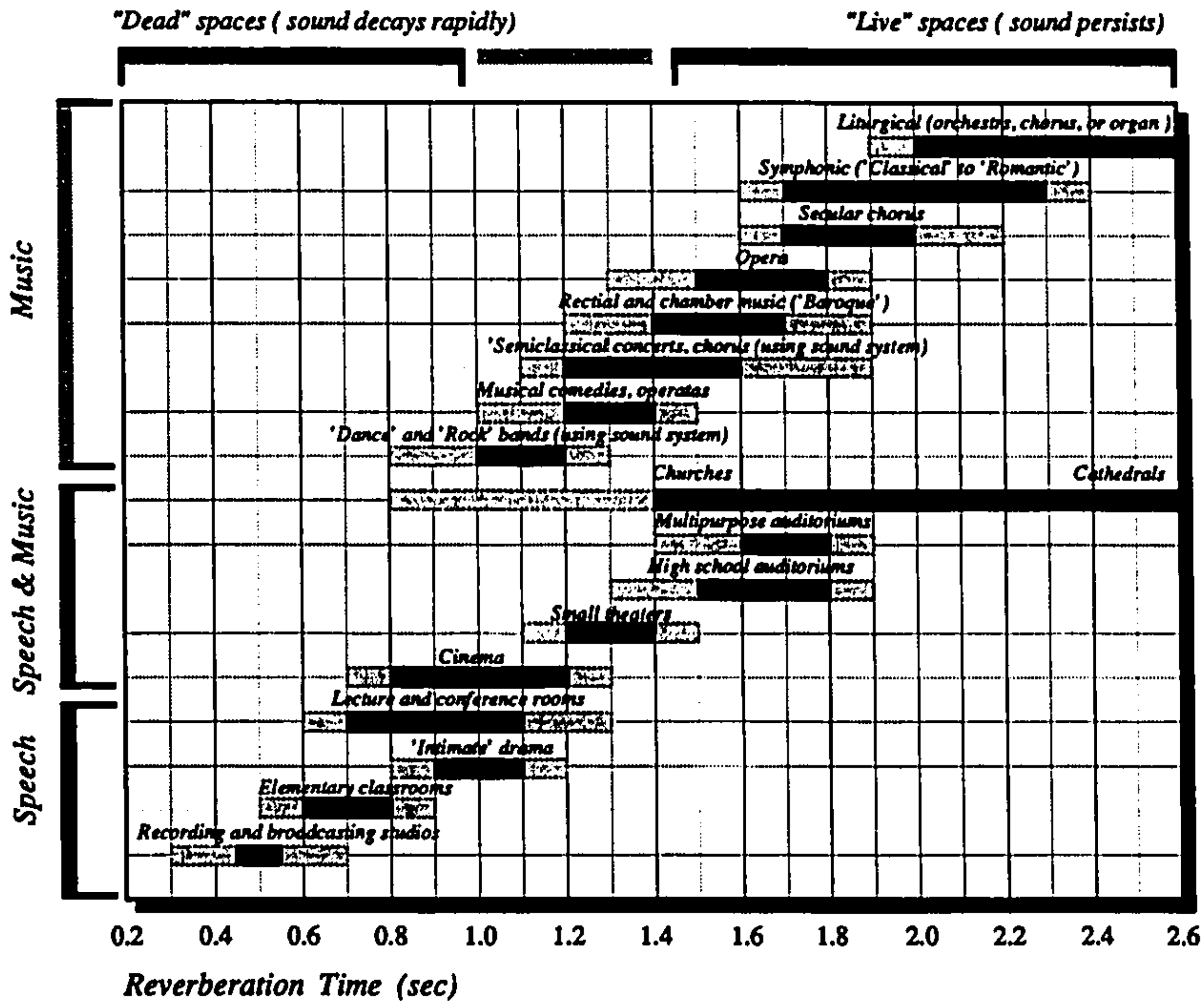


Fig. 2.1 The preferred ranges of Reverberation Time, RT at mid-frequency (i.e. average of RT's at 500 and 1000 Hz) for a variety of activities [Adapted from reference [*]].

Jordan[29] summarized the conclusions of Gottlob's analysis of pulse measurements recorded in twenty five halls to investigate correlations between the physical criteria and subjective tests. From the results of these measurements reverberation time was found to become important only if the value is greater than 2 seconds, whilst at lower values the important physical indicator becomes the early-to-late energy ratio.

Recent studies conducted by Gade[30,31] proved that the classical reverberation time in addition to room volume, is the main factor governing the behaviour aspects of most of the newer measures such as sound level, reverberance and clarity and from which other indicators can be predicted to serve in the design process stage. Proposed empirical models for their predictions based on statistical analysis were presented.

In addition to the traditional graphic level recording method, several techniques have been proposed for evaluating reverberation time, the most practical and commonly used is the "Integrated Impulse Method" introduced by Schroeder[26]: he proved that the decay curve resulting from a simple integral over the squared impulse response is identical to the ensemble average of the squared noise decay curves obtained with the traditional interrupted noise excitation method. The smooth decay curves resulting from the new method improve the accuracy of *RT* evaluation. The method yields highly reproducible decay curves and facilitates the detection of non-exponential decays. While analogue techniques have been mostly used in the implementation of this method, Bodlund[32] used a digital measurement procedure to realize the integrated impulse response by utilizing a sound waveform recorder and a mini-computer. Both analogue and digital techniques were analyzed and their advantages together with associated errors discussed.

In a study by Chu[33], particular attention was given to the question of repeatability and the influence of signal-to-noise ratio on the application of the Schroeder's integrated impulse method. Aoshima[34] devised a new method by which the *RT* of room responses excluding effects of filters can be obtained by using Fourier Transforms but it has the disadvantage of lengthy computation. The application of a correlation technique to reverberation time measurements was considered by Yanagisawa and Uemura[35] to maintain the signal-to-noise ratio in a high ambient noise field. A simple procedure for correcting reverberation times of large reverberant enclosures as a function of humidity and temperature was suggested by Benedetto and Spagnolo[36].

Earlier (1965) Atal *et al.*[37] found by subjective tests that the early decay of signals in concert halls is more relevant for subjective judgment of reverberance, they therefore recommended the use of an "Initial Reverberation Time" as a room-acoustic criterion instead of the traditional *RT*. It is defined as the decay slope from a straight line fitted to the decay curve observed during the first 160 ms or alternatively the first 15 dB of the actual decay. Later, with the development of the Schroeder's integrated impulse method this recommendation became practically useful to include in acoustical evaluations. Based on the observation that the total decay process over 60 dB is rarely heard by the listener, Jordan[8] was the first to propose the parameter "Early Decay Time", *EDT* as a criterion to judge the degree of subjectively perceived impression of sound reverberance in rooms. *EDT* evaluation is similar to *RT* but the sound decay range permissible for linear regression is restricted to the first -10 dB.

Xiang[38] has recently proposed an alternative method using a nonlinear iterative regression approach for evaluating *RT* from Schroeder's decay curve rather than the linear regression model commonly used. The purpose is to improve the uncertainty of the evaluated *RT*'s due to the finite end selection of the impulse response. However the method increases computational time and further investigations for its applicability remain.

2.2.3 Music definition, Clarity and Blend

The subjective balance between clarity, definition and reverberation can be judged by early-to-total sound ratio indicators such as Definition, D_{50} proposed by Thiele [references[3,10]] and defined as the ratio of the early sound energy in the first 50 ms after the arrival of the direct sound to the total sound energy. Optimum values for the mid-frequency range are reported to be in the 0.40-0.60 range[16], however a value of 0.34 has also been suggested as optimal[3].

Clarity, C_{80} proposed by Reichardt [reported in references[3,10]] and defined as the ratio of the early arriving sound energy (in the first 80 ms after the direct sound) to the late sound energy arriving after 80 ms. It has been suggested[8] that optimum clarity corresponds to mid-frequencies C_{80} values between 0 and ± 2 dB. However, a values of -3 dB is still tolerable[5] based on the style of the music played (i.e. Classical, Baroque or Romantic music). Fischetti and Jouhaneau[39] carried out subjective experiments concerning relationships between subjective clarity and objective criterion C_{80} and the effect of early reverberant energy was investigated. They found that subjective clarity could be increased by controlling the temporal interspike gaps in the early reflection-sequences. However further investigation is required to give more details on the effect of reflections' gaps positions and how this could be practically calculated.

Centre Time, T_S or centre of gravity time proposed by Cremer[3,10] and defined as the ratio of total arriving energy weighted by the time arrival intervals to the total arriving energy has been proposed as an alternative to early-to-late sound ratios. It is suggested[3] that its value should not exceed 140-144 ms. For clarity and blend balance expressed by the foregoing indicators, a low value indicates poor definition referred to subjectively as "***muddy sound***" while a high value indicates that it is possible to discriminate the sound details but the sound may also be subjectively "***very dry***" as if it is produced in a room with too much absorption[107].

2.2.4 Sound Loudness

The overall loudness expressed by the "Relative Level" or "Strength", G is defined as the sound field level relative to that for the same source at a distance of 10 m in a free field in the absence of any reflections (i.e. anechoic or free field conditions). It is related to the enclosure's ability to amplify sound from the stage[3,14,19]. It is also used to compare results from different positions as this shows the sound distribution in the hall under investigation. From measurement-results in three of the best classical concert halls[19], mid-frequency G values of + 2 dB to + 4 dB. However higher values for smaller halls can be considered as indicating ideal levels of sound loudness.

2.2.5 Tonal colour, "Timbre"

The influence of the room on the tonal colour of the instrument and on the balance in level in different registers is known as "Timbre". Tonal colour can also be defined as the ability to judge the musical sound according to its frequency balance or dominance by its bass or treble content. Jordan[8] tentatively proposed after Lehmann and Wilkens that tonal balance can be objectively quantified simply by the slope of the frequency dependence of the indicator $EDT(f)$. Barron[76] also found that the subjective judgement of "Timbre" is strongly related to the variation of EDT values with frequency. Gade[65] suggested a crude description of the tonal characteristics of a hall by low (250-500 Hz)/high (1-2 kHz) frequency ratios of

EDT. Bradley[19] proposed that more specific Timbre-related parameters can be calculated by grouping measurements as low (125-250 Hz), medium (500-1000 Hz), or high-frequency (2-4 kHz) values. "Timbre" quantification definitions vary and accordingly, no tolerance limits have been firmly established.

2.2.6 Spaciousness

The possible importance of early lateral reflections in concert halls were first introduced by Marshall[40]. Based on previous work on auditory masking, he investigated the reflection-sequences in two idealized hall shapes. He concluded that for an auditorium to be acoustically good it should have "Spatial Responsiveness" which he attributed to the presence of many temporally and directionally distributed early lateral reflections. Since his original suggestion, there has been a growing body of opinion that spatial impression is a critical attribute in appreciating the acoustical quality of sound in large enclosures.

Earlier work of Schultz and Watters[41], and Sessler and West[42] stressed the importance of low frequency sound energy in the late reflections for desirable subjective sensation of envelopment by sound. Lack of low frequency in the early sound was not crucial since it is attenuated at grazing incidence by the seated audience, and as long as a judgment of adequate bass is provided by sufficient low frequency content in the late reverberant field. Lawrence[43] proposed a parameter called "Volume"; a similar concept to spatial responsiveness. This parameter is related to the perception of reflected sounds coming from many different directions. Barron[44] reported experiments with simulated reflections in an anechoic chamber; the subjective effects of a single and two side reflections were investigated. The variation in the degree of spatial impression for variations in different reflection parameters were also investigated in detail.

Barron and Marshall[45] carried out an extensive series of experiments using a simulation system to investigate the determinants of the subjective effect created by lateral reflections. In these tests, the effect of reflections delay, direction, level and spectrum were investigated. The results led to the introduction of a new objective acoustic measure; the "Early Lateral Energy Fraction", *LEF* which was found to be linearly related to an associated subjective perceived sensation termed "Spatial Impression", *SI*. *LEF* is defined as the ratio of lateral to none lateral energy. It was found that as the lateral reflections level is increased the apparent source width appears to broaden and the music gains body and fullness. This means that the impression of being in a three dimensional space is enhanced (though without any real knowledge of the size of that space) or in other words the listener experiences the sensation of being enveloped by sound. The study had other conclusions which made clear the importance of early reflections; this however does not agree with later results[147] suggesting that envelopment is only achieved when late arriving lateral reflections are present. Cremer[46] described and discussed some of the means by which lateral reflections of sound have been provided with respect to the seating arrangement in newer large concert halls.

Today, the importance of lateral sound particulary of early reflections is almost universally recognized however, its method of measurement is not yet standardized. Pirn[47] showed that total energy measured with an omnidirectional microphone is not compatible with a summation of the directionally measured energy as a result of instrument characteristics. Therefore, he suggested a new procedure for the measurement of the lateral energy fractions; measured directional sound is simplified to frontal and lateral within a specified frequency range. In a study reported by Williamson[48] a number of objective indices were measured from the room impulse response and an optimum lateral energy condition was established from subjective evaluation.

Olive and Toole[49] studied the influence of reflected sounds on the timbre, and the spatial characteristics of live and reproduced sounds were also discussed. The effects of reflected sounds in typical rooms as they would occur in a stereophonic reproduction were also examined and classified. Kliener[50] presented a new method to measure the lateral energy fraction of the early reflection pattern of a room. The method is based on the dual-microphone technique which is used in intensity measurements. The concept is to allow for a correct evaluation of *LEF* via *cosine* response measured by microphones as originally proposed by Barron and Marshall[45] as well as the more commonly used squared *cosine* response measured by figure of eight microphones.

As a result of an extensive subjective evaluation of some well known European concert halls, Schroeder *et al.*[51] found that the binaural dissimilarity of the two signals recorded from a specially designed dummy head was highly correlated with the subjective preference data. This strong correlation with preference was more than with any other objective parameter investigated including *RT*. Schroeder[52] suggested "Binaural Similarity" i.e. inter-aural coherence, as a criterion of acoustic quality; this is usually termed "Inter-Aural Correlation Coefficient", *IACC*. It is defined as the peak value of the correlation function of the first 80 ms of the impulse response within an inter-aural delay range of 1 ms. The value of binaural similarity should ideally approach zero[6] i.e. received signals are fully dissimilar. However this condition seems unlikely to occur and thus low values ($<= 0.4$) are considered more realistic. It is related to the directional distribution of the reflected sounds. Greater binaural dissimilarity leads to a more stereophonic listening experience as opposed to monophonic. He also showed that the highest correlation between subjective preference and geometrical parameters of the hall was with the width of the hall. He then, suggested that the problem of coherent ceiling reflections may be overcome if specially modelled diffusing surfaces are used. Two ceiling designs to maximize early lateral reflections were proposed. Effects of diffusing areas in the ceiling were discussed by Marshall *et al.*[53]. West[54] also demonstrated the possible subjective significance of the height to width ratios of concert halls.

Another subjective acoustic parameter proposed by Blauert and Lindemann[55] termed "Auditory Spaciousness" has much in common with Schroeder's binaural dissimilarity and spatial impression concepts. It is a perceptual attribute predominantly caused by early lateral reflections with all contributing spectral components. Mainly, it is said to describe the ability of listeners to develop an intuitive auditory impression of a hall's type and dimension.

In a pilot study of simulated spaciousness, Bradley *et al.*[56] reported the results of initial subjective evaluations of simulated sound fields with varied level of early lateral reflections. Judgments of the "Apparent Source Width", *ASW* were found to be well correlated with both objective measures of spaciousness i.e. *LEF* and *IACC*(early) at mid-frequency. The study confirmed results of earlier research work that louder sounds were judged to be more spacious and lower frequencies also led to greater *ASW*. However, Soulodre *et al.*[57] showed that neither *IACC* nor *LEF* was able to correctly predict *ASW* for all sound fields and that in general, when one measure successfully predicts *ASW* the other fails, therefore they suggested that an ultimate measure of spaciousness should incorporate aspects of both *IACC* and *LEF*.

Morimoto[58] investigated the relation between *ASW* produced by different structures of reflections and the degree of *IACC*. He found that *ASW* perceived in different sound fields with the same *IACC* are equal to each other, regardless of the number and arriving direction of sound reflections but *ASW* perceived in a sound field where a direct sound arrives from 30°, 60° or 90° in azimuth is narrower than *ASW* in a sound field where the direct sound arrives from the front, even if their degree of *IACC* are equal to each other. These conclusions were also confirmed even when the subjective test method was changed[59].

The effect of equipment and procedure on IACC measurement had been studied by Siebein *et al.*[60], variabilities were found and attributed to the characteristics of the sound source and/or undersampling of the data acquisition process. A need for more detailed comparative evaluations of the signal processing methods and refining the method of field recording for binaural measurements were identified.

To compare *IACC* and *LEF* measurements in halls Bradley[61] studied measurement results from 85 locations in ten different halls and indicated varying relationships with frequency. *IACC* and *LEF* individual measurements and hall average values were strongly related at middle octave band frequencies and therefore can be used to reasonably rank order measured halls.

Recently, Morimoto[62] showed as a result of psychological experiments that envelopment sensation grows as the energy of the reflections coming from the back of the listeners increase even if the degree of *IACC* of late reflections are equal, he hence suggested a new parameter i.e. Front/Back sound energy ratio, *F/B* and emphasized the need to measure such a ratio but no means of how objectively one can measure it in sound fields was indicated.

2.3 MUSICIANS' CRITERION, "PLATFORM-ACOUSTICS"

It is well known that the acoustical requirements for the performers on the orchestra stage are quite different from those of the listeners in the audience area. Beranek[2] proposed that the criterion "Balance and Blend" for good acoustical quality in concert halls is related to the design of performers' platform based on the response they receive back from the audience area. However, Jordan[63] was the first to consider the performers' or musicians' acoustical requirements on the orchestra stage. He proposed the hypothetical concept of "Inversion Index", defined as the ratio between the average value of a short criterion such as *EDT* or C_{80} measured on the stage and the average value measured in the audience area. It is based on the judgement that the blend of different musical groups and the creation of musical impression should develop faster in the stage area than in the audience area. The simple idea behind this concept is that any acoustical signal should attain its final level more rapidly on the stage than in the audience area. He also suggested that for the musical impression, dimensions of the orchestra stage, together with the diffusion properties of its boundaries should be more closely studied using the speculative parameter "Early Energy Balance", *EEB* as a new acoustical criterion for the stage of a concert hall. *EEB* was defined as the amount of diffused energy received between 0 and 35 ms compared with the energy of the direct sound within the first 5 ms time interval. A tolerance range from 2.6 dB to 11.7 dB with an average of 6.2 dB was suggested.

In detailed studies Marshall *et al.*[53], investigated ease of ensemble dependency and reverberation role in acoustical conditions for musicians. Several useful hypotheses were proposed for ease of ensemble and hence some design guidelines for stages intended for orchestral music were proposed. Other studies, most promising by Gade[64,65] and Naylor[78], all tried to establish relationships between the subjective judgement of musicians on the stage and measurable objective acoustic parameters. Gade[64] found for ensembles that efficient transmission of the direct sound and early reflections plays an important role in a musician's ability to hear his colleague. Based on this concept, two acoustical parameters for the objective measurement of musicians' impression of support and ensemble conditions on orchestra platforms were proposed. The parameters were derived from results of subjective measurements in the laboratory. The first parameter introduced to quantify the musician's ability to hear his colleague, Gade referred to as the "Early Ensemble Level", *EEL* defined as the ratio between the energy of the first 80 ms of the impulse response measured at the receiver and the energy of the first 10 ms of the impulse response measured at 1 m from the source. In a following publication, the *EEL* parameter was discounted by Gade[65]. He confirmed that the proposed parameter "Support", *ST* correlate well with the subject averaged judgments; he attributed adequate support to the existence of sufficient early reflections on the stage. For the objective measurement of musicians' impression on the stage, the indictor "Support" ST_{100} and ST_{200} were proposed, and defined[65] as the ratio of energy found in the first 100 ms, ST_{100} or 200 ms, ST_{200} (i.e. early reflections) to that of the direct sound within the first 10 ms. "Support" is the property which makes the musicians feel that they can hear themselves and that it is not necessary to force the instrument to develop the desired tone. The ratio is measured 1.0 m from the source on the stage; this is comparable to the distance from the performers' ears to their own instrument. High values correspond to strong feeling of support. Based on experimental work and measurements in existing halls, optimal ranges for these parameters were suggested. Optimal ST_{100} values range for ensemble in symphony orchestra are suggested to be -12±1 dB, but for smaller groups higher values may be allowed. ST_{200} is suggested to be between -8 and -12 dB. Gade[66] revised his "Support" measures and now uses ST_{early}, ST_{late}, and ST_{total}.

O' Keefe[67] summarized the conclusions of Naylor[78] work; Naylor found that the "Modulation Transfer Function", *MTF* used by Houtgast and Steeneken[68] to model speech intelligibility was a good model for what he called "Hearing Of Others", *HOO*. A linear relation between them exists, but when reverberation time is added the relation becomes non-linear. To better understand stage or platform acoustics employing the indicators proposed by Gade and Naylor, modern acoustic measurements were undertaken by O' Keefe and Bracken[69] on some Canadian stages.

2.4 MULTI-DIMENSIONAL SUBJECTIVE EVALUATION STUDIES

Although objective evaluations are essentially needed for judging the acoustical conditions in auditoria, objective room-acoustic indicators are of not much value if at all, unless they relate to subjective evaluations of the same acoustical conditions. Therefore, subjective tests to first link and second to validate objective indicators are essential. However, it is difficult to carry out such investigations without an ideal controlled experiment.

Yamaguchi[70] conducted a study to examine a multivariate procedure for subjective assessment of sound field qualities among seats in a music hall in place of "*Word List*" used in conventional techniques. He proposed a method to estimate sound field characteristics of music halls in physical terms. Latham[71] comprehensively reviewed the theoretical basis of subjective measurement methods in auditorium acoustics. In his survey two schools of thought had been identified and represented mainly by two groups of studies; the first school favoured preference comparisons (i.e. *Gottingen Group*[51]); the second used a semantic differential ratings method (i.e. *Berlin Group*[3]). The advantages and disadvantages of both methods were discussed and a comprehensive list of related references was provided. Bradley[9] also summarized some details of these studies, their major difference in procedure and approach were demonstrated. While criticism of the details of these studies may vary, they have revealed that a relatively small number of subjective dimensions explain almost all of the variance in subjective evaluation of concert hall sound and therefore a limited number of objective indicators are required.

Ando and Imamura[72] introduced the design concept of "Overall Preference" for optimal acoustic design of concert halls. Preference tests for simulated sound fields with various combinations of early discrete reflections and subsequent reverberation were conducted and results were compared with objective parameters. In a notable study by Ando[73], he represented a method of calculating the subjective preference of sound fields in concert halls before construction. Examples of calculating the preference values by use of the plan and the cross-section of a concert hall were described.

Plenge *et al.*[74] conducted a series of experiments in a number of concert halls to evaluate their acoustical quality from a comparison of subjective assessments with a set of specified parameters describing physical characteristics of sound field such as reverberation time, brilliance, intimacy and feeling comfort.

Hojan[75] used questionnaire techniques with semantic differential scales that reflect four perceptual attributes : "Clarity", "Sharpness of Localization", "Spaciousness", and "Overall Impression". Impulse responses, binaurally recorded using an artificial head in six European concert halls were subjectively evaluated.

Barron[76] in a recent study, used a short questionnaire with nine bipolar, continuous semantic differential scales to be filled by expert listeners in eleven major British concert halls to make use of the realistic presentation of the sounds. The responses were correlated with objective measurements at the same locations. He concluded that when all subjects were included in the analysis the overall impression with respect to preference was found to be determined by perceived reverberance, envelopment and intimacy. His subjects could be grouped into two groups, one seemed to prefer reverberance while the other preferred intimacy.

Ando and Kurihara[77] conducted tests to determine which horizontal angles are most effective in simulating subjective diffuseness. Results showed that the most effective angle of arrival for minimizing the magnitude of the *IACC* and maximizing the subjective diffuseness depends on the frequency range. For frequencies below 500 Hz, the most effective horizontal angle was found in the range centred on ±90° from the listener's straight ahead while for frequencies 500 Hz and 1 kHz the angle is in the range centred on ±55° and commonly found to be centred on ±20° for frequencies above 2 kHz.

Concerning the subjective evaluations of the musicians on the orchestra stage limited studies had been conducted. Naylor[78] conducted experiments with musicians in simulated sound fields to judge their ease of hearing themselves and others. Amplitude modulation transfer principles were applied to the conceptual model of "Hearing-of- OTHER" and the importance of signal levels to both

"Hearing-of-SELF" and "Hearing-of-OTHER" was emphasised. Architectural implications for musicians' platform design were identified.

All subjective studies agree that the overall sound strength is important in addition to clarity or definition as measures of reverberance and blend. The variation of early decay time as a function of frequency is found to influence the timbre (i.e. the tonal quality) of the perceived sounds.

2.5 INDICATORS OF SPEECH INTELLIGIBILITY IN ENCLOSURES

The ability to accurately predict expected levels of speech intelligibility is a primary concern in judging both the acoustical quality in rooms, and in sound system design applications. Generally speech intelligibility scores may be obtained using a "*Fairbanks Rhyme Test* " which is one of many commonly used speech intelligibility tests; the procedure and the world lists are similar to the ones described and used by Latham[79] and can be found in references[80,81].

Today, there are several types of acoustical measures used as predictors of speech intelligibility. The "Articulation Index", *AI* procedure; originally introduced by French and Steinberg[82] and later refined by Kryter[83] is the simplest one. It is a frequency dependent measure based on the steady state signal-to-noise concept. The *AI* standardized calculation procedure is available in reference[81].

A "Speech Transmission Index", *STI* has been proposed by Steeneken and Houtgast[85] based on the concept of the "Modulation Transfer Function", *MTF* originally introduced by Schroeder[84]. The *STI* measure can be calculated both with and without including the effects of interfering background noise following the procedure outlined by Houtgast and Steeneken[85,86]. For the purpose of a fast evaluation of speech intelligibility in auditoria and public address systems a condensed version of the *STI* method may be used to replace the *STI* lengthy calculations. This measure is termed the "Rapid STI", *RASTI*. The resulting objective index definition, description, precision, limitations and possible applications are given in references[86,87]. *RASTI* measurement procedure is standardized by the International Electro Acoustics Commission (IEC)[87]. The analysis in the *RASTI* method is restricted to the 500 Hz and 2 kHz octave bands and to four and five modulation frequencies in these bands respectively.

Another measure of speech intelligibility was introduced with the concept of "Useful-to-Detrimental" sound ratios obtained from early-to-late sound ratios, speech, and background levels. This measure is based on the weighted "Signal-to-Noise", (SNR_{95} ratio devised by Lochner and Burger[88]), to account for the impact of the reflection patterns of sound in rooms and the masking characteristics of hearing on speech intelligibility under reverberant conditions. A comprehensive study conducted by Latham[79] re-evaluated Lochner and Burger SNR_{95} to account for the effect of fluctuating ambient background noise on speech intelligibility and a modified SNR_{95} was derived.

Bradley[89] discussed and compared three different types of acoustical measures as predictors of speech intelligibility in rooms. He devised an expression for Useful/Detrimental ratio obtained from an early-to-late sound energy corresponding to a specific time limit and the ratio of background noise to speech energies. He concluded as a result of various measurements in halls that an A-weighted signal-to-noise ratio combined with reverberation time *RT* (at 1 kHz) is the simplest, for optimum conditions an *RT* of slightly less than 1.0 second and a signal-to-noise ratio of at least 15 dBA are required. In large rooms with volumes of 10,000-30,000 m^3 ideal maximum background levels of 27-30 dBA are required. He also suggested that the complicated weighting procedure involved in calculating the value of the weighted "Signal-to-Noise" ratio, SNR_{95} is not necessary as its accuracy is not superior to the other less complicated measures. He recommended the use of the first 80 ms "Useful-to-Detrimental" ratio, U_{80} as a preferred predictor of speech intelligibility.

Investigations by Peutz[90] as to the relative contribution of vowels and consonants to speech intelligibility led to the hypothesis that the "Articulation Loss of Consonants", AL_{cons} was a good parameter to measure the quality of speech intelligibility. A simple expression for AL_{cons} was provided in addition, a tolerance value range was suggested based on listening conditions. The accuracy of various available techniques for evaluating speech intelligibility in rooms were assessed and correlated with measured results by Smith[91]. Their respective drawbacks and limitations were also reviewed. Jacob[92] compared the results of subjective intelligibility tests to the predictions of three objective speech intelligibility measures. The *STI* based on *MTF* concept and the SNR_{95} methods were found to predict intelligibility accurately while AL_{cons} was shown to be the least accurate.

2.6 IMPULSE RESPONSE MEASUREMENT TECHNIQUES

The initial step in room-acoustic analysis and assessment is to measure the impulse responses of the space or enclosure under investigation. From the impulse response most of the room-acoustic indicators can be computed without the need for further measurements. The basic idea when measuring impulse response of any linear system such as a room in our case, is to excite the system by a known stimulus signal with certain characteristics. This could be done using a variety of techniques, each based on the sound source type, the excitatory signal itself and the type of receiver used in the measurement. The source signal can be a reasonably repeatable impulsive signal produced by a blank pistol[93] providing it is omnidirectional with adequate sound energy in the frequency range of interest. The source signal can also be a periodic pulse or non-impulsive signal with flat spectrum introduced to the room via a loudspeaker so as to give the same effect of an impulsive source signal. The merits and limitations of using either type of these source signals were well discussed in a comparative approach by Bradley and Halliwell[105].

Barron[94] reviewed the particular requirements for the visual display of the impulse response. Test signal requirements and analysis options for objective measures involving integrated energy were also discussed and the advantages of two basic test signals were reported. Aoshima[95] discussed a computer-generated pulse signal for sound measurements and introduced a pulse expansion (i.e. time stretched) and compression technique. Berkhout *et al.*[96] reported a new method to acquire impulse responses in a concert hall employing a deconvolution technique to a broad-band sweep with high signal-to-noise ratio and high resolution. Measurements were made in two different concert halls to illustrate the practical implications of the proposed new technique.

In practice, there are generally four types of stimulus signals. These are *periodic pulse, white noise, chirp signal* or symmetrical *Maximum-Length Sequence* (*m-sequence*). The periodic pulse is simple and involves less equipment. The basic idea is to stimulate the room under test with a narrow pulse whose width is narrow enough (i.e. less than (1/5f), where f is the system bandwidth)[109]. When the generated pulse is introduced to excite the room the output approximates the room impulse response without further complex processing. The periodic pulse test is usually done by applying a periodic square pulse to the system so as to facilitate the averaging required to obtain a satisfactory signal-to-noise ratio. The system output is then band-limited by an anti-aliasing filter and sampled to yield an approximation to the system impulse response. The main drawbacks of using a periodic pulse have been elaborately discussed by Rife[109].

The impulse response of a linear system may be also obtained by introducing an adequate-level white noise into it, then cross-correlating the noise input to the system output. The system Transfer Function approximately equals the Fast Fourier Transformation (*FFT*) of the output noise divided by the *FFT* of the input noise. The method may appear simple and easy to conduct, but it has disadvantages as discussed in references[109,110] in detail.

Another technique for obtaining the room impulse response, is to excite the room by a deterministic, non-random signal such as linearly frequency modulated sine wave which is known as chirp signal. This method possesses most of the disadvantages of using white noise, in addition, it is still considered complex to accurately generate.

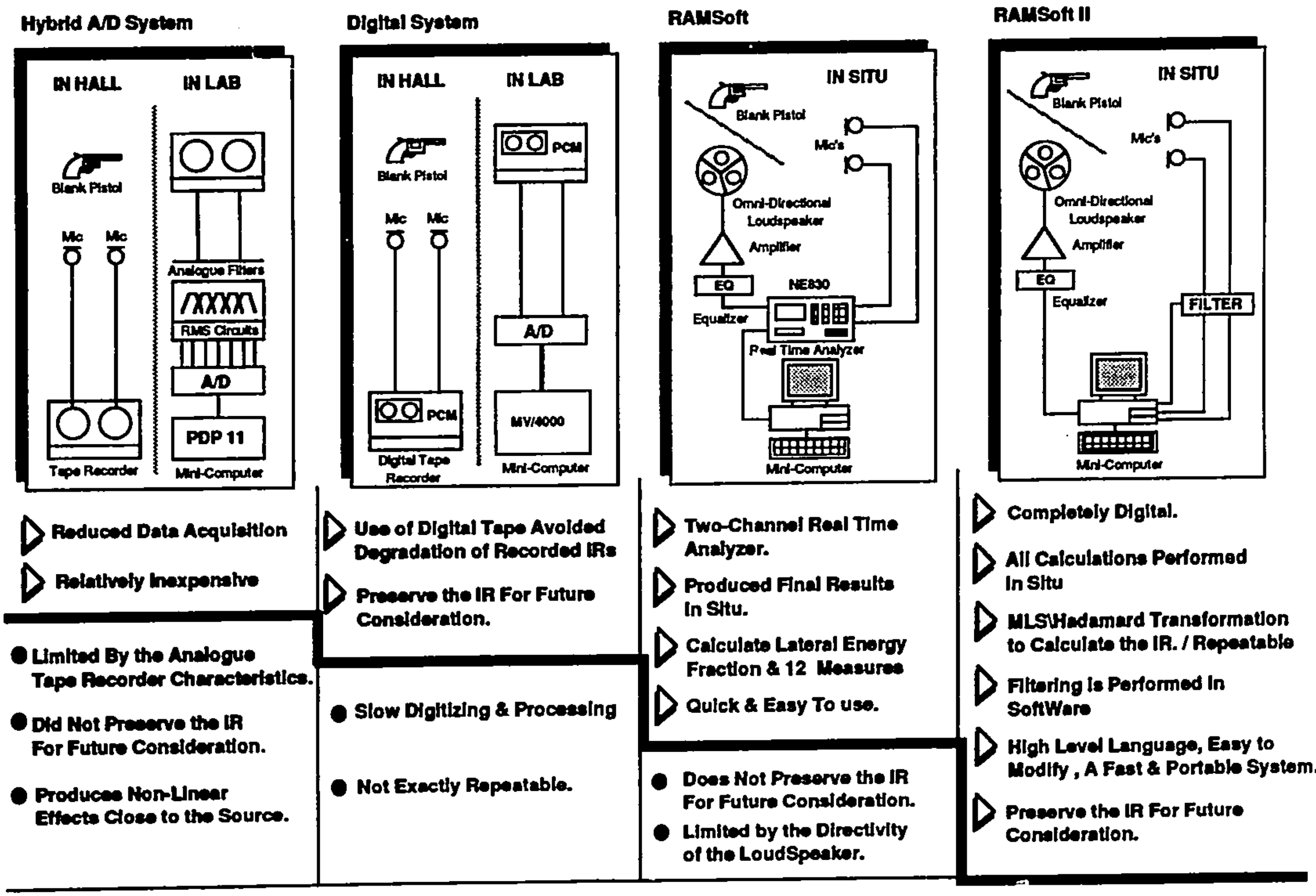
To obtain the impulse response, an approach employed almost two decades ago by Schroeder[97] has recently gained popularity, that is to use a stimulus signal which is periodic, deterministic and has the auto-correlation of a perfect impulse. This stimulus is known as the Maximum-Length Sequence (*m-sequence*). Definition and discussion of properties of the *m-sequence* as a stimulus signal have been extensively reported by Davies[98], and Ziemer and Peterson[99]. The *m-sequence* also possess other advantageous properties which were demonstrated by Rife[109-111] and will be summarized later in Chapter 3. An *m-sequence* can be generated by programming its characteristic polynomial on a personal computer to simulate a shift register stages status. A number of authors[99,100] had generated tables of irreducible and primitive polynomials which yield *m-sequence*. The use of these tables is described in reference[101].

The basic idea of using an *m-sequence* as stimulus is to convert the sequence from its digital form as a binary sequence to an analogue signal with a given amplitude, then to excite the room under test with this deterministic and periodic predefined signal. It is then only, necessary to cross correlate the system response with the same *m-sequence* signal used for exciting the room. A technique for efficient and speedy cross correlation was reported originally by Borish and Angell[102], later by Alrutz and Schroeder[103] and Ando[4] and recently by Chu[104]. However it is also possible to speed the processing operation by performing *FFT* of the *m-sequence* and the system response, multiplying the two *FFTs* then, the inverse *FFT* of the result yields the system impulse response as suggested and used by Schroeder[97].

2.7 COMPUTER-BASED ROOM ACOUSTIC MEASUREMENT SYSTEMS AND INSTRUMENTATION

The foregoing literature survey shows that over the last twenty years, new subjectively relevant objective room-acoustic indicators for evaluating the acoustical quality of an enclosure have been introduced. However, for these objective indicators to be efficiently measured, accurate and reproducibly, a need to develop a new measurement system exists. The early measurements of some indicators in an auditorium were often done by laborious field recording and subsequent computer analysis in the laboratory.

Recently, with the advent of portable computers and signal analyzers, it became more convenient to use them in situ. There are already a few computer-based room acoustics measurement system. The most well documented in North America are the four generations of measurement systems developed at the National Research Council of Canada (*NRCC*) as products of ten years of experience in measuring new quantities to evaluate the acoustical conditions in auditoria. Bradley and Halliwell[105] comprehensively reviewed this experience in a comparative approach. The description of each measurement system, its merits and limitations were also discussed. A summary of these along with depictions of the systems' hardware components are shown in Fig. 2.2. The study also explored philosophies applicable to a measurement system development on a personal computer. Moreover they had identified a number of important aspects which must be addressed by any auditorium measurement system. The most recent generation of the *NRCC's* systems; a two microphone system, *RAMSoft-II* is based on using an *m-sequence* as an excitation signal. The system components description, data filtering, analysis considerations, other features and possibilities were discussed by Halliwell and Bradley[106]. One of their conclusions emphasized the need to further develop and use measurements using an array of microphones so that directional information can be gained from the sound field.



Note : IR = Room Impulse Response

Fig. 2.2 Measurement systems developed at *NRCC*, (Canada) in a ten year experience, (Summary of merits and limitations)[105]. [Adapted]

Another measurement system enabling in-situ calculations of some room-acoustics parameters was developed in Denmark. Hansen[107] described the characteristics and benefits of that system, shown in Fig. 2.3. The system uses a proprietary modular precision sound level meter (Brüel & Kjær *Type 2331)* equipped with an IBM-compatible lap-top computer with a plug-in application software module. A powerful hi-fi amplifier and an omnidirectional loudspeaker constitute the sound source. The system is flexible, gives immediate results, easy to setup and operate, but it is limited by being a single channel system. Therefore, indicators such as *LEF* cannot be measured.

Recently, a powerful single channel system analyzer, called the "Maximum Length Sequence System Analyzer", *MLSSA* manufactured by *DRA*[108] based on the developed technique using *m-sequence* has been introduced. The system consists of a hardware component; a plug-in Board (A/D-160), a software driver and a signal generator and processor. The system provides the most important functions needed in data acquisition, processing and analysis for room-acoustic investigations utilizing the impulse response. In addition, it provides fully programmable digital filters (1/1 octave and 1/3 octave bands). The system components, merits and possible applications were presented and discussed in detail by Rife[109-111].

Marshall[112] briefly described, as a part of an interactive design aid in acoustical modelling, a measurement system called "*MIDAS*" which performs room-acoustic

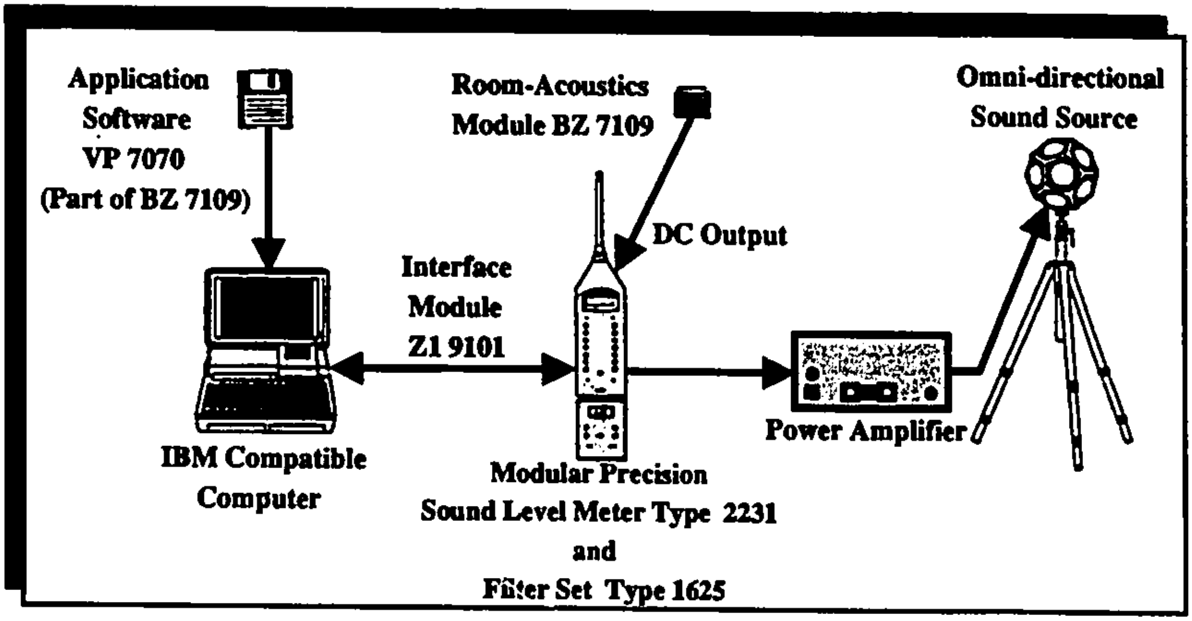


Fig. 2.3 A measurement system, SLM/Lap-Top computer by Hansen[167], (Denmark), [Note: instrument/module type numbers refer to proprietary products available from Bruel & Kjaer, Denmark].

measurements based on *FFT* techniques at full scale or at model scale using a variety of sound sources both impulsive and continuous. The system is implemented on an *Apple Mac-II* computer to compute the full range of accepted auditorium indices. It can be operated in single or dual channel modes. The use of "*MIDAS* " in the evaluation of the new Hong Kong Cultural Centre was demonstrated. Menyial *et al.*[113,114] reported in detail the same commercial computer-based measuring system along with results from specific applications.

Puria *et al.*[115] described a measurement system "*SYSid*", (SYStem Identification) developed at AT&T's Bell laboratories. The system is a software package that runs on DOS platform with the *Ariel DSP-16* dual channel data acquisition and signal processing peripheral. The *SYSid* excites the system (e.g. a room) being measured with a stimulus and synchronously averages the measured response of the system; the *FFT* technique is then used to deconvolve the stimulus from the measured response and further analyze the data. The system, if in the dual channel mode, is limited to only 8192 data samples per channel which is quite short to fulfil time domain requirements e.g. for capturing long reverberation time and simultaneously cover the frequency range of interest in room-acoustic applications.

Siebein *et al.*[116] reported a measurement system, "*ARIAS*", "The Acoustical Research Instrumentation for Architectural Spaces", for obtaining monaural and binaural objective room acoustic indicators. The *ARIAS* system consists of 1/2" microphones with preamplifiers, a multichannel 12-bit analog to digital convertor system with simultaneous sampling and an 486-based personal computer with an IEE-488 interface. The *ARIAS* software runs on IBM compatible computers running MS-DOS and Microsoft Windows. However the system is limited to processing responses to approximately impulsive sources (i.e blank pistol shots), therefore other excitation signals such as an *m-sequence* cannot be employed.

Tachibana *et al.*[117] demonstrated a measurement system using a technique for the measurement of the impulse response not only in real auditoria but also in scale models. Their measurement system was described along with some example applications were demonstrated. A block diagram of their system operational sequence is illustrated in Fig. 2.4.

Measurement systems which have been described earlier are mainly developed to measure the contemporary acoustical indicators, therefore, being single or dual channel systems is enough for this purpose. Nowadays, with well established objective room-acoustic indicators, research work in room-acoustics investigations is shifting to the study and development of spatial information of sound fields in rooms.

Earlier, Broadhurst[118] studied sound directivity using a group of microphones arranged in s solid-grid form in space. The array was shown to be capable of locating the direction of sound reflections in a room, but in addition to hardware complexity, the technique and data processing is time consuming.

Recently, Endoh *et al.*[119] have developed a technique by which they are able to obtain the spatial information, especially that of the early reflections period. The measurement technique developed by Yamasaki and Itow[120] employs a four-channel microphone array to determine the virtual image source positions and directivity patterns. Powers of virtual image sources are then calculated by a correlation technique. The technique gives new insight and valuable information about the directional characteristics of sound in enclosures.

Sekiguchi *at al.*[121] developed a sound measurement method by applying Yamasaki and Itow's technique to four microphones arranged close to each other at the apex of a regular tetrahedron but utilizing a deconvolution method instead of correlation to improve time resolution and shorten the calculation time. Fig. 2.5 depicts a block diagram of the main components for the commercial version of Sekiguchi's system, called the "*FOURMIC* "[122] measurement system.

Ikezawa and Nishi[123] developed a new multi-beam array microphone, "*MAM*". The MAM is composed of concentric rings, each of which has 12 microphone elements, and a common centre microphone. This configuration allows 12 main beams pointing in 12 different directions at intervals of 30° in the horizontal plane. The *MAM* can be applied to precise acoustic measurement of sound fields and reproduction employing simultaneous multi-directional sound pickup. A tone-burst

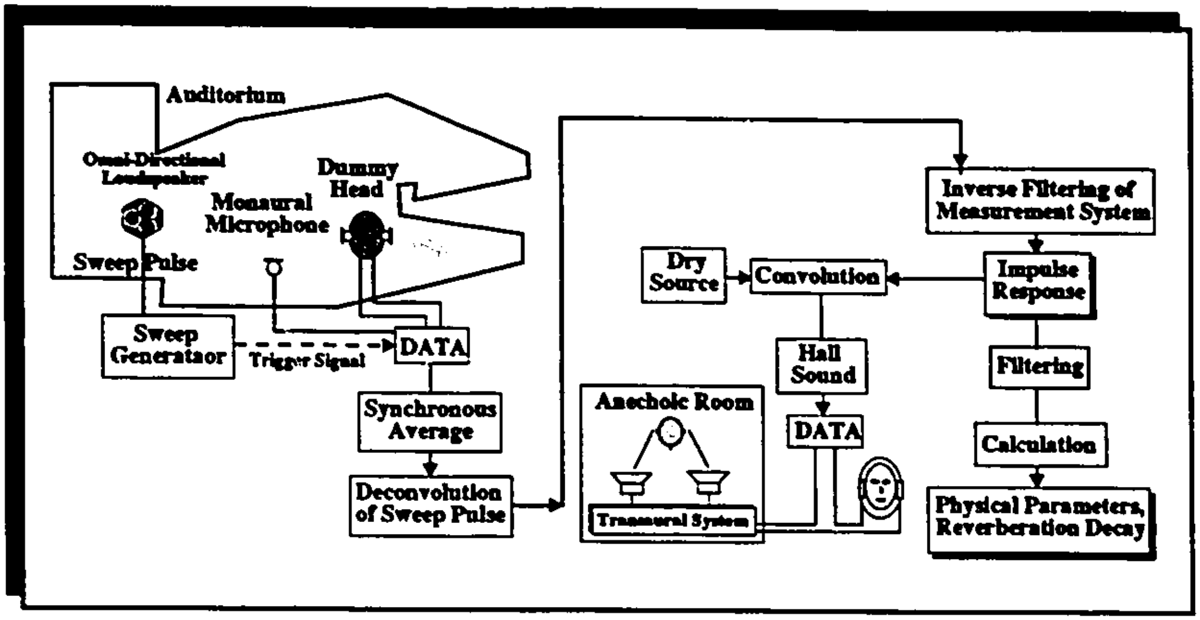


Fig. 2.4 Block diagram of a contemporary measurement system, components and operational sequence, Tachibana *et al.*[117], (Japan). [Adapted]

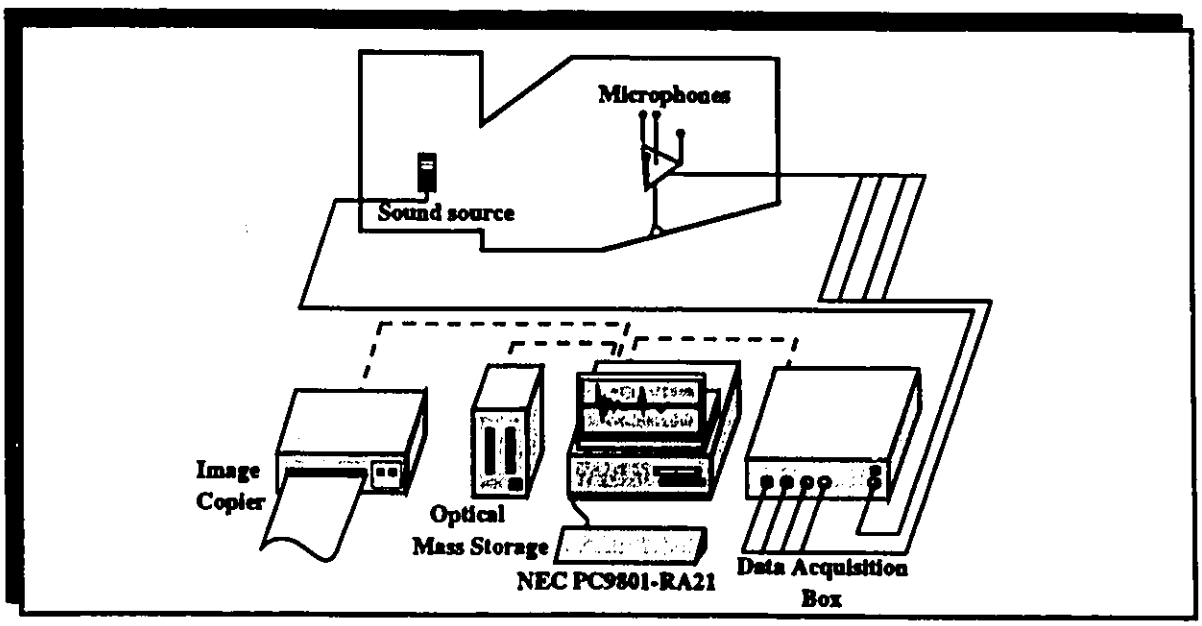


Fig. 2.5 Block diagram of the FOURMIC measurement system[122], hardware components, Sekiguchi et al.[121], (Japan). [Adapted]

signal is used as a test signal. However, in addition to the complexity and sophistication of *MAM*, emphasis was put on horizontal sound capture and reproduction, therefore this system is less practical for obtaining spatial information.

With the existence of many measurement systems for room-acoustics evaluations, it was interesting to study the systematic errors of the effect of measurement procedure details and equipment on the variations of various acoustical measures, Bradley *et al.*[124,125] compared the results of three measurement teams who independently measured the same halls. Systematic differences in the results due to the measurement systems' elements and the different procedures employed were found and represented by the mean and the standard deviations of differences.

2.8 CONCLUSIONS

Over the last two decades, many objective room-acoustics quantities or indicators for evaluating the acoustical quality of an enclosure have been introduced. Numerous subjective aspects of the listening experience in auditoria, particulary concert halls can be described well by such objective indicators or measures. Some of these indicators relate to reverberance, music definition, clarity and blend, others to spaciousness or spatial impression, some to speech intelligibility and others to acoustic conditions at performers' locations.

Studies in several countries have reported extensive measurement results for evaluating existing auditoria. The newer room-acoustic measures could be sub-grouped based on the important subjective attributes widely accepted to be essential for acoustical goodness or appreciation into the following subgroups:

- Indicators relating to the subjective reverberance; those are, "Reverberation Time", *RT*, and "Early Decay Time", *EDT*.
- Indicators relating to the subjective balance and blend between clarity and reverberance; those are ,the Early-to-Late sound Index, "Clarity", C_{50}, or C_{80}, "Definition", D_{50} and "Centre Time", T_S.
- Indicators relating to the subjective sound loudness such as "Relative Level" or "Strength", *G*.
- Indicators relating to the subjective "Spaciousness" or "Spatial Impression", *SI*, and/or "Apparent Source Width", *ASW*, both can be judged by "Early Lateral Reflections", *LEF* and/or "Inter-Aural Correlation Coefficient", *IACC*.
- Indicators of speech intelligibility ; those are: "Speech Transmission Index", *STI* and "Rapid Speech Transmission Index" *RASTI*, weighted "Signal-to-Noise Ratio" SNR_{95}, "Useful-to-Detrimental Sound Ratio", U_{80}, and "Articulation Loss of Consonants" AL_{cons}.
- Indicators of Musician Acoustics; those are: "SUPPORT" ST_{100}, ST_{200}, "Early Decay Time" versus frequency *EDT(f)*, and Modulation Transfer Function" *MTF*.

Early-to-late arriving sound energy ratios are related to the subjective clarity of the sound. A low value indicates a poor definition referred to subjectively as "*muddy*" sound, while a high value indicates that it is possible to discriminate details but also that the sound may be subjectively "*very dry*" as if it is produced in a room with too much sound absorption. "Definition" is preferred for speech and "Clarity" and "Centre Time" for music. The indicator "Relative Strength", *G* is related to the enclosure ability to amplify sound from the stage and thus provide a larger dynamic range. It could be also used to compare results from different positions as this shows the sound distribution in the hall under investigation.

Most of the newer indicators are based on sound energy ratios calculated from the room impulse response and except for *LEF* and *IACC* do not touch upon the sound directional characteristics. Numbers of them are found to be inter-related and to some extent considerable overlap is exhibited; this implies that knowing a few will allow others to be deduced. They have also been correlated to overall geometric variables such as room width, height and wall angles; and empirical models for their prediction based on statistical analyses already exist. Techniques for accurate measurement of these indicators have become common however some of them are position dependent and more sensitive to the early reflection sequence, subsequently they are dependent on both the overall geometrical characteristics of the space and architectural details in the vicinity of the measurement location.

The value of such indicators give new insight into the acoustical goodness of a receiver position but in order to design halls, or to correct an acoustical defect in an existing enclosure, there is a need to quantitatively understand the extent to which they are influenced by the various physical design features of the enclosure. The effect of the enclosure interior design from the point view of cause and effect related to objective room-acoustic indicators, although usually appreciated and desired, are not yet fully investigated. In order to develop this understanding or knowledge, indicator values must be obtained together with directional characteristics information; thus a need exists for suitable comprehensive instrument development. The purpose is to link the measured indicators to the interior physical features and to assess diagnostic capability with respect to the effect and its cause.

Bradley[9] commented that situations can be encountered where visually obvious changes in a room that are thought to have significant effects on the listening experience may result in minor changes and are unlikely to be audible. Others[126] encountered impulse responses captured in particular locations in an auditorium, where delayed distinguishable reflections that affected the clarity value were observed, yet the directions of such reflections were not readily identified or beyond investigation due to lack of a measuring diagnostic tool.

Most objective room-acoustics indicators are based on the capture and subsequent analysis of room impulse responses but subjective criteria are also influenced by spatial distribution of sound energy. The spatial distribution of sound energy is usually not considered due to lack of an efficient, accurate and easy to perform measurement procedure.

The measurement of sound spatial information in rooms would help in building supplementary knowledge to better understand conditions that are critical for acoustical quality and moreover it would contribute towards diagnostic capability, especially in large reverberant enclosures. This spatial information if obtained and then linked with measured values of the objective room-acoustic indicators for the same listener location would provide a better picture of the acoustical quality and may also help interpret their values in a more reliable way leading to remedial treatments. Commonly, in a case where the acoustical quality in a hall is found not satisfactory or bad at particular location(s), as a result of evaluating known room-acoustic indicators, the cause(s) for such defects may be partially identified. However the direction-related physical features that contributed to the defect(s) are not readily identified for a complete diagnosis.

With the advent of portable computers and signal analyzers, sophisticated data acquisition, evaluation of newer room-acoustic indicators can be made in situ and a number of commercial and research PC-based room-acoustics measurement systems (previously described, see section 2.7) are available; each possesses some merits and exhibits some shortcomings. However, most of the existing systems have been developed primarily to measure newer room-acoustic indicators and while some of them display the basic characteristics or potential for directional sensing, namely, digitized raw data capture, software driven and post processing, and two or more channel acquisition, this step has not yet been taken. In addition, their current frequency range of application is restrictive. The systems are either low frequency limited (125 Hz) by a lack of low frequency energy content of the source, or upper frequency limited (4 kHz) by the temporal length of the digital signal able to be captured and processed. Thus a need exists for a measurement system which improves on existing systems with respect to room-acoustic indicators' evaluation and also to obtain and assess sound directionality.

Earlier attempts to measure directionality of sound fields in enclosures were difficult, time consuming and required complicated equipment. In the early nineties, the idea of obtaining sound field spatial information has been revived by research teams in Japan[119-122]. They have introduced new measurement techniques (i.e. correlation[120] and deconvolution[121]) with less hardware which give new insight and valuable information with respect to the directional characteristics of sound in enclosures. Although these techniques have been applied in existing auditoria, no attempt has yet been made to further establish relationships between known room-acoustic indicators and the directional characteristics of the sound field. Moreover the usefulness of obtaining sound energy directional distribution for acoustic problem diagnostics or quantification of sound diffuseness in a way that allows ready evaluation and interpretation in conjunction with the values of room-acoustic measures are not touched upon.

CHAPTER 3

"CBS-RAIMS" : A NEW MEASUREMENT SYSTEM

3.1 INTRODUCTION

As a result of the literature review described in Chapter 2, section 2.7 concerning both newer room-acoustic indicators and available measurement systems, it was concluded that a need existed to develop a new comprehensive measurement system. The primary objective being to obtain and analyze the directional characteristics of sound fields in large enclosures together with known objective quality indicators.

Existing measurement systems with exceptions (Yamasaki[120], and recently Sekiguchi[121]) were developed primarily to measure known room-acoustic indicators, therefore their hardware and software considerations were oriented toward this objective. The measurement system based on Yamasaki's measurement technique demands lengthy computations and consequently required the use of a mainframe computer for timely data processing, although very recently this problem has been avoided by a software arrangement allowing the use of a stand alone system with a personal computer. The computational complications and the scarcity of published information dictated against this approach.

3.2 REQUIRED ATTRIBUTES OF THE MEASUREMENT SYSTEM

The intended system is required to possess features in advance of those presently in use. It should be portable and efficiently perform the following functions.

- Capture the enclosure response to various stimulus signals.
- Calculate the room impulse response.
- Perform subsequent signal analysis (time - frequency domain) in a routine manner.
- Easily calculate a number of potentially useful room-acoustic indicators from the collected data, preferably in situ.
- Provide reproducible measurements.
- Measure spatial information of sound at listeners' locations in the enclosure under investigation. This requirement dictates the use of microphone arrays (i.e. multiple channels).

The development of the complete measurement system can be broken down into three major tasks or modules; these are: *1.* hardware interfacing for data acquisition control and synchronization which in turn includes analog data acquisition module, stimulus signals analog output module, a calibration module and other supplementary utilities; *2.* design and coding of a programmable stimulus signal (i.e. *m-sequence*) generator; *3.* algorithms development of all necessary processing and analysis modules. Fig. 3.1 depicts a block diagram of the measurement system development components.

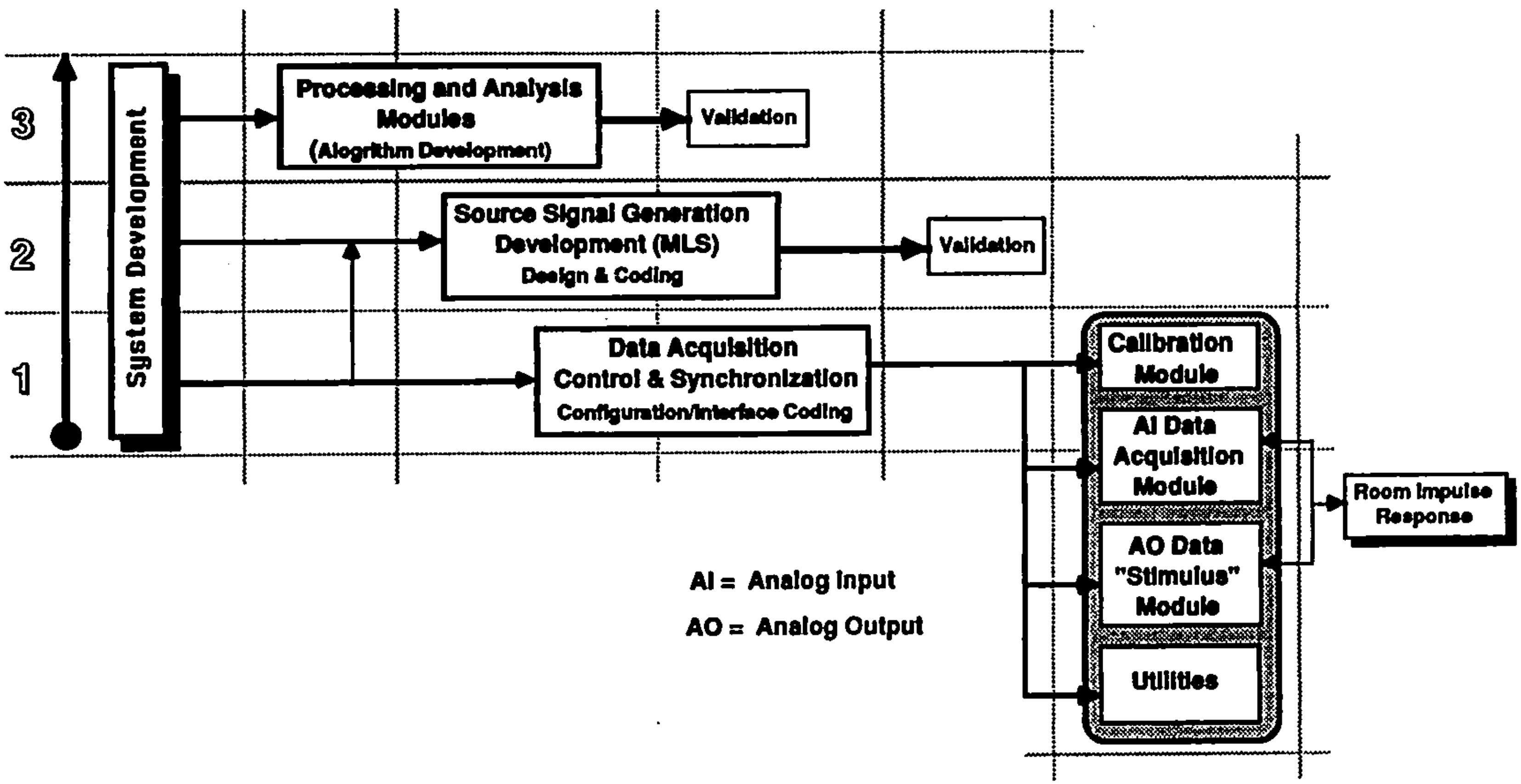


Fig. 3.1 Block diagram of the measurement system development components.

3.3 ARCHITECTURE OF "*CBS-RAIMS*": HARDWARE COMPONENTS

To fully evaluate sound quality in an enclosure, it is required to calculate a number of potentially useful room-acoustic indicators preferably in situ, provide reproducible measurements, and to provide directional information in the enclosure in a manner which allows ready interpretation. The measurement system must be accurate, flexible, easy to calibrate, update, set up and operate.

The *CBS-RAIMS* (Centre for Building Studies-Room Acoustic Indicators Measurement System) is a comprehensive system for obtaining known room-acoustic indicators together with the directional characteristic of the sound field. It involves software developments together with hardware interfacing.

The measurement system is based upon acquiring spatial information at the listener location by using an array of 3 microphone pairs arranged in cartesian coordinates or by sequentially measuring in three directions employing one microphone pair. By exciting the enclosure under test with a periodic and well defined signal i.e. Maximum Length Sequence (*m-sequence*), six impulse responses can be calculated by a fast Hadamard transformation algorithm and their results presented for preliminary examination at the point and time of measurement. This enables on site validation to be made. Post processing the calculated impulse responses then yields the values of most room-acoustic indicators as well as providing a sufficient averaging base to further enhance the signal-to- noise ratio. Subsequent signal analysis can also be performed which involves digital filtering, energy analysis, intensity evaluation, and correlation analysis. The intensity and correlation analysis will be essential tools for subsequent directional information measurement and evaluation.

An impulsive sound produced either by a blank pistol or via a loudspeaker can also be utilized, however in the case of employing any non-reproducible source, the six microphone probe should be employed for signal capture. The system is also equipped with a triggering device to be used if impulsive sound is desired while maintaining synchronized triggering with the data acquisition process.

The measurement system operates in two stages; *the first* is data acquisition or collection, and *the second* is data processing and analysis. The system is presently based on a portable AT-386 computer 33 MHz and the main hardware components and interface scheme are shown in Fig. 3.2. The system employs an eight channel signal conditioning board (*SCXI-1140*)*, and a data acquisition board (*AT-MIO-16F-5*)* both driven by a software driver. The signal conditioning board allows simultaneous channel sampling using the "Hold and Sample" facility; this feature is useful for preserving inter-channel phase relationships thereby enabling subsequent sound intensity evaluation. At the same time as the analog input channels are scanned simultaneously, the board analog output transmits an excitation signal (*m-sequence*) of user selected length up to 65355 samples. This

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* Products of National Instruments, Texas, USA.

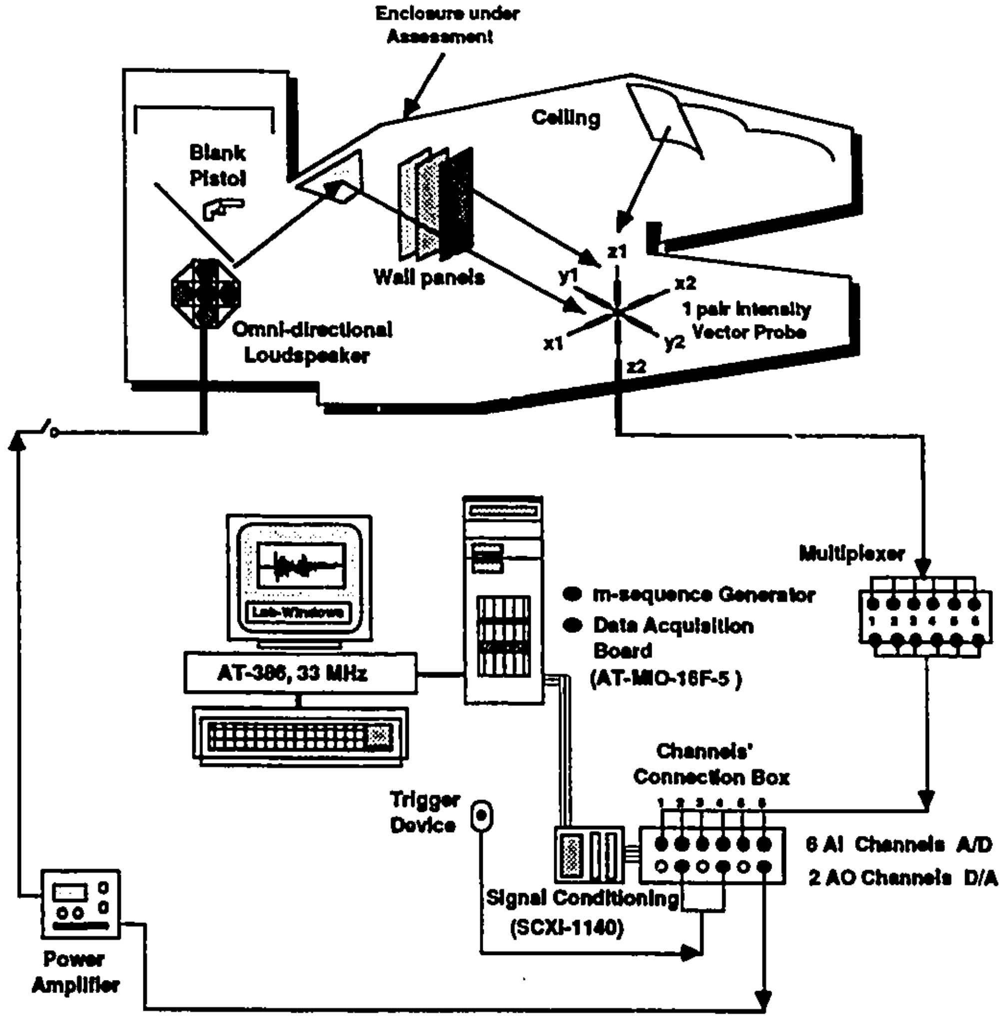


Fig. 3.2 The hardware components and interface scheme of the *CBS-RAIMS* measurement system.

allows the capture of lengthy impulse responses to cover late reverberation in large reverberant enclosures. As a result of experience gained in development and use it is currently intended to upgrade the current system to take advantage of newly available PC's and real time signal conditioning, acquisition and processing boards. The new system is depicted in Fig. 3.3

The transducers currently used are 1/4" pressure microphones (B&K Type 4135) mounted on an appropriate three-dimensional intensity vector probe (B&K Type WA0447) or one pair of 1/2" microphones (B&K Type 4165) in an adjustable holder. The half inch pair arrangement offers the advantages of higher microphone sensitivity, ease of microphone spacer adjustment and calibration, plus higher sampling rate should that be needed (currently up to a total of 143,000 samples/second). It should also be noted that the 3-D probe dimensions may cause distortion of the incident sound field at higher frequencies (i.e. 4 and 8 kHz) due to interference and diffraction effects or intensity assessment error due to the fixed microphone array spacing; thus if the full frequency range from 63 Hz to 8 kHz is of interest then one microphone pair, possibly with a frequency to microphone spacing adjustment, is a better prospect; in this case however, particular attention need be paid to maintaining the geometric centre upon probe reorientation. A suitable single microphone pair holder has been constructed in-house (Fig. 3.4) to allow three successive measurements in orthogonal directions to be made while maintaining the geometric centre and moreover presenting an non-obstructive longitudinal symmetric perspective to the chosen axis of

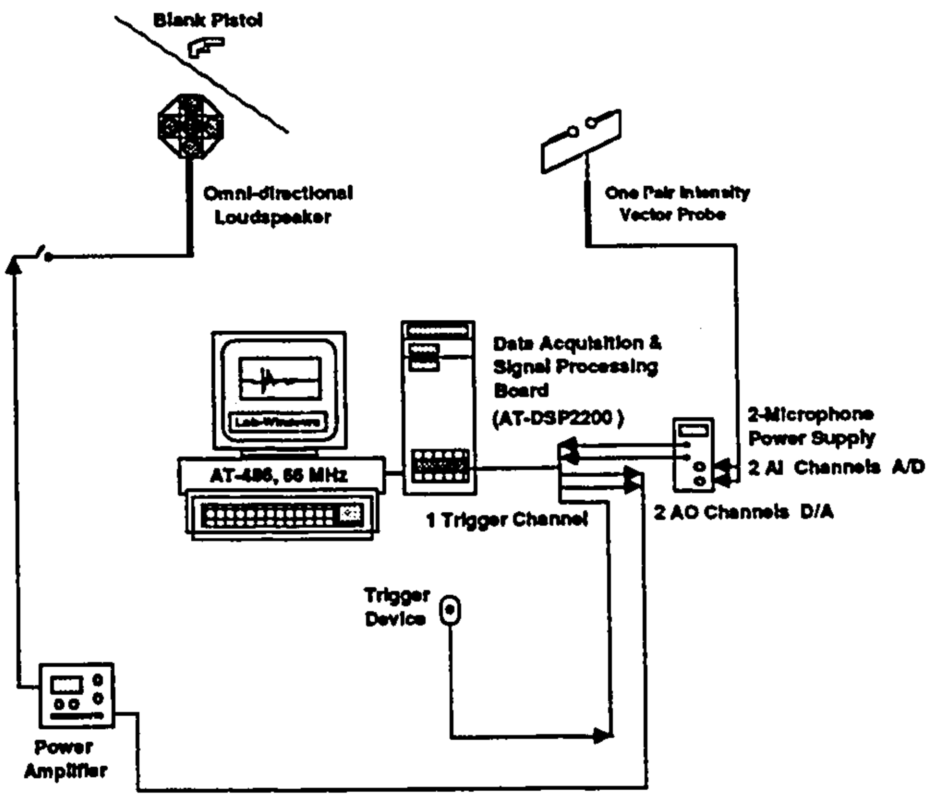


Fig. 3.3 The revised hardware and interface scheme of the measurement system.

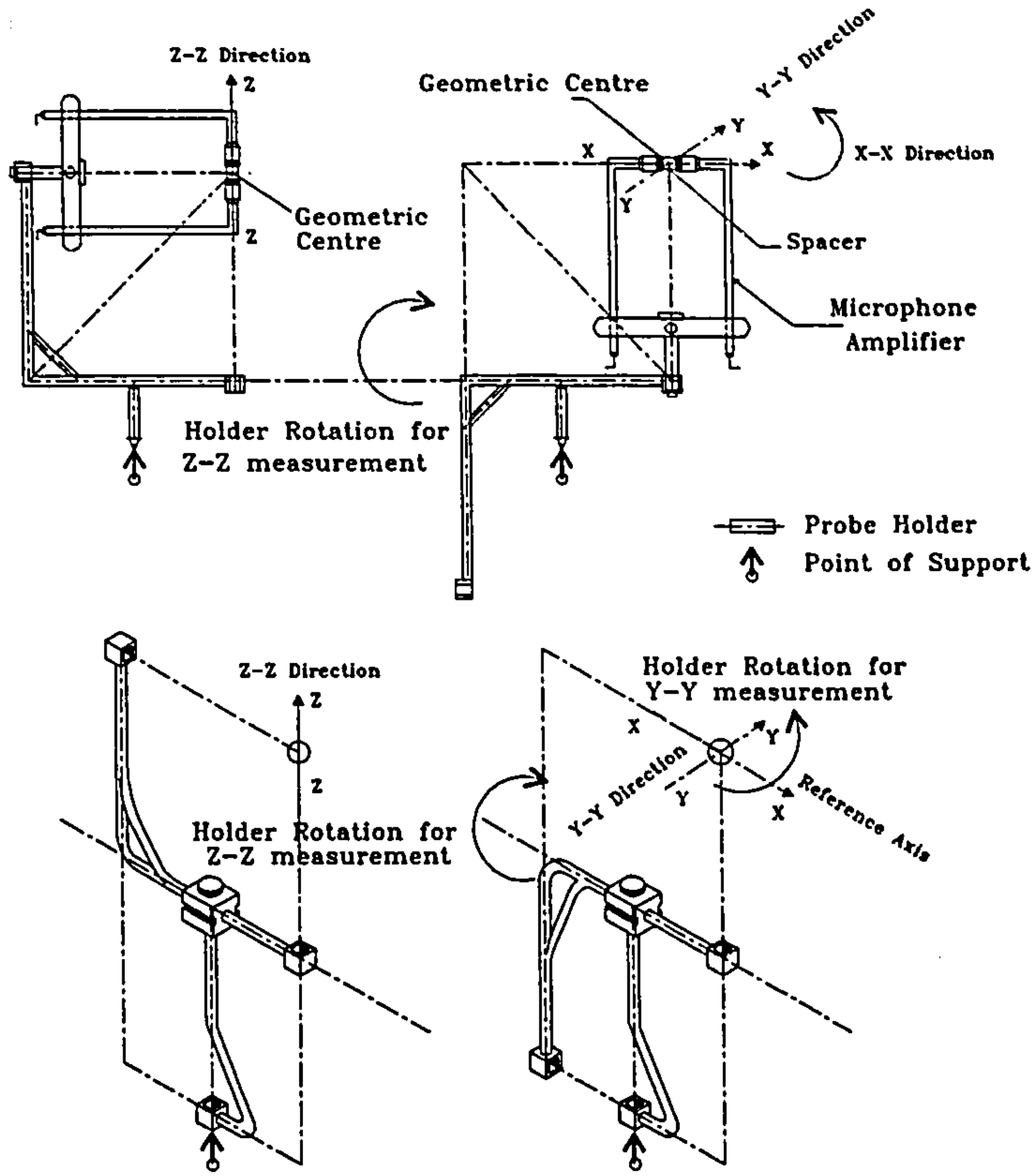


Fig. 3.4 "Three-axis" holder for a single microphone pair.

measurement, thereby minimizing holder effects; this arrangement also lends itself to bias error minimization via probe reversal should higher accuracy (typically directional) be required[156]. Reverse measurements may also avoid the necessity of frequency based probe spacing adjustments.

The system analog output channel is connected to a power amplifier (Currently professional power amplifier QSC, USA 1300 watts) which in turn feeds the signal to an isotropic sound source.

3.3.1 Sound Source Design and Characteristics

ISO/CD 3382 (1993) Draft Standard[127] allows different sound sources to be used to excite the room under test in order to evaluate sound quality by employing newer room-acoustic indicators. One source may be a loudspeaker capable of radiating omni-directionally a broad-band noise spectrum. However broad-band noise excitation puts severe requirements on the power handling capacity of the loudspeaker system to maintain the required signal-to-noise ratio; a sufficient ratio is required to achieve an adequate decay range without contamination by background noise in each octave frequency band and without signal distortion.

Initially the analog source generation channel of the measuring system was connected to a power amplifier (B&K Type 2706) which in turn fed the signal to an isotropic sound source. The loudspeaker (B&K Type 4241), consists of a high frequency unit composed of 12 loudspeakers mounted in a dodecahedral body and a low frequency unit. The sound source is isotropic within ± 3 dB for frequencies below 1 kHz. The power of this source is limited particularly at low frequencies, thus synchronous averaging is necessary to improve the signal-to-noise ratio in the frequency bands of interest.

An ideal sound source should be able to produce a peak sound pressure level sufficient to ensure decay curves starting at least 45 dB above the background noise in the frequency bands of interest. Thus to minimize the distorticn of the input signal, a two unit loudspeaker system with a power rating of 240 watts was constructed. This source is currently used as a standard sound source to cover a wide frequency range while maintaining isotropic characteristics. It has been constructed for the acoustic reproduction of the stimulus signals such as white noise, periodic impulses, m-sequence or sweep pulses for the purpose of room excitation and is designed to radiate sound equally in all directions. To facilitate these requirements, the signal is passed through an electronic cross-over filter which separates its low and high frequency components. The low frequency signal is reproduced by a 300 mm diameter Cone Type loudspeaker inversely mounted in a sealed wooden enclosure with a total volume of 0.15 m^3. The speaker high frequency unit is composed of 12 small Cone Type loudspeakers each of 110 mm diameter optimally placed on a dodecahedral wooden case to make a single high-frequency loudspeaker unit. The dodecahedral unit is mounted about 500 mm above the low-frequency enclosure. Both the low and high-frequency units can be operated simultaneously or separately as required. Fig. 3.5 shows the sound source design configuration with both high and low frequency units.

The directional characteristics of the speaker system has been investigated in a free field measurements according to ISO 3745, 1977(E)[128]. The frequency response of the combined high and low-frequency units resulting from a constant input voltage is shown in Fig. 3.6. The directional characteristics of the loudspeaker have been measured in the horizontal plane in a free field over a reflecting surface employing excitation with octave bands of pink noise. The measuring microphone was located in the far field of the source. Results are shown in Fig. 3.7 which indicate that the directional characteristic is spherical within ± 3 dB for frequencies below 1 kHz. The loudspeaker omni-directionality when averaged over segments of 30° arcs satisfies the maximum directional deviations allowed by ISO/CD 3382. For frequencies less than 500 Hz (i.e 63, 125 and 250 Hz) the deviation of the sound pressure from omni-directionality around the combined unit is found to be less than ± 0.5 dB and it increases to ± 2.0 dB at 500 Hz and to ± 3.0 dB at 1 kHz. The deviation increases with frequency and is of the order of ± 4.5 dB for frequencies greater than 1 kHz and ± 6.0 dB for frequencies greater than 4 kHz. The sound pressure variations around the speaker system in the vertical plane were not investigated, however it is expected to be greater than those in the horizontal plane as a result of reflections from the low-frequency wooden enclosure cubic shape. In general, it can be said that the spherical radiation and maximum power performance of the loudspeaker system

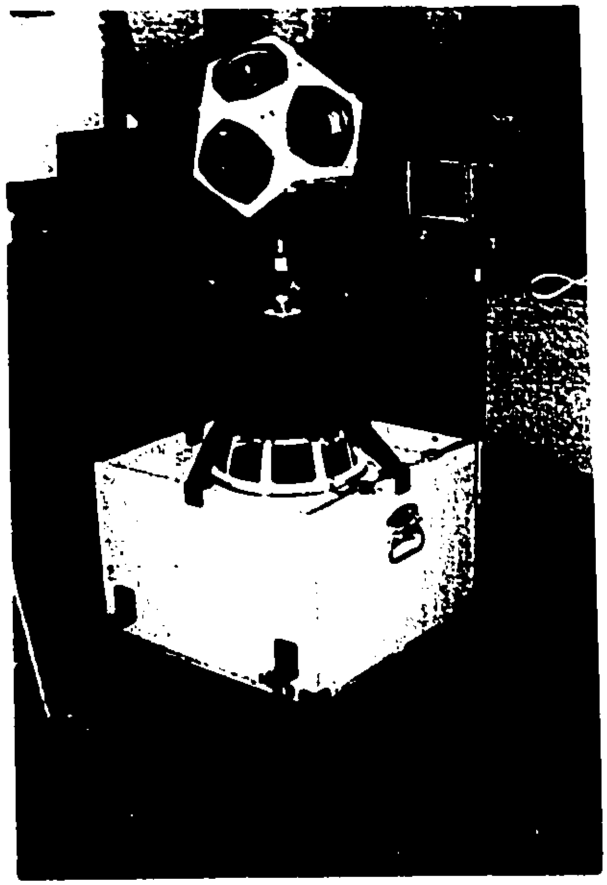


Fig. 3.5 The sound source design configuration with both high and low-frequency units.

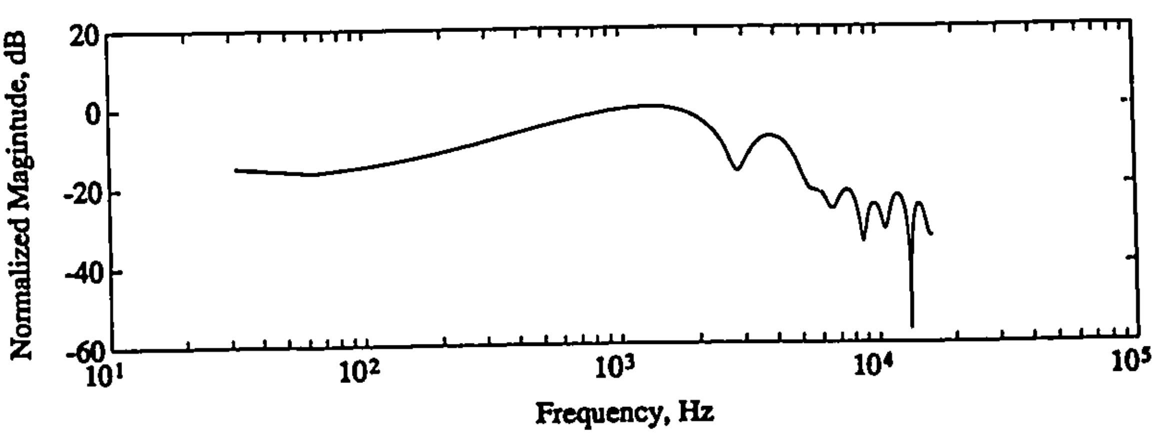
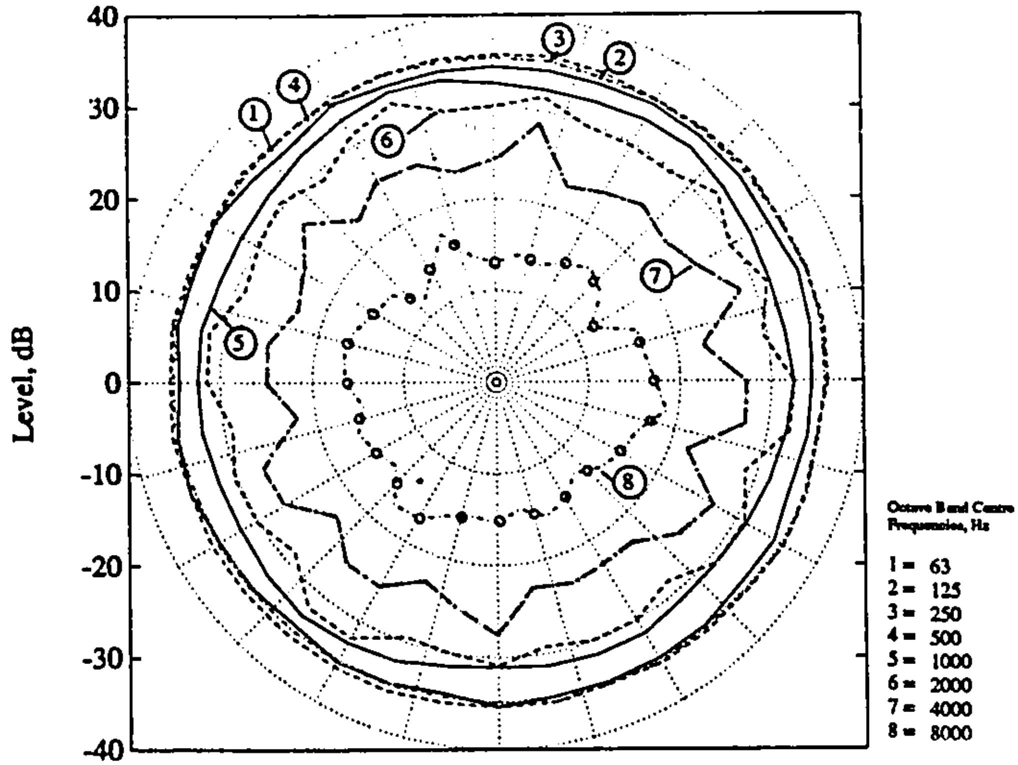
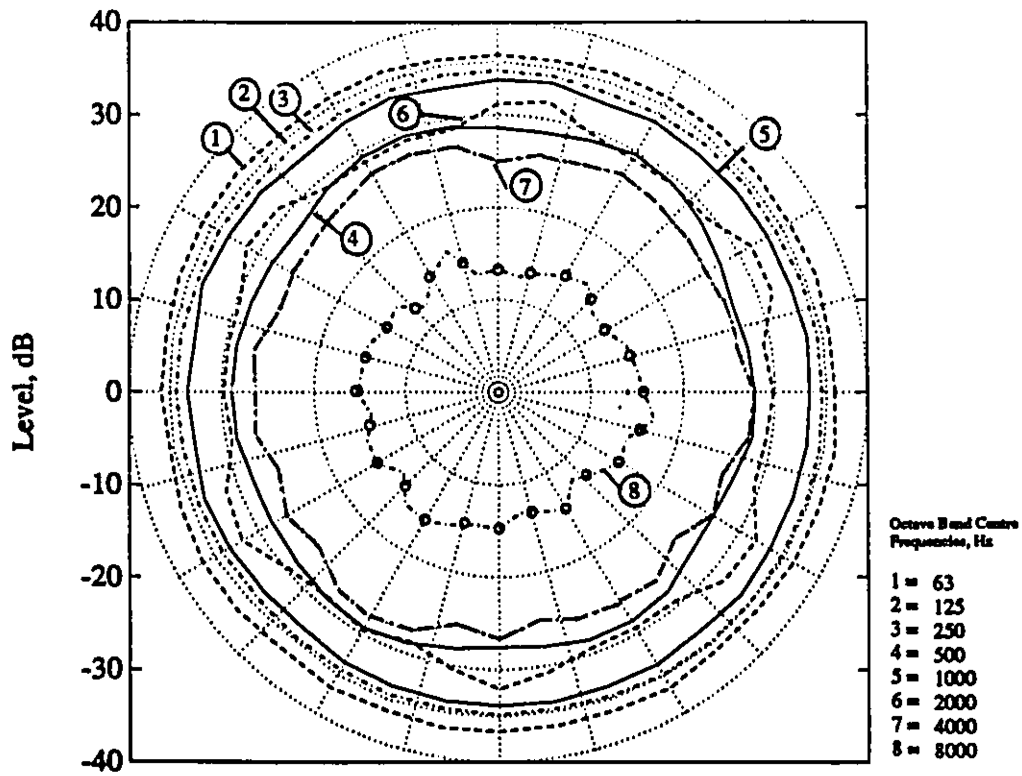


Fig. 3.6 The frequency response of the combined high and low-frequency units of the constructed loudspeaker.



a) High-frequency unit.



b) Combined low and high-frequency units.

Fig. 3.7 The directional characteristics of the sound source system (omnidirectional loudspeaker) in the horizontal plane, a) High-frequency unit and b) Combined low and high-frequency units.

is satisfactory for the purpose of room-acoustic evaluations.

3.4 SOFTWARE COMPONENTS

3.4.1 Source Signal Generation and Validation

CBS-RAIMS development is based primarily upon employing an *m-sequence* as the room excitation signal for the reasons discussed below, however three other sources can also be utilized. These are: white noise; time stretched pulses (i.e Sweep signals) and an impulsive sound produced either by a blank pistol or via the isotropic loudspeaker. The white noise is a segment of 32767 samples stored in a file for subsequent D/A conversion when in use. Two Sweep signals ($Sweep_w$[95] and $Sweep_p$) can be generated and employed by the system; the first 4096 samples of both signals are shown in Fig. 3.8(a), (b), $sweep_p$ is low-frequency pre-emphasized as shown in Fig. 3.8(c) but the spectrum of both signals covers frequencies up to the upper limit of the 8 kHz octave band frequency.

An *m-sequence* is an ideal stimulus compared to the other excitation signals discussed earlier (see section 2.6) as can be judged from the following points:

1. An *m-sequence* is a deterministic signal, that is, it can be pre-computed once and stored for further usage, therefore **there is no need for it to be measured simultaneously with the system output as should be done when white noise stimulus is used.**

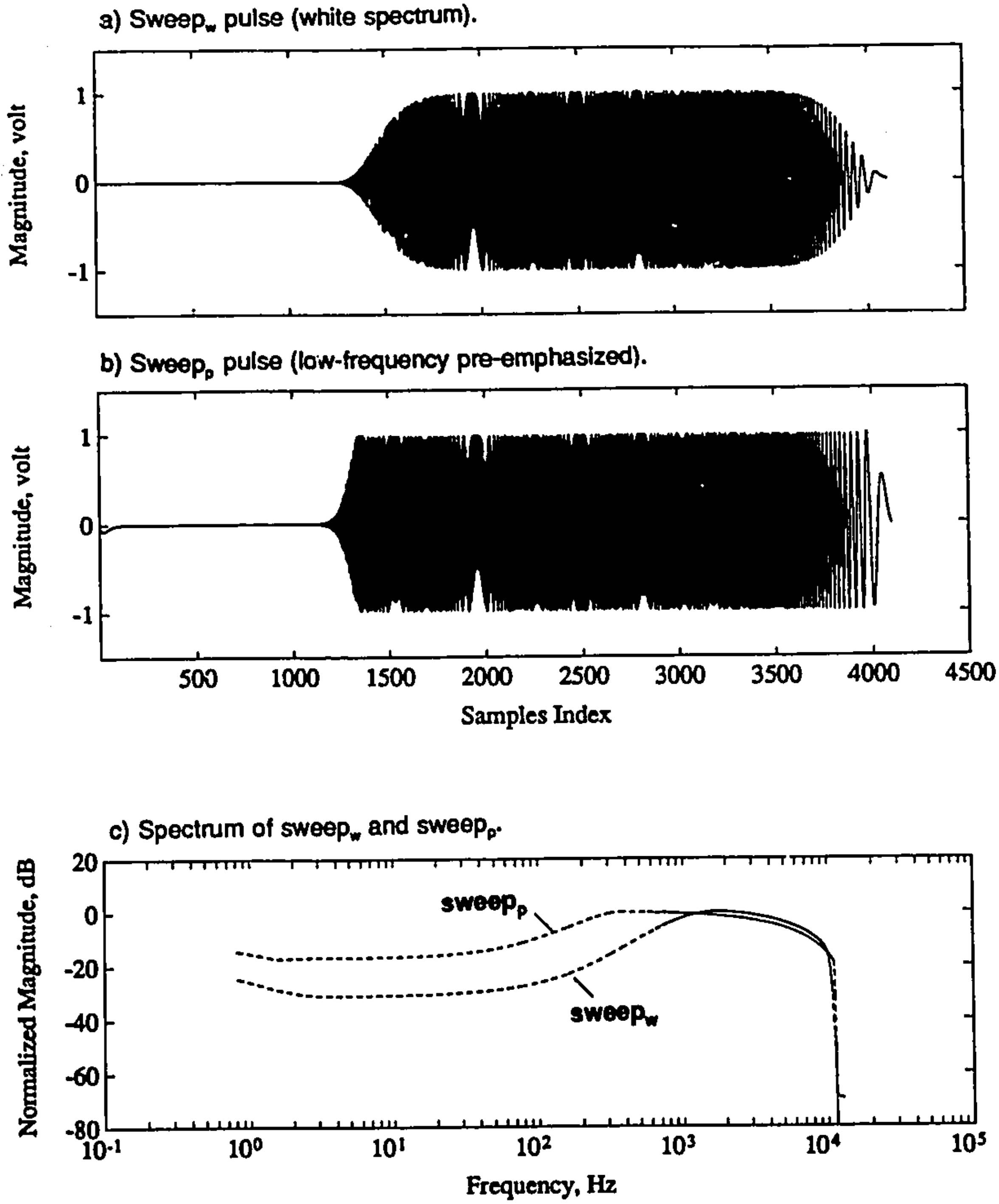


Fig. 3.8 Sweep pulses employed by the *CBS-RAIMS* for room excitation, a) Sweep$_w$ pulse[85] (white spectrum), b) Sweep$_p$ pulse (low-frequency pre-emphasized), and c) Spectrum of sweep$_w$ and sweep$_p$.

2. Under normal conditions the impulse response obtained from an *m-sequence* measurement is identical to that obtained directly by periodic pulse excitation, with the exception that **the system and instrumentation hardware noise and distortion immunity is highly improved.**

3. An *m-sequence* is a periodic signal, this implies that **no windowing is required to calculate the system impulse response** as long as its entire period (i.e. $L = 2^n - 1$) is used to undertake the measurement, where, L is *m-sequence* length and n, the sequence order.

4. Because of the periodicity of a *m-sequence*, both **cross and auto-correlation evaluations are improved in speed.** The use of the binary property (i.e. two possible states ± 1) of *m-sequence* greatly reduces the complexity of the multiplication operation needed to calculate the cross-correlation function. In addition, the cross-correlation can be implemented by the fast Hadamard transformation algorithm[102].

5. The *m-sequence* approach allows wide-band impulse response measurements containing up to 65535 samples. **This feature facilitates the capture of long-time transients (e.g. up to 5 seconds) with the proper hardware and sampling rate settings.**

6. The most important properties of an *m-sequence* for the purpose of room-acoustic measurements is that its *FFT* has the same magnitude for all frequency components, i.e. it has a perfectly flat spectrum except for the D.C. component. This means that an *m-sequence* has L constant magnitude pulses per period, instead of a single pulse. The MLS delivers $(2^n - 1)$ times more energy than a corresponding periodic pulse of equal amplitude, therefore it contains the maximum possible energy for a given peak amplitude; in consequence **the problem of radiating or exciting the**

resolved. Practice demonstrates that single impulse devices such as blank pistols and electric sparks, although powerful, may not contain this flat spectrum.

7. The power of an *m-sequence* in the frequency band of interest is not a fixed parameter. **It can be controlled by correct combination of clock frequency (i.e. update rate setting) and sequence period, so that the flat part of the power spectrum lies in the frequency range of interest.**

8. An *m-sequence* provides high noise immunity, thus its use not only allows one to make measurements in "quiet" enclosures (i.e. high signal-to-noise ratio) but in halls in the presence of an audience, hence **the problem of low signal to noise ratio present when a periodic pulse is used as a stimulus signal, is overcome.** The *m-sequence* method is highly immune to noise transients such as pops, foot steps, and other short impulses; all are transformed into benign noise, distributed evenly over the entire impulse response.

9. The advantage of the conventional techniques is that the Transfer Function of each device used in the measurement system can be eliminated from the measurement. This may be done by matching the characteristics of the devices used in the measurement system such as the anti-aliasing filter, transducers and amplifiers used in each channel. However, by using an *m-sequence* approach the method is even simpler; **by connecting together all the components (i.e. Devices) whose effects are desired to be eliminated and then taking a single measurement and storing it. Subsequent measurements can be equalized to remove those component effect from the measurement.**

In order to generate an *m-sequence*; first, it is necessary to design the corresponding shift registers' circuits each having a particular number of stages. A number of authors have generated tables of irreducible and primitive polynomials which yield *m-sequences*[99,100].

Characteristic polynomials in the order from 8 up to 16 have been selected for generating *m-sequence* signals with different lengths to be used by the system. The circuits of the corresponding shift registers then are configured with the proper feedback connections. The circuit designs and the stage status were then coded into a programmable generator. The following **Table 3.1** shows *m-sequence* with different lengths, order, feedback connections. The *m-sequence* programmable generator output has been validated by comparing the properties of the generated *m-sequence* signal to its well known properties. These properties have been extensively discussed in references[98,99]. The properties of the system-generated *m-sequence* signals comply well.

When an *m-sequence* approach is used, the minimum measurement time should be considered. In most cases, in order to avoid time aliasing, a minimum measurement must be ($2.L.\Delta t$) seconds, where, Δt is the clock pulse period (i.e the update interval in seconds). It is preferable to apply one full MLS period to the system to stabilize it and then apply another full *m-sequence* period to actually undertake the measurement. The *m-sequence* period should also be selected to be long enough to overcome time aliasing effect and to allow the impulse response

TABLE 3.1 Design information of several m-sequences[98-100,108] for a programmable generator.

M-Sequence Order	M-Sequence Length	Tap Positions**
8	255	2, 3, 4, 8
9	511	4, 9
10	1023	3, 10
11	2047	2, 11
12	4095	1, 6, 4, 12
13	8191	1, 3, 4, 13
14	16383	1, 6, 10, 14
15	32767	1, 15
16	65535	1, 3, 12, 16

** Feedback connections of shift register, others may be selected.

to decay sufficiently over the first of its L samples. The *m-sequence* length can be decided upon by estimating the reverberation time (*RT*) of the room under assessment and then selecting the sampling frequency required to cover the frequency bands of interest having a duration longer than the estimated *RT*.

3.4.2 Room Impulse Response Calculation

To obtain the room impulse response when an *m-sequence* is used as stimulus, it is sufficient to cross-correlate the room response with the same *m-sequence* used for excitation. Three techniques are available and employed in the measurement system to obtain the impulse response.

The first technique involves performing the radix-2 *FFT* on the *m-sequence* after interpolating the missing sample and on the room response; multiplying the two *FFTs* then performing an inverse *FFT* on the result, this inverse is the system impulse response[97]. This attempt to use radix-2 *FFT* and interpolating the missing sample or padding it with a zero since an *m-sequence* is always one sample less than power of 2, is less accurate because truncation effects will appear[109,110]. *The second technique* is the same as the first except no interpolation is done. This process is slower but gives more accurate results than the first. *The third technique* is the most accurate and speedy. This is done by using an algorithm proposed by Borish and Angell[102]. The room response samples (i.e. the acquired data sequence) are permuted (prescambled) in a special way and a zero is appended to the start of the resulting sequence to form a 2^n length sequence; the result is then transformed using a radix-2 Hadamard transformation and a final output permutation step yields the cross-correlation which in turn is equal to the impulse response. This algorithm has been coded as a call function and its results have been validated by comparison with the other two techniques.

3.4.3 Room Response Acquisition Module

The ***CBS-RAIMS*** data acquisition procedure is designed to be flexible and to accommodate different measurement situations. The data collection and the enclosure excitation are done simultaneously or delayed (pre- or post-triggering) as desired by presetting the necessary parameters for both processes. The number of the analog input channels can be set from 1 to 8 channels via the signal conditioning board, preserving their phase relationship. The number of samples or data points collected from each channel may vary up to 65536 samples or more based on available disk space. The collected data is then transferred directly to the computer hard disk via Direct Memory Access (*DMA*), using the approach of *first IN first OUT* (*FIFO*). A RAM Disk has been created for this data transfer process to accommodate acquisition with high sampling rates and overcome the problem of matching the "on disk" drive writing speed and data transfer speed. The total sampling rate is also variable up to 143,000 samples/sec, however if six channels are used a sufficient sampling rate per channel would be about 23,800 samples/sec; this is enough to recover octave frequency band data up to 8 kHz.

For room-acoustics evaluation purposes the frequency range of interest is set by the Standards[127] from 125 Hz to 4 kHz octave bands, but it is also desirable to include the 63 Hz octave band. Other variable setting features can be seen in Fig. 3.9 which shows the in-house designed interactive control panel for setting the analog input (acquisition) parameters, functions' descriptions are detailed in *Appendix A-I*. For example in a dual-channel mode, a sampling rate of 25,600 samples/sec per channel and 32767 samples/channel is sufficient for impulse responses each of 1.3 seconds. By halving the sampling rate to 12,800 samples/sec the sequence length would be 2.6 seconds. If 5 second impulses are desired, either the sampling rate can be reduced by a factor of 2 and the 4 kHz octave band dropped from the analysis or, alternatively employ an *m-sequence* of 65535 samples. Averaging is also possible in the time domain, however with the current system hardware no more than 6 signals can be averaged due to memory limitations; this enhances the signal-to-noise ratio (*S/N*) by approximately 7.7 dB.

3.4.4 "STIMULUS" SIGNALS OUTPUT MODULE

The CBS-RAIMS uses different *m-sequences* starting from 255 to 65535 samples and the analog output generation update rate (i.e. clock frequency, F_S, Hz) can be varied. By selecting a low clock frequency (i.e.sample update interval), one can concentrate signal power at low frequency or spread it over a wide band with a higher clock frequency. In general an L-sample sequence will excite the system at (L-1)/2 (excluding D.C.) discrete frequencies. The effective frequency band is governed by the equation[98],

$$f_{effective} = \frac{1}{L\Delta t} \quad to \quad \frac{1}{3\Delta t} \quad Hz \tag{1}$$

where, Δt is the clock pulse period (= $1/F_S$), sec

The frequency resolution is $1/(\Delta t.L)$ and the spectrum amplitude at zero frequency is approximately $[A^2\Delta t(L+1)]/2L$, where A is the amplitude of the sequence. Other setting parameters for *m-sequences* and other stimulus signals are shown in Fig. 3.10, while function descriptions are detailed in *Appendix A-II*. An example result of dual-channel room impulse responses is shown in Fig. 3.11.

3.4.5 Calibration Module

The measurement system calibration procedure is implemented, as a separate module, to obtain accurate and comparable values from the 6 impulse responses. One advantage of using the technique of pressure microphones is that easy and accurate calibration can be performed using a pistonphone (B&K Type 4220) or similar calibration device.

At the beginning of a measurement test with particular hardware settings, the pistonphone signal is applied sequentially to each microphone in turn and the

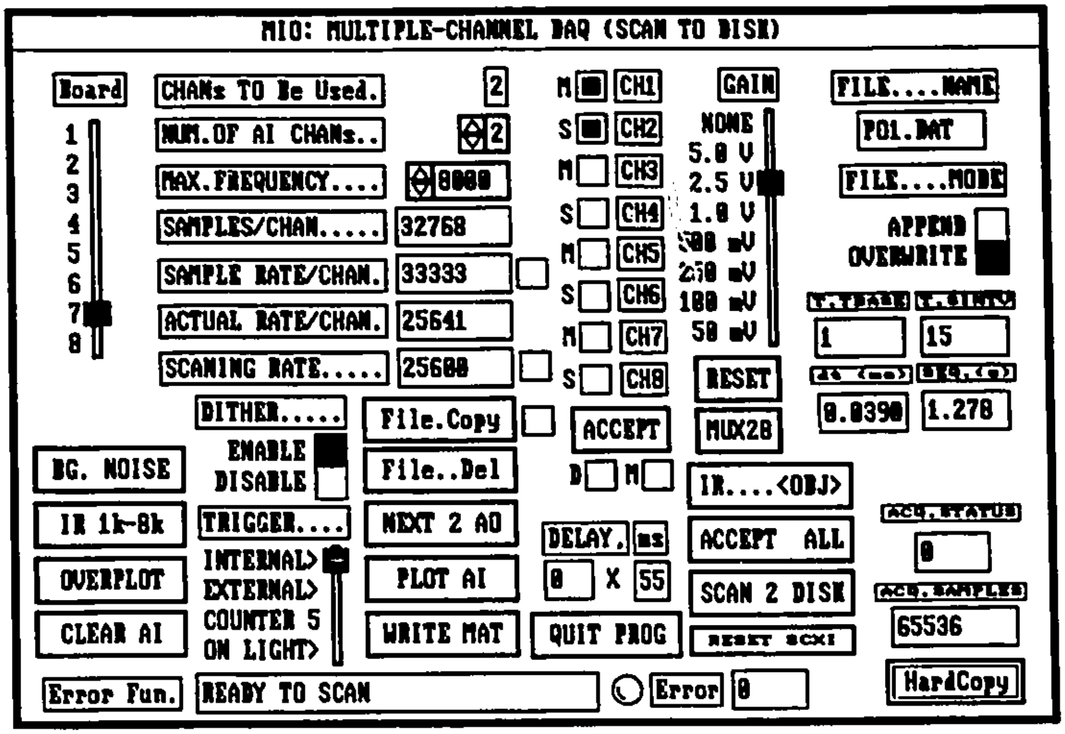


Fig. 3.9 The interactive control panel for setting the analog input (acquisition) parameters.

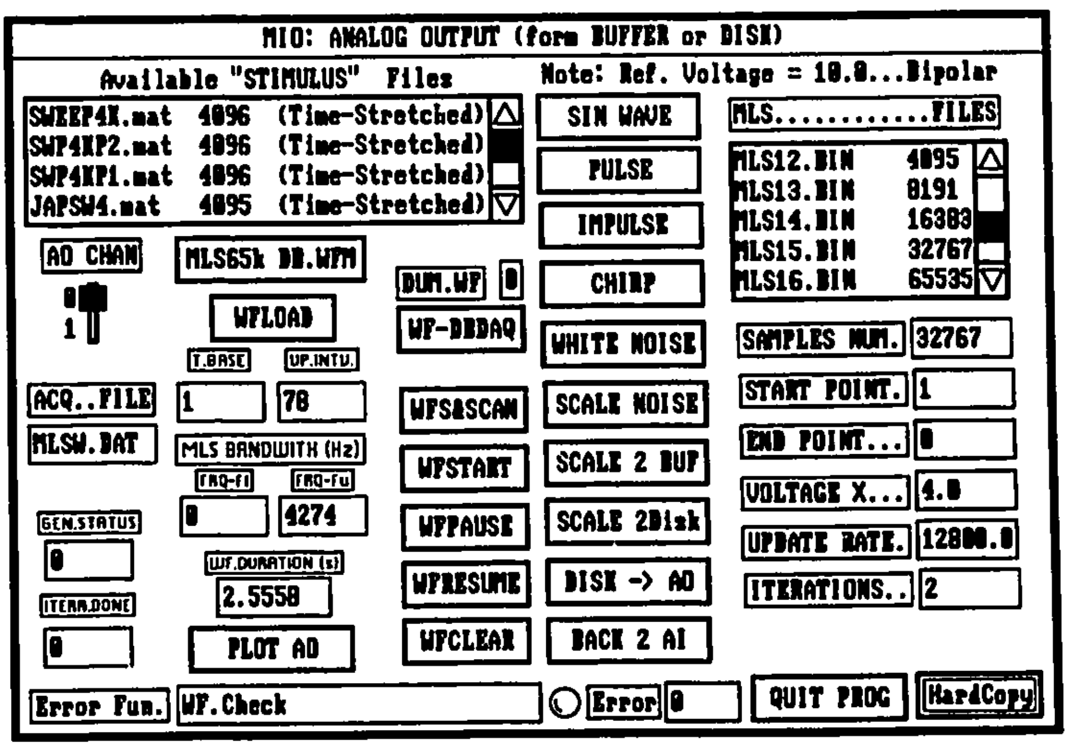


Fig. 3.10 The interactive control panel for "Stimulus" signals analog output settings.

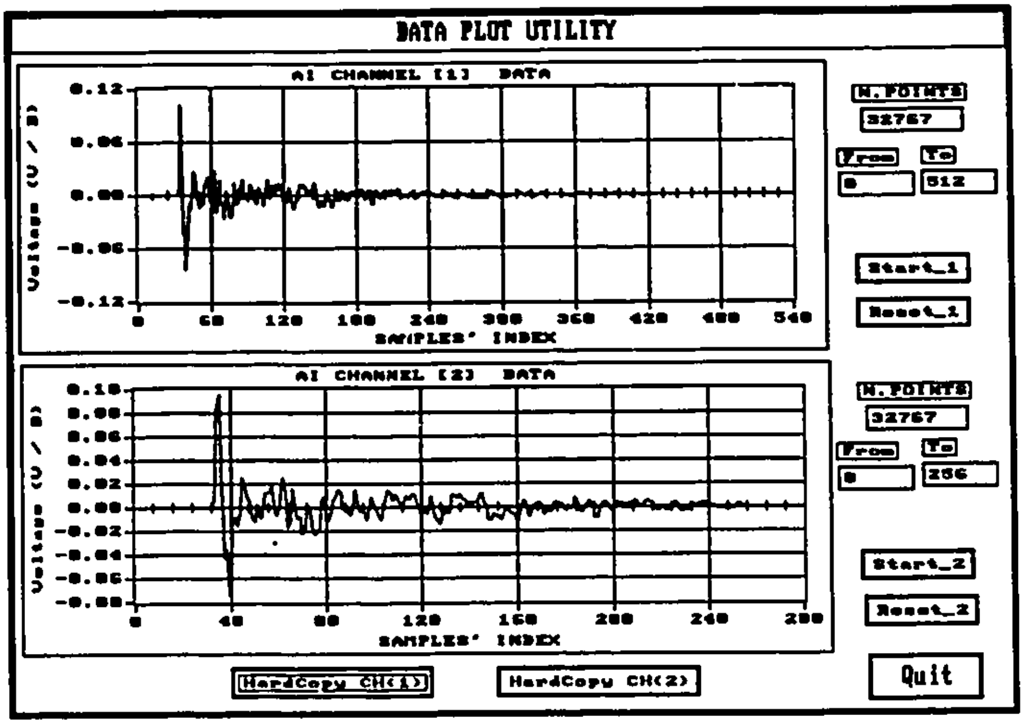


Fig. 3.11 An example output result of the system dual-channel room impulse responses.

resulting digital data is acquired; the signal level is then automatically calculated against the known level of the pistonphone signal and a correction factor for each of the microphones is stored in file for later amplitude correction of all data acquired with the same hardware settings. Another approach which is also implemented is to adjust the input gain of the used multiplexer channels to compensate for different transducers sensitivities. Fig. 3.12 shows the in house designed interactive panel for the calibration of data acquisition process via the microphones; panel functions are detailed in *Appendix A-III.*

The sound power output of the isotropic loudspeaker system also required calibration. This is done by way of standard sound power measurement in a reverberation chamber in accordance with ISO (3741,3742)[129] within the *CBS* reverberation chamber. This procedure is essential for the measurement of the room-acoustic indicator "Strength" or Relative Level, *G*. Other utilities such as measuring background noise (shown in Fig. 3.13) are also implemented.

3.5 SIGNAL FILTERING, PROCESSING AND ANALYSIS

3.5.1 Filtering Process Validation and Considerations

Measurement of room-acoustic indicators starts with the capture of a transient signal in the enclosure under investigation. Subsequently, the impulse response is then calculated using different techniques based on the type of sound source

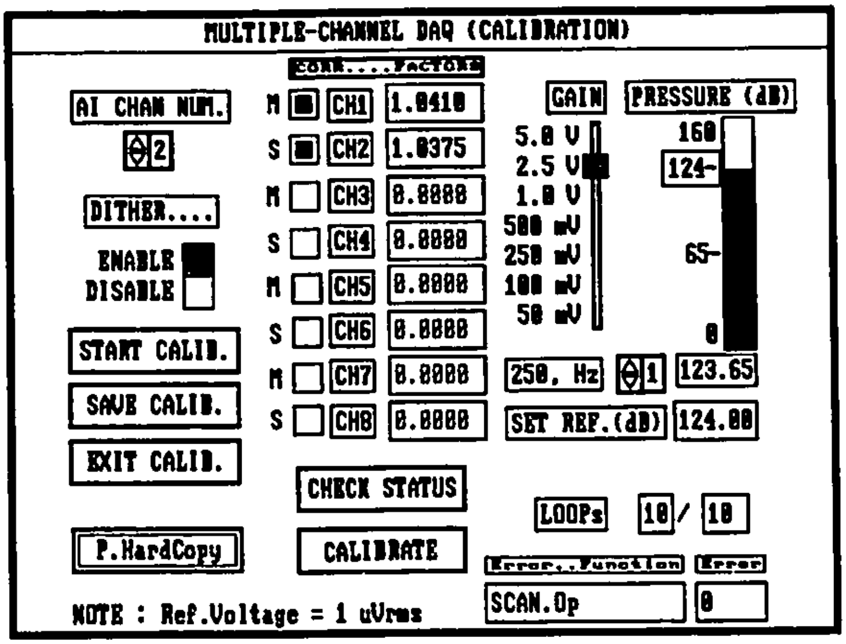


Fig. 3.12 The interactive control panel for the calibration process of the data acquisition via the microphones.

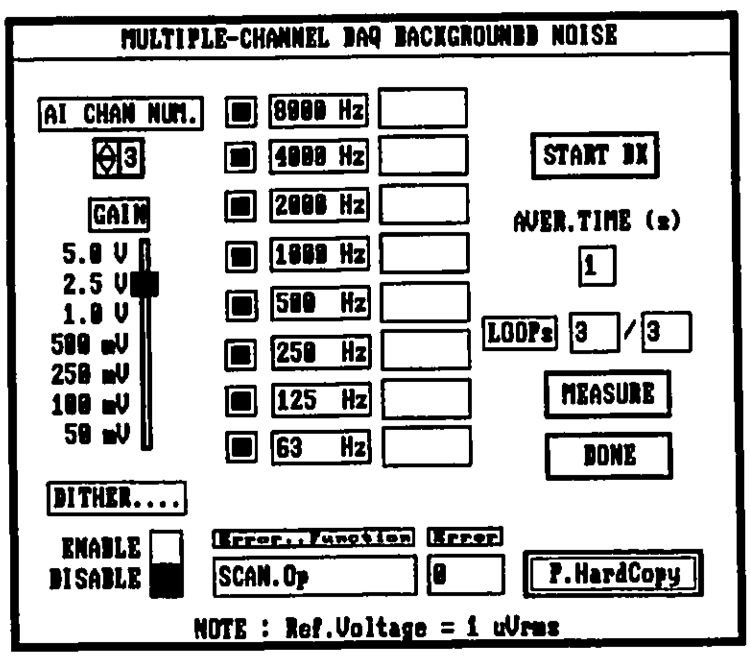


Fig. 3.13 The interactive control panel for measuring background noise.

and source signal. The impulse response includes almost all acoustic information describing the enclosure characteristics. To derive the subjectively relevant objective quantities from the impulse response, the first step is to filter the data into the standard one-octave or third-octave bands. This is achieved in the measurement system by software using digital filters.

The filter design used significantly affects the values of the acoustic indicators to be calculated, especially for those indicators which are based on early to late sound energy ratios. Also the accuracy required is based upon choice of filter design for a given measurement. Therefore, the resulting filter characteristics must compare well to applicable standards and output data validation should be thoroughly implemented.

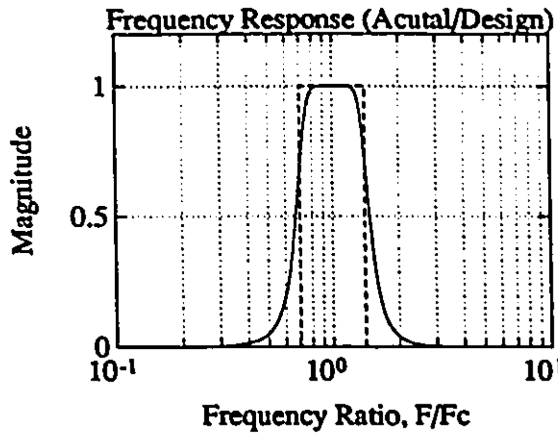
The possible approaches to data filtering, the one selected and its use are now discussed. In a relevant Standard (ISO/CD 3382, 1993)[127] no criterion is set for the filter type to be used but the filters must conform with IEC publication 255[130]. Other standards concerning filter characteristics and requirements also exist (i.e. ANSI S1.11. 1966, Reaffirmed 1975)[131].

Employing available software routines and built-in signal processing functions, it has been possible to realize one of the most commonly used digital filters, a multiple-pole "Butterworth" filter with 8 poles. One of the main advantages of realizing a digital filter is that by using the same software, it is possible to generate any filter shape by recalculating the filter coefficients . This characteristic allows the changing of centre frequency or filter bandwidth as desired. The filter coefficients completely determine the filter properties which in turn do not change with time nor are affected by the operational environment (i.e. external conditions) such as temperature, humidity or radiation fields.

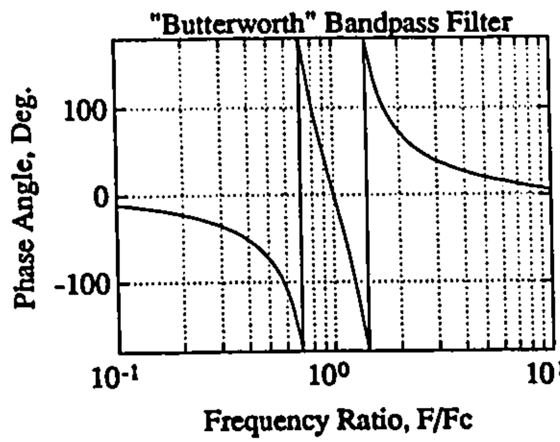
The realized Butterworth filter has a flat bandpass and approximately 48 dB/octave roll-off rate. The filter nominal centre frequency can be programmed to yield both one octave and third octave bands as desired with a range from 63 Hz to 8 kHz frequency in octave bands and from 50 Hz to 10 kHz in third octave bands. The input parameters to the filter type are simply the sampling frequency, F_S used for data acquisition and the set of unfiltered data sequence.

Standard requirements are limited to one-octave ,**"Class II"** (i.e Moderate) band and Third-octave band filters, **"Class III"** (i.e. High) must now be confirmed with respect to the utilized filter. Fig. 3.14 depicts the frequency response of the employed 8-pole "Butterworth" bandpass filter design compared to the standards. The filter can be seen to comply well with standards requirements

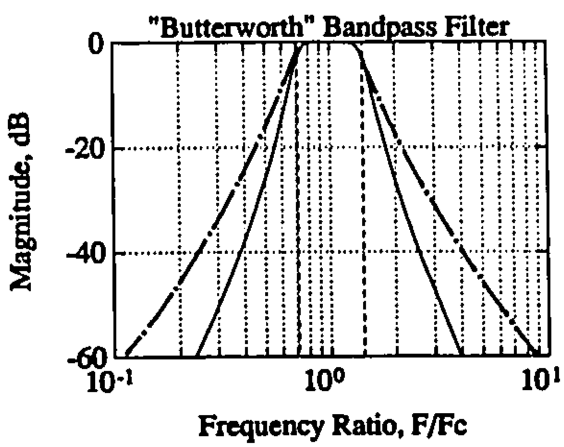
The second step is to check the validity of the output data obtained from the filter at all desired one octave and third octave bands. This is done by following two approaches. *The first approach*: Comparison with the results of available commercial programmable digital filters by applying the same input signal to both.



a) Frequency response.



b) Phase characteristics.



c) Design characteristics vs standards.

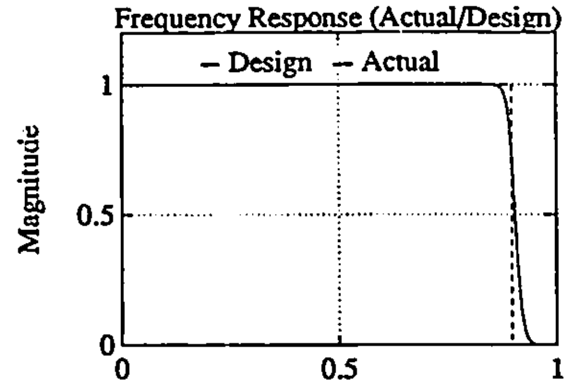
-- Design —— Actual -·- Standards

Fig. 3.14 The frequency response and phase characteristics of the system-employed 8-pole "Butterworth" bandpass filter design, a) Frequency response, b) Phase characteristics, and c) Design characteristics vs standards. [Note: F_c= Octave band nominal centre frequency, Hz].

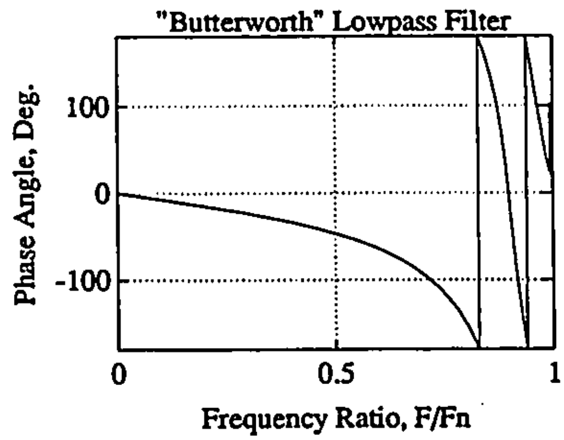
The data of a transient captured in a normal size living room by *MLSSA* Software (An available demonstration released, February, 1991) was acquired. The data sequence is 544.7 msec long and of 16383 samples acquired with a sampling frequency of 30,075 kHz. The *MLSSA* software consists of built-in functions for digital filtering using a similar 8-pole "Butterworth" filter. Its range is 50 Hz up to 45 kHz with programmable bandwidth and centre frequencies. The filtered data at each one-octave band from 63 Hz to 8 kHz were plotted and related data such as the maximum, minimum, mean, standard deviation and total energy components were also extracted. The same data sequence then was fed to the in-house designed digital filter. The output data at each one-octave band were thoroughly compared in all aspects and the results proved to be satisfactory. The sum of the energy components in each octave band compared well to the total energy of the unfiltered signal. *The second approach* included feeding in a pure tone which has a specific nominal centre frequency that lies in the range of the filter bandpass (e.g. 500 Hz). The output was a sinusoid of the same frequency and amplitude as the original; the signal was completely recovered back at its octave band with approximately 96% of its original energy content. The application of a sinusoidal signal also gives an indication of the response time of the filter in octave band; this was found to be in the order of 1.5 periods and since the relative bandwidth of the filter is 70.1%, one would expect the response to be 1/B, where B is the filter bandwidth (i.e. 1/0.701 = 1.4). Thus filter rise time is reasonable. In addition two sinusoidal signals of different octave band frequencies were added to form a periodic signal, the resultant then applied to the filter. The frequency components were recovered back each at both specific octave bands with almost all original energy content. To conclude it is evident that the realized digital filter is satisfactory.

To filter a sequence of data in the time domain after signal acquisition it is preferable, before transformation into digital form, to lowpass band filter (via the signal conditioning board) in order to avoid aliasing. In the present system this is done with a programmable 8-pole "Butterworth" digital filter with very steep flanks (approximately 90 dB attenuation). The filter cutoff frequency can be set as desired but it is commonly set to be in the range of 0.8 to 0.9 of the Nyquist frequency (= $F_S/2$, Hz). This step is essential to ensure that high frequency components are attenuated sufficiently to avoid aliasing. Fig 3.15 depicts the frequency response of the used 8-pole lowpass digital filter characteristics.

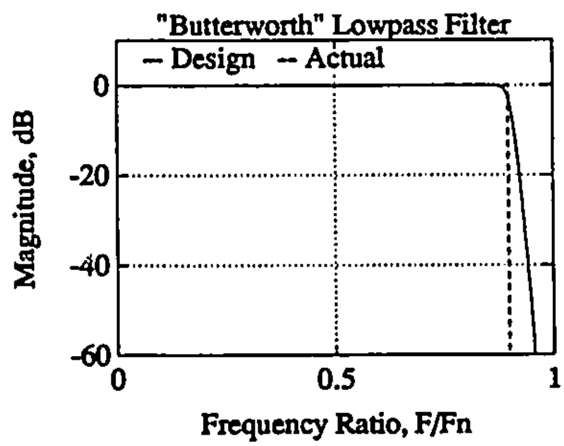
To filter a room impulse response two different approaches are available in the system for filtering the data into standard octave and third octave bands. The first approach is to pass each data sequence acquired from the A/D convertor through the 8-pole bandpass "Butterworth" filter seven times for each pass; the filter coefficients are automatically calculated based on the required band centre frequency and thus the filter response is changed so as to obtain successively the 8 one-octave frequency bands starting from 63 Hz to 8 kHz. The filtered data is then processed to yield known room-acoustic indicators. This approach is often slow but is still viable. The second approach is to pass the full length of the data



a) Frequency response.



b) Phase characteristics.



c) Design characteristics.

Fig. 3.15 The frequency response and phase characteristics of the system-employed 8-pole "Butterworth" lowpass filter design, a) Frequency response, b) Phase characteristics, and c) Design characteristics. [Note: F_n= Nyquist frequency, Hz].

sequence through the bandpass filter set at the highest octave band (i.e. 8 kHz); half the previous sampling frequency can now be created by discarding every second sample without losing information affecting recovery of the next lowest octave band and so on. The sequence is first lowpass filtered up to the highest frequency of the next lower octave and then sub-sampled taking every other data point; hence it is fed to the same filter with the same set of coefficients since halving the sampling rate will move the filter centre frequency down one octave. The process is repeated until the desired 8 octave bands are filtered. This process is known by the term "Decimation"[132-134]. This process reduces the volume of data to be processed each time but carries the penalty of reducing the sequence temporal resolution; this may affect the accuracy of energy integral calculations. The *CBS-RAIMS* uses a combination of the two approaches. Decimation is used to recover low frequencies when the Nyquist frequency (i.e. half the sampling frequency) is high compared to low frequency bandwidth. However, the decimated filtered sequence which is half the original data samples is interpolated to the original sequence length to preserve the temporal resolution for further processing.

Besides the anti-aliasing precaution described earlier, two other concerns should be addressed. These are "Filter Ringing" and "Phase Shift"[134]. When a signal is input to a filter the filter takes a finite time to respond i.e rings, this leads to signal delay. The delay is approximately the reciprocal of the filter bandwidth, (B) and depends on the characteristics of the filter. As discussed earlier the ringing length of the present design is reasonable, however this delay influences the perceived time of start at each octave band which in turn influences later calculations such as the Schroeder's decay curve or other early-to-late energy ratios.

A recursive filter will impose a phase angle on the filtered signal, that is the phase relation between the input and output signals is nonlinear. These phase changes or shifts can cause distortion in subsequent data processing and to remove its presence the data is filtered first in a forward direction, then the output is taken in the reverse direction through the same filter. The two phase shifts at any frequency will be removed and cancel each other. However this procedure precludes on line filtering and data processing, and analysis is delayed until the data acquisition cycle is complete.

3.5.2 Room Impulse Response Processing and Room-Acoustic Indicator Calculations

Processing of the filtered impulse responses involves first the calculations of the room-acoustic indicators. The Schroeder integrated impulse method is used to obtain the decay curves from which "Early Decay Time", *EDT* and "Reverberation Time", *RT* values can be calculated. *RT* values are calculated from the regression analysis (i.e. A linear least squares best fit) applied to the part of the decay restricted between -5 and -25 or -30 or -35 dB based on the available dynamic range. *EDT* values are obtained in the same manner but the decay is restricted to the first -10 dB.

To minimize the problem of the background noise effect and the decay length on the calculated values of the decay times as described by Chu[33] and Bradley[105], a scheme similar to that proposed by Chu is implemented, that is to subtract the background noise mean-square value from the original squared impulse response before performing the integration. Long after the response of the room has vanished, a temporal average value of the background noise can be obtained. This can be done graphically in an interactive way by examining the impulse response. The scheme completely eliminates any uncertainty in choosing the finite upper limit of integration. The finite upper limit of integration can also be chosen automatically by the program; a two-pass technique is used, where first the *EDT* is calculated based on the full impulse length resulting in an approximation of *RT* value which is then used to decide on an appropriate time record finite end. Two-thirds of the estimated *RT* value seemed to give optimum *EDT* and *RT* values with higher linear fit correlation coefficients.

Calculation of the various early-to-late energy ratios is done by integration applying either Simpson's Rule or the Trapezoidal Rule. Early-to-late energy ratios such as: "Definition", D_{50}, "Clarity", C_{50}, C_{80} and "Running Liveness", R are calculated. The "Centre Time", T_S and "Relative Level", G are also included. Indicators of speech intelligibility such as "Rapid Speech Transmission Index", *RASTI*, and weighted "Signal-to-Noise" ratio, SNR_{95} are also calculated using the procedures described in references[68,86,88]. All room-acoustic indicators are then output in comprehensive tables for ease of assessment and comparison. **Table 3.2a** shows as an example output of newer room-acoustic indicators versus frequency while **Table 3.2b** shows speech intelligibility assessment via *RASTI* calculations[85,86] and rating for a measurement taken in a concert hall.

Several sub-routines have been developed to achieve further signal analysis such as spectral analysis using fast Fourier transformation, power spectral density calculation, cross spectrum, complex transfer function and cepstrum analysis. The mathematical formulae and their applications in signal analysis are provided in reference[134]. The system also incorporates routines to calculate the sound intensity from two data sequences employing the Finite Difference Approximation Approach[135,136]. The results include pressure, instantaneous particle velocity, instantaneous intensity and average intensity. Envelope intensity[137] is also employed. Fig. 3.16 shows a data-processing flow diagram for the purpose of obtaining both objective room-acoustic indicators and the directional characteristics of the sound field; the directional characteristics information shown in Fig. 3.16 will be discussed in more detail in Chapter 4.

TABLE 3.2 An example output of a measurement undertaken in a small concert hall. (Position values)

TABLE 3.2-a Room-acoustic indicators versus frequency.

Space Average
STD
Position Values

			Room-Acoustic Indicators								
		FREQ., Hz	EDT Sec.	RT Sec.	D_{50} Ratio	R dB	C_{80} dB	T_s Sec.	G dB	LEF_s Ratio	SNR_{sa} dB
	1	63	0.622	0.622	0.70	-3.7	8.5	0.040	8.0	0.03	
	2	125	0.708	1.239	0.73	-4.3	7.9	0.048	8.5	0.05	
	3	250	1.103	1.030	0.26	4.5	0.2	0.096	8.8	0.12	
LOW-FREQ.	●	Mean	0.811	0.964	0.56	-1.2	5.6	0.061	8.4	0.10	
	4	500	1.285	1.095	0.44	1.0	0.9	0.081	10.5	0.15	0.6
	5	1000	1.027	1.092	0.50	0.0	2.8	0.072	10.2	0.15	1.60
MID-FREQ.	●	Mean	1.156	1.094	0.47	0.5	1.85	0.077	10.4	0.15	1.1
	6	2000	1.077	1.085	0.58	-1.4	3.2	0.059	9.2	0.10	2.4
	7	4000	1.083	1.063	0.54	-0.7	2.9	0.063	8.8	0.13	
	8	8000	0.860	0.994	0.56	-1.0	6.8	0.038	8.1	0.12	
HIGH-FREQ.	●	Mean	1.007	1.047	0.61	2.1	4.3	0.053	8.7	0.12	
				Value		%AL		Rating			%I.S.
	●	STI		0.57		7.8		FAIR			95
	●	RASTI		0.55		8.7		FAIR			

Tonal Colour	1.14	Bass Ratio	0.78	Treble Ratio	0.93

Note: % I. S. = Intelligible Syllables

TABLE 3.2-b RASTI calculations and rating.

Position Values

			Modulation Transfer Function Matrix					
	Modulation Frequency / FREQ., Hz	125	250	500	1000	2000	4000	8000
1	0.71					0.944		
2	1.00			0.862				
3	1.41					0.823		
4	2.00			0.649				
5	2.80					0.589		
6	4.00			0.392				
7	5.00					0.342		
8	8.00			0.209				
9	11.20					0.179		
			Value		%AL		Rating	
●	RASTI		0.55		8.7		FAIR	

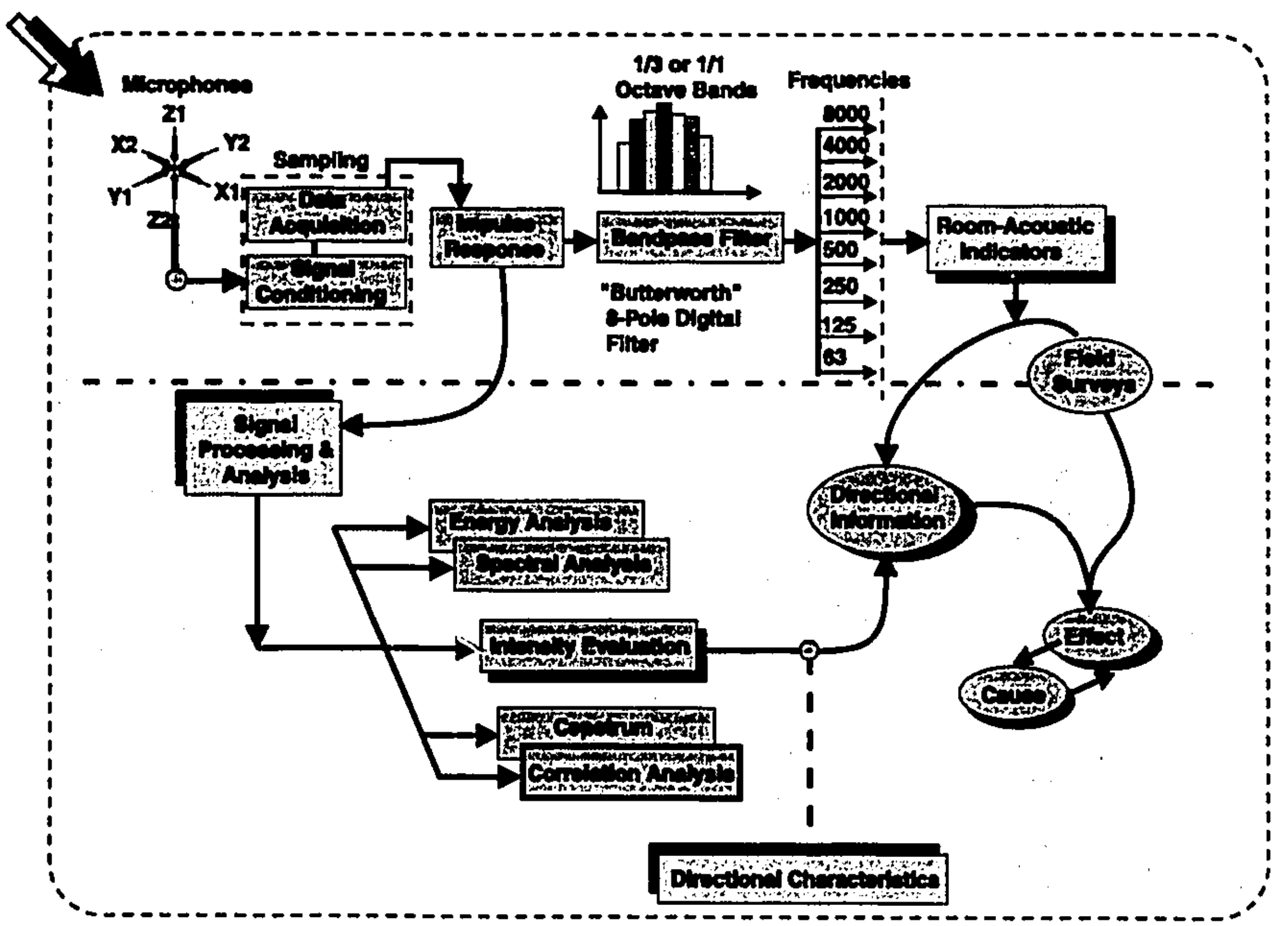


Fig. 3.16 A data-processing flow diagram for calculating room-acoustic indicators and directional information of the sound field.

3.5.3 The System Operation and Results Validation

To validate the *CBS-RAIMS* operation and results two approaches are employed. The first is to examine the precision of the numerical calculations and procedures implemented in *CBS-RAIMS*. For this purpose an impulse response, captured by the measurement system *RAMSoft-II* in the "***LOYOLA***" concert hall at Concordia university is used. *RAMSoft-II* has been compared with other measurement systems[124,125] and has been extensively used for evaluating existing halls[19-21,61]. The corresponding acoustical indicators results are also provided by Dr. J. Bradley[138]. The impulse response has then been applied to the data processing and analysis module of the *CBS-RAIMS*. The output results are shown in Fig. 3.17(a-e). As can be seen from the graphs both systems' results are almost identical, the very minor differences found can be attributed to possible calculations details and filter characteristic differences. Fig. 3.17(e) compares the speech intelligibility indicator SNR_{95} calculated from the impulse response by *CBS-RAIMS* to a similar indicator i.e. U_{80} values; although differences are expected due to different mathematical expression, both indicate the same trend with values in close agreement. It can be said that the performance of the processing and analysis module of *CBS-RAIMS* compares well with *RAMSoft-II*.

The second approach is to validate a complete *CBS-RAIMS* operation and results are now of concern. Since results of measurements undertaken by *RAMSoft-II* in "***LOYOLA***" concert hall were available, the opportunity was taken to perform

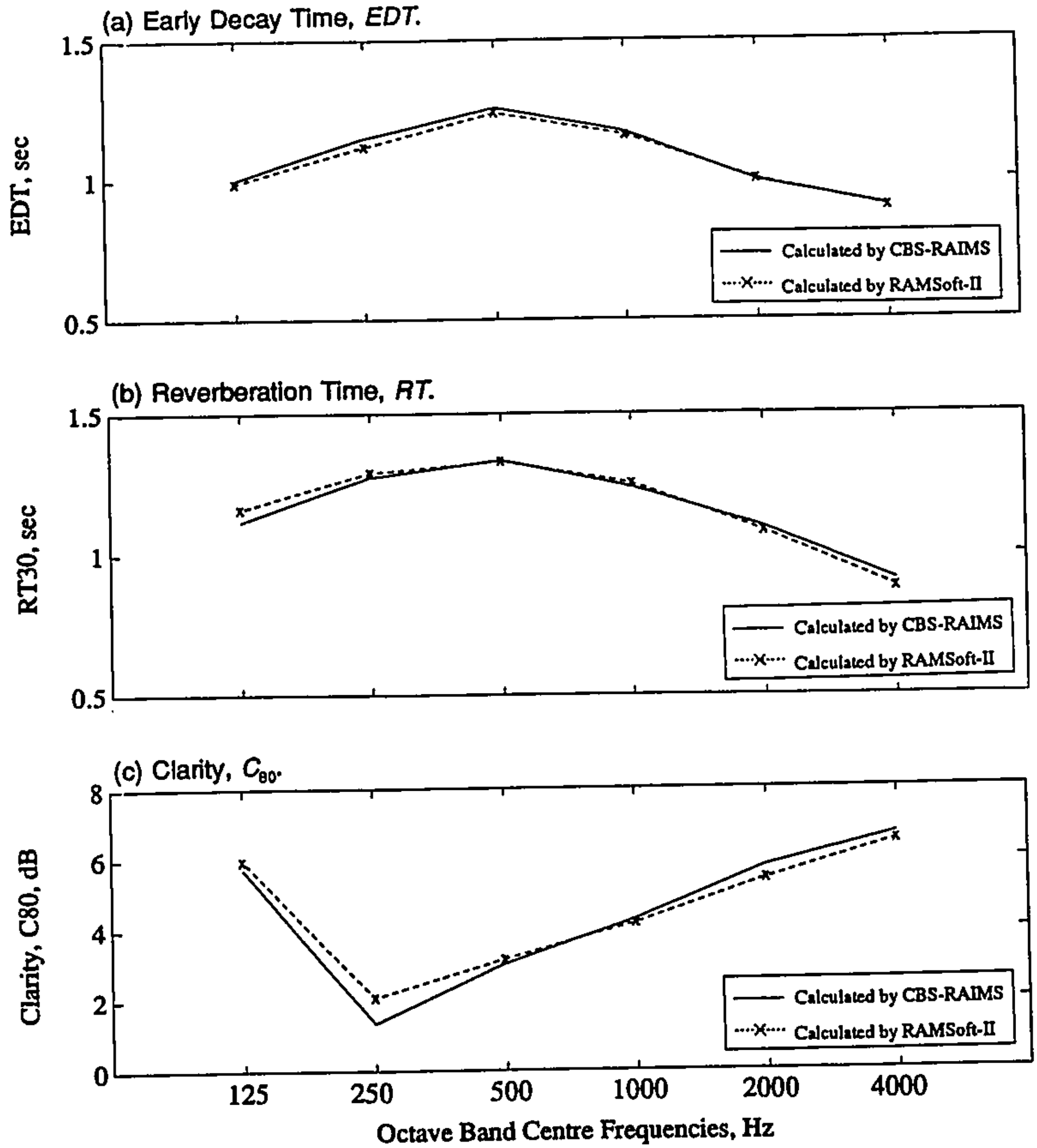


Fig. 3.17 Comparison of room-acoustic indicators calculated by both *RAMSoft-II* and the *CBS-RAIMS* data processing module when the same impulse response is applied.

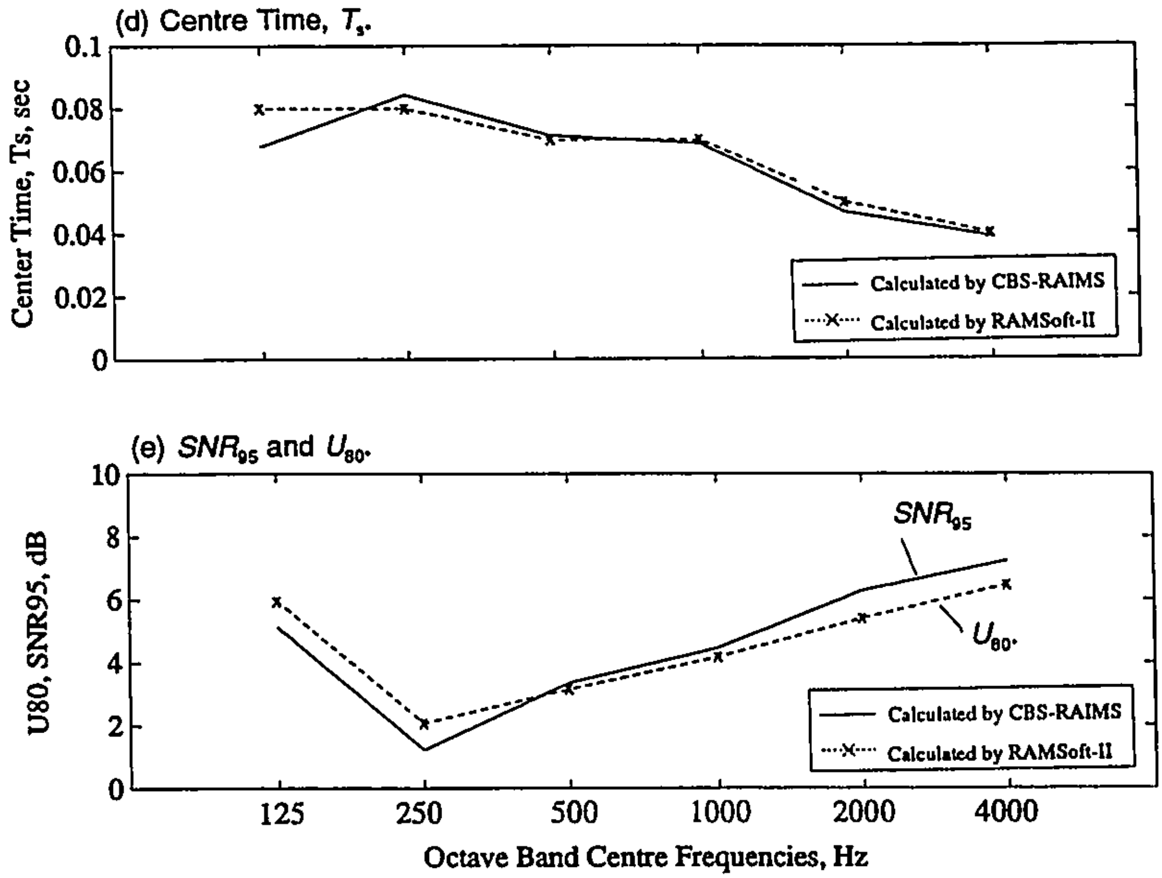


Fig. 3.17 Continued

similar measurements. The purpose of this step is to compare the results of both measurement systems with respect to room-acoustic indicators and if necessary explain the differences. Measurements were conducted at different times in the same listeners' seat locations but not necessarily the "exact" positions, therefore comparing results on a position by position basis might, in this case, be misleading. Bradley and Halliwell[105] and Pelorson *et al.*[25] showed that small variations of the source or the microphone positions displacement (e.g. ±10 cm) leads to measurable variations for all acoustical indicators. Therefore this comparison approach was disregarded. A general position-averaged (spatial average) comparison is favoured to indicate how close the results are and to detect discrepancies or abnormal differences. Comparison here refers only to six principal acoustic indicators; these are *EDT*, *RT*, C_{80}, T_S, *G* and *LEF* in the frequency range from 125 Hz to 4 kHz. Fig. 3.18(a-f) shows a comparison of the hall-averaged results of *CBS-RAIMS* and *RAMSoft-II* systems. Although differences are found, results are close and express similar trends. *RT* differences can be attributed to the different available dynamic range of evaluation; i.e. RT_{30} dB in *RAMSoft-II* while it is RT_{20} dB in *CBS-RAIMS* due to a lack of dynamic range. Differences in C_{80} at mid-frequencies are in the order of 1 dB while it is less than 25 ms for T_S indicator. *G* values also vary in the range of 1 to 1.5 dB. Fig. 3.18(f) compares the spatial average of *LEF* values obtained by a figure of eight microphone used by *RAMSoft-II* and those obtained by a pair of sound intensity microphones employed by *CBS-RAIMS*; differences are in the order of 0.05.

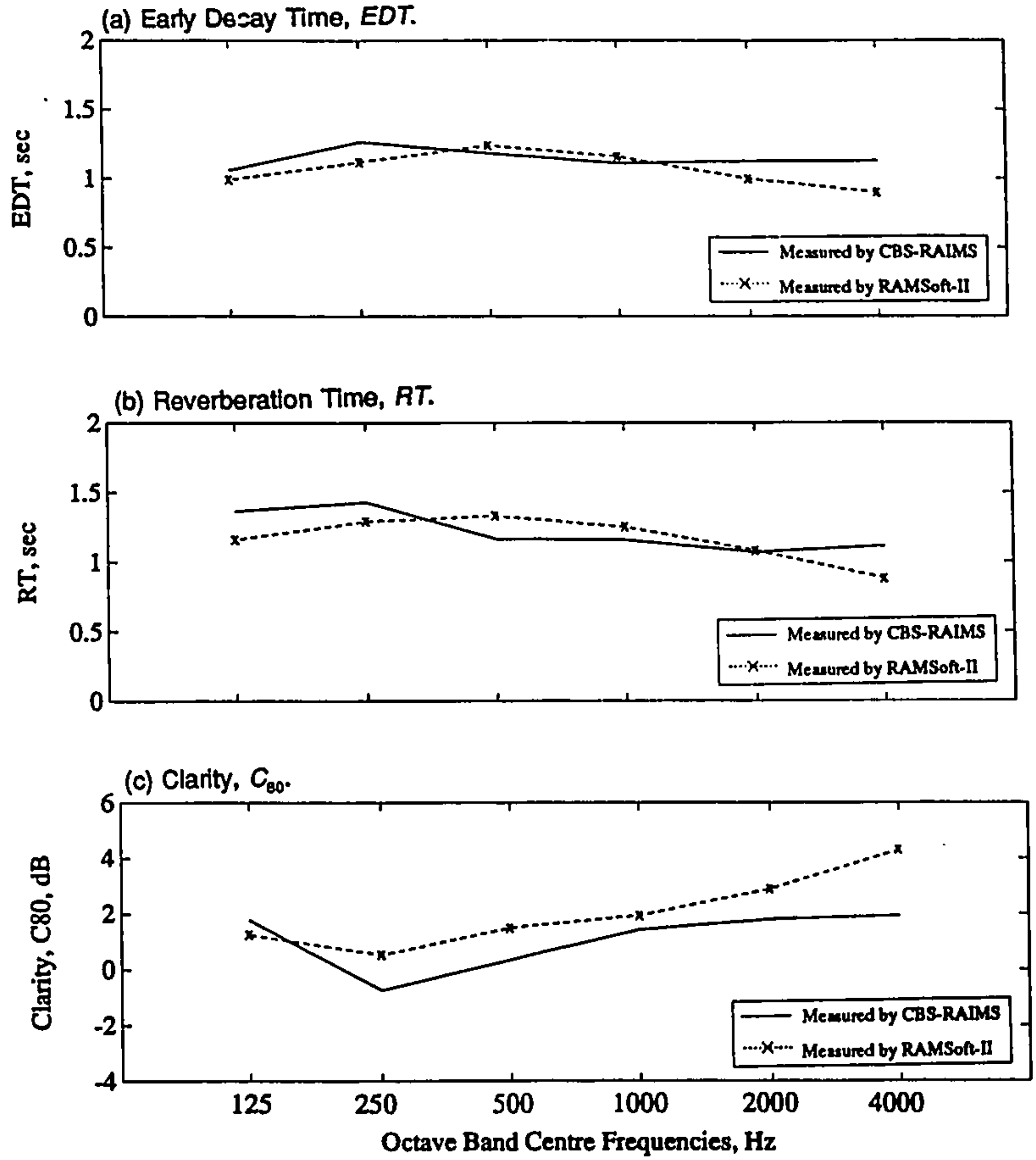


Fig. 3.18 Comparison of spatial average of room-acoustic indicators obtained by both *RAMSoft-II* and the *CBS-RAIMS* evaluated for the "LOYOLA" concert hall.

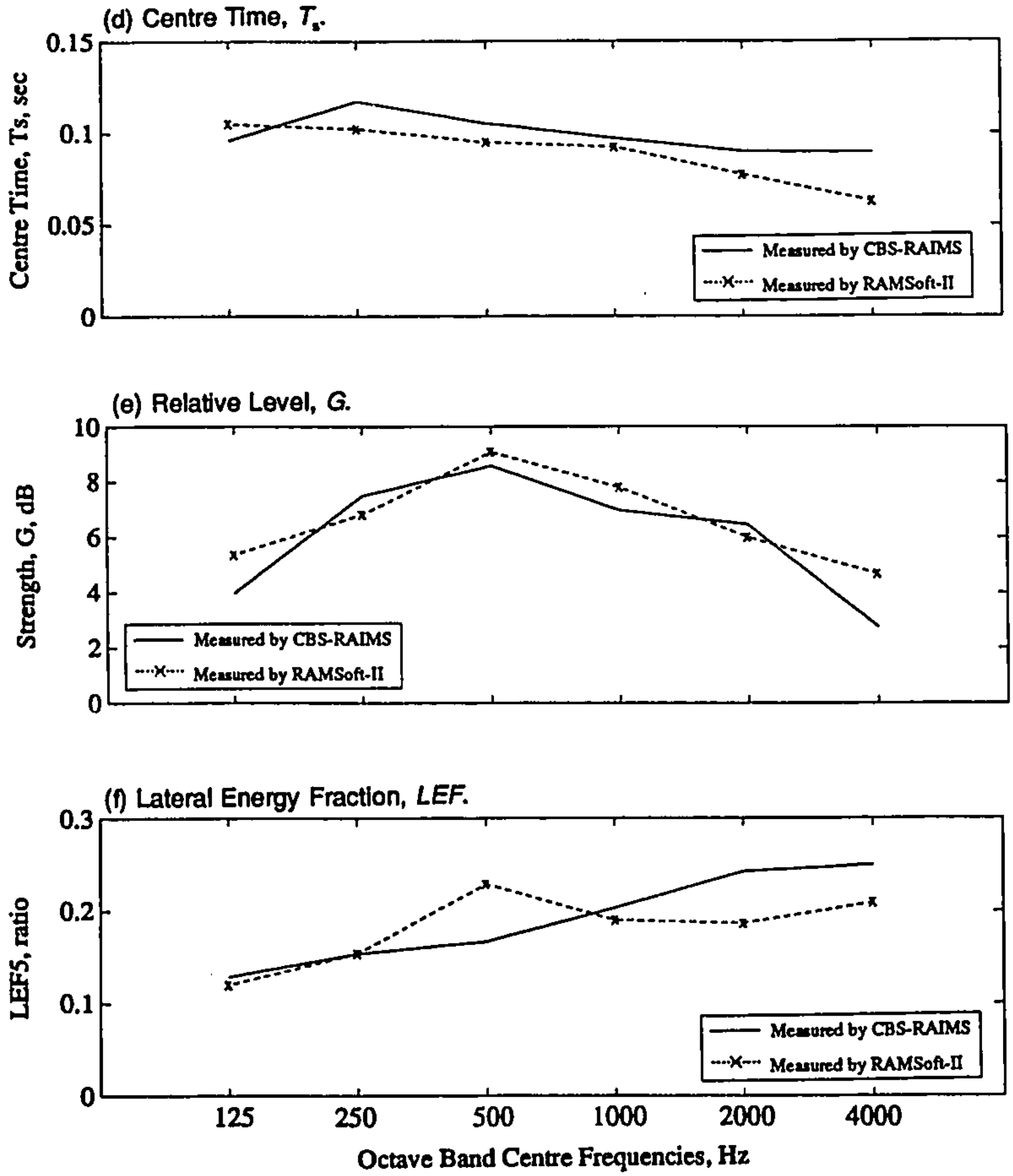


Fig. 3.18 Continued

These differences are expected as a realistic result of systematic differences in the physical characteristics pertaining to the loudspeaker system, microphone influences, data-acquisition accuracy and procedure details adopted by both systems. Similar measurable systematic differences have been reported by other authors[25,124,125]. In general it can be concluded that the *CBS-RAIMS* performance and results with respect to room-acoustic evaluation are within acceptable bounds.

CHAPTER 4

THREE-DIMENSIONAL TRANSIENT SOUND INTENSITY MEASUREMENTS : A NEW METHOD FOR OBTAINING SPATIAL INFORMATION

4.1 INTRODUCTION

Traditional sound-field measurements are usually made employing one omni- or gradient-microphone. By measuring a single impulse response at a listener position, most of the objective acoustical indicators can be derived. Recordings made with a dummy head also allow the reproduction of the sound field for subjective evaluation. However a detailed acoustic analysis of the sound field in an enclosure such as the spatial structure of the reflections arriving at a listener location can not be obtained or analyzed by these measurement methods. An acoustical measurement method that enables visualizing and analyzing the incident sound field in three dimensions is needed. This requires both multiple microphones and a processing method to sense sound vectors incoming at an observation point with differing magnitudes and incident angles within a short time period, typically the reverberation time.

Sound intensity is a vector quantity which indicates not only the magnitude but also the direction of the energy transport in the sound field. Intensity measurements have been applied to many problems in areas such as sound power measurements, noise source localization and diagnostics, transmission loss measurements, diffraction quantification, sound fields inside cars and near field investigations around sources, vibrating surfaces or structures. These different applications of intensity measurements and analysis methods have been developed throughout the eighties and nineties and many are described in reference[139]. The advantage of the vector nature of acoustic intensity to describe sound fields were highlighted in reference[140]. From these applications, vector intensity measurements have proven to be a valuable tool in describing complex sound fields. However, the technique is seldom if at all utilized in transient measurements[141,142] such as objective room-acoustic evaluations although it is potentially more informative compared to the usual sound pressure level measurements, particularly for obtaining acoustic spatial information. For example, the theoretical basis for determining the intensity and IACC of the early reflections in a control room was attempted[143]. Intensity measurements are usually not considered in room-acoustics due to the lack of an efficient and easy to perform measurement method. Other reasons could be the difficulties inherent or the associated instrumentation requirements or simply the time lag between areas of acoustic endeavour.

4.2 DIRECTIONAL DISTRIBUTION OF SOUND : STATE OF THE ART

In recent years considerable interest has been shown in the directional characteristics of sound fields in rooms particularly as they relate to the perception of sound quality. In a room, the total sound field is composed of the direct sound and many subsequent reflections from surface boundaries. Each reflection can be characterized by three attributes, its direction, relative strength to the direct sound, and arrival time. On the assumption that the room response is linear, the room impulse response provides the strength and temporal structure of the received sound at a particular listener position. The room impulse response is typically then analyzed in three regimes; the direct sound and early reflections, a transitional period, then late subjective reverberation which is usually characterized by randomness.

Studies[44,49,72] have shown that it is subjectively quite different to receive the first distinguished lateral reflections in the early reflection part than within the uniformly random weak reflections arriving at a later time, that is spatial and temporal distribution are important parameters of subjective quality. However, whether the reflected sounds arrive uniformly from all directions or from only one direction is unquestionably significant in binaural perception, and the importance and desirability of obtaining the directional sound distribution in rooms had been emphasized by a number of workers, Barron and Marshall[45], Schroeder[52] and Bradley[105].

Sound directionality in rooms was first studied by Meyer and Thiele as reported in reference[3]. Directional distribution was assessed using a steady state warble tone as the excitation signal while simultaneously acquiring the sound from a rotating microphone placed at the focus of a parabolic reflector. This technique has limited use since transient response is not captured. Broadhurst[118] studied sound directivity using a group of microphones arranged in a solid-grid form in space but, in addition to hardware complexity, the technique and data processing is time consuming. Recently, Endoh *et al.*[119] have developed a technique by which spatial information especially that of the early reflection-sequences (i.e. first 100 msec) is measured. A measurement technique developed by Yamasaki and Itow[120] employs a four channel microphone array to determine virtual image source positions and directivity patterns; powers of virtual image sources are then calculated by a correlation technique. The technique depends upon the similarities of incident waveforms, therefore separation of waveforms in the late reverbertion time where sound waves are mixed or deformed is difficult. Sekiguchi *et al.*[121] have developed a sound field measurement method by applying Yamasaki and Itow's technique to four microphones arranged close to each other at the apex of a regular tetrahedron but utilizing a deconvolution method instead of correlation to improve time resolution and shorten the calculation time. The objective is to obtain virtual sound source positions via geometrical reflection, but listening experience at an auditor position is affected by the resulting sound field from all incident waveforms and relating a sound reflection to a particular source is of less concern. This measuring approach however has been recently adopted in a commercial measurement system analyzer developed by Matsumi Takeucki, Matsushita Communication Industrial Co., Ltd., Skokichiro Hino, Etani Electronics Co., Ltd., and their Associates, Tokoyo, Japan, under the name "*FOURMIC* "[122]. The

"*FOURMIC*" system has clearly been under development during the same time, as the system reported here and there appear to be certain similarities. There are however, concept (i.e. to obtain subjectively relevant sound directional charateristics) and operating differences between systems, but absence of reported "*FOURMIC*" details and applications precludes detailed comparison. We can observe in passing that the "*FOURMIC*" is costed at $ 140,000, while the system reported here can be constructed from proprietary components both in hardware and software for $ 25,000 or less.

In any event whilst these later techniques give new insight and valuable information about the directional characteristics of sound in enclosures, no systematic attempt has yet been made to establish relationships between known room-acoustic indicators and the directional characteristics of the sound field as will be presented here.

4.3 THREE-DIMENSIONAL TRANSIENT SOUND INTENSITY METHOD

It is required to yield a visually detailed image of arriving sound intensity vectors at the listener location on a time base. The filtered sets of impulse response X-X, Y-Y and Z-Z in each selected octave or third octave band allow three orthogonal intensity vectors components to be calculated in the time domain using a finite sound pressure difference approximation given by the equation[135]:

$$I_n(t) = \left(\frac{1}{2\rho_o d}\right) [p_1(t) + p_2(t)] \int_{-\infty}^{t} [p_1(\tau) - p_2(\tau)]\, d\tau \qquad (4.1)$$

where,

p_1 = sound pressure of channel 1, Pa.
p_2 = sound pressure of channel 2, Pa.
ρ_o = air density, kg/m^3
d = spacing between the microphone pair, m

Because transient energy formulation is based on the evolution of equation (4.1) the full transient record length for each set is used to avoid erroneous results from segmentation and time windowing procedures[135]. The resulting instantaneous intensity vectors are then used to obtain specular sound reflection directions; however if one is only interested in sound energy direction then the envelope intensity technique can be used. The envelope intensity has a property between the instantaneous intensity and the time averaged intensity and is evaluated by the expression[137]

$$\vec{I}(t)_r = RE\,[\tilde{P}(t)\,.\,\tilde{V}(t)^*] \qquad (4.2)$$

where,

$\tilde{P}(t) = p(t) + j\,\mathrm{H}[p(t)]$ = analytic pressure, Pa
$\tilde{V}(t) = v(t) + j\,\mathrm{H}[v(t)]$ = analytic velocity, m/s
RE = Real part of complex function
$*$ = Complex conjugate
H = Hilbert transformation

The result is smooth sound intensity components which may correspond better to subjective hearing impression due to the ear inertia and the masking of weak reflections by strong ones (inferred from reference[3]). Fig. 4.1 displays a comparison of instantaneous and envelope intensities obtained in a reverberant field.

The 3-D intensity vectors versus time are then calculated by applying a conversion from rectangular to spherical coordinates and are displayed in the present work by employing "AutoCad"* software. A data-processing block diagram for obtaining and visualizing sound field spatial information is shown in Fig. 4.2.

Fig. 4.3 illustrates the vector orthogonal components of sound intensity with respect to the median, lateral and horizontal plane at a listener position. A typical result is presented in Fig. 4.4; it shows the full time and directional representation of sound intensity vectors for a measurement conducted in the laboratory. In practice the graphical output of vectors is directionally colour coded for ease of interpretation; visualizing the temporal arrival, direction and magnitude, particularly of early sound reflections, will allow further detailed study of their relative relationships. For example the full directivity patterns at the same location can be displayed with time of arrival, or viewed from different angles with respect to the listener as shown in Fig 4.5 or processed to reveal particular directional components in isolation, say left and right reflections in the horizontal plane as shown in Fig. 4.6.

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* AutoCad software, by AutoDisk, USA.

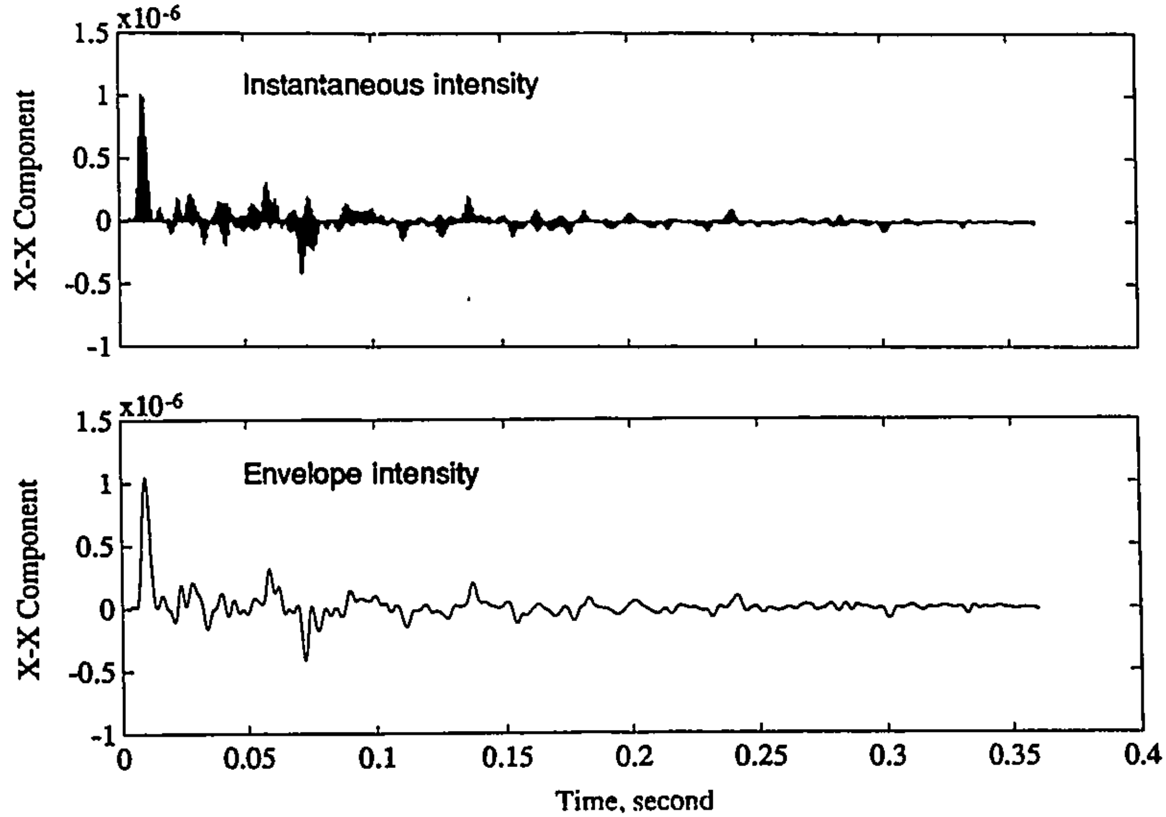


Fig. 4.1 A typical example of an orthogonal sound intensity component at 500 Hz in a reverberant field.

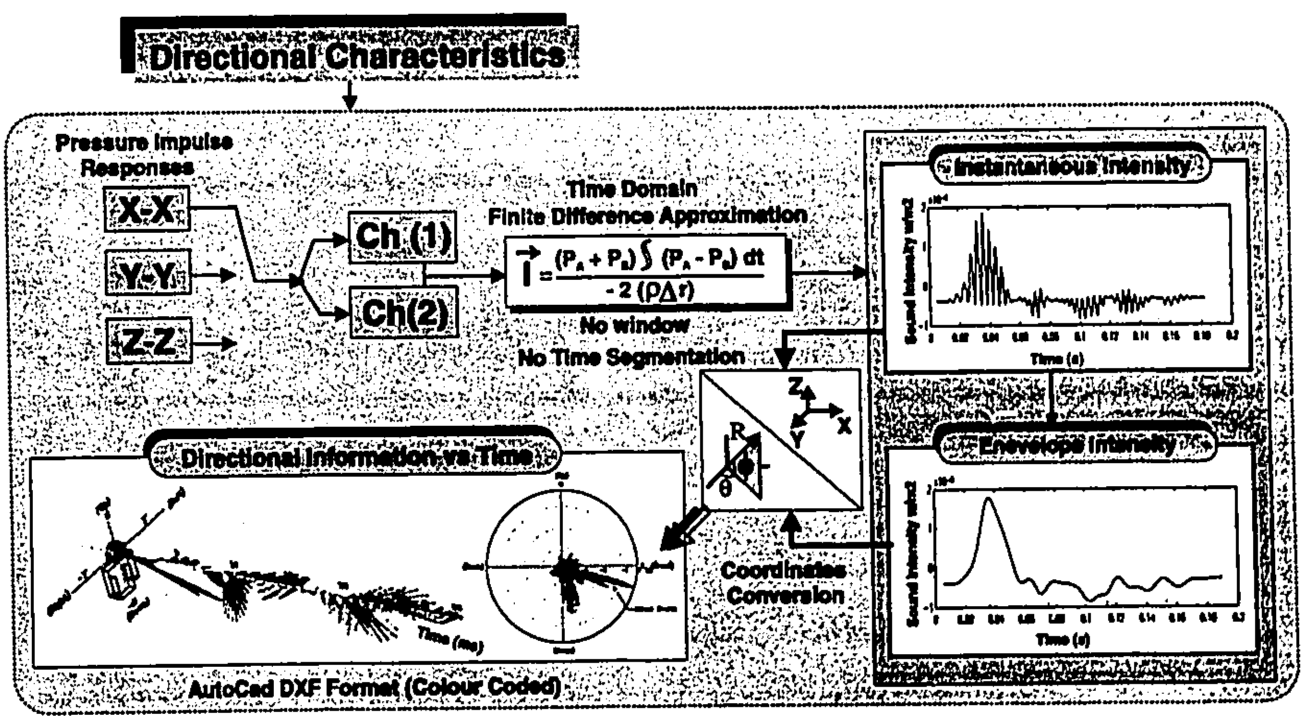


Fig. 4.2 A data-processing block diagram for obtaining and visualizing the spatial information of the sound field.

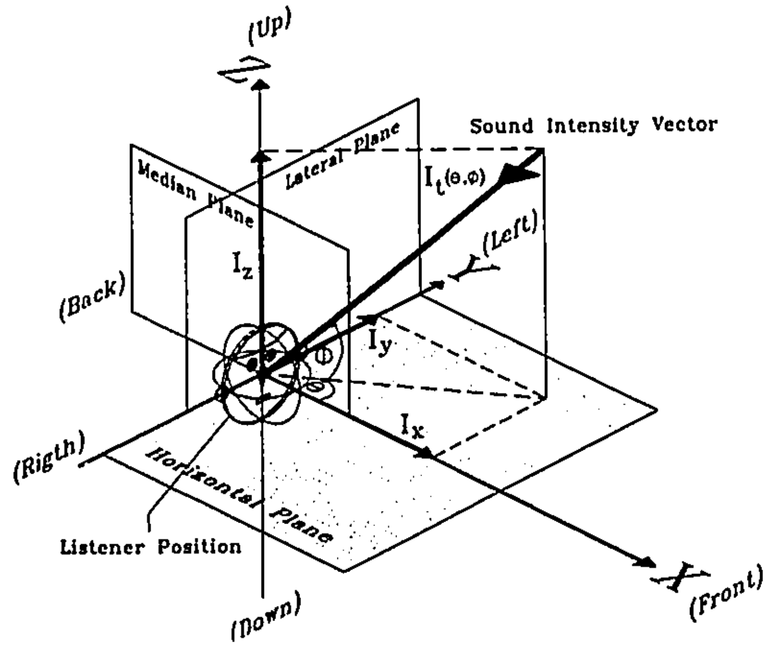


Fig. 4.3 Components of 3-D sound intensity vector with respect to the median, lateral and horizontal planes at a listener position.

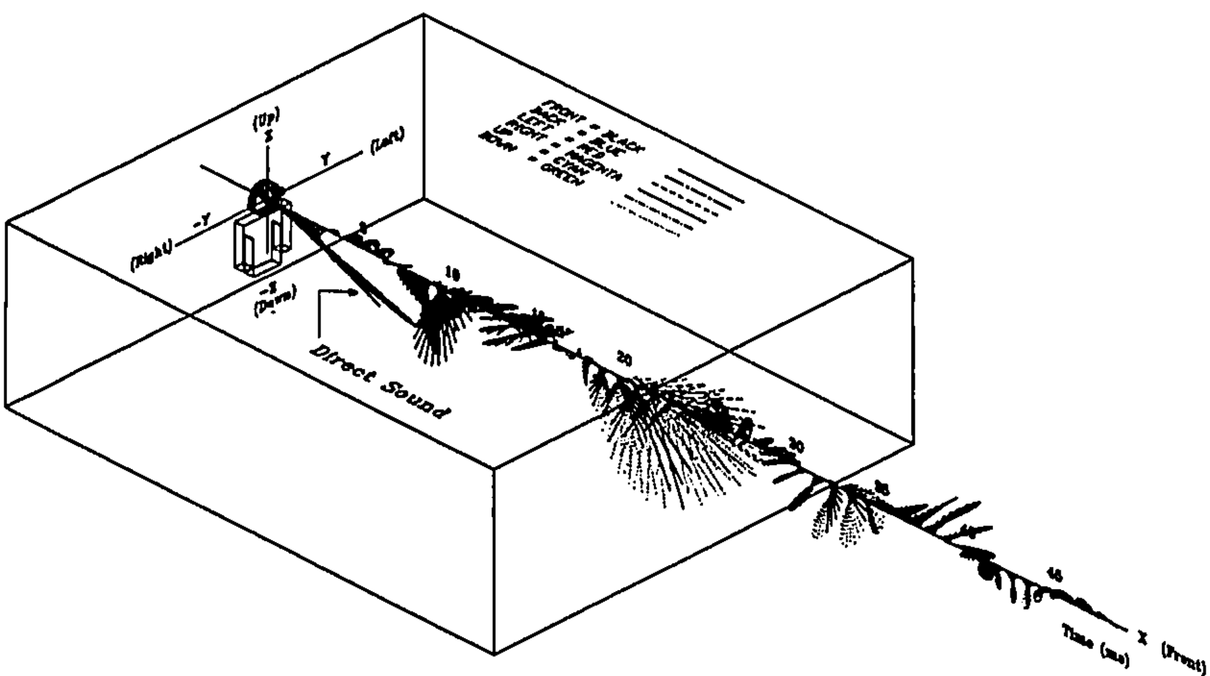


Fig. 4.4 Example result of sound field directional information for a measurement in a reverberant field at 500 Hz, versus time, 3-D illustration. [Note: Sound intensity vectors are normalized with respect to the maximum direct sound intensity]

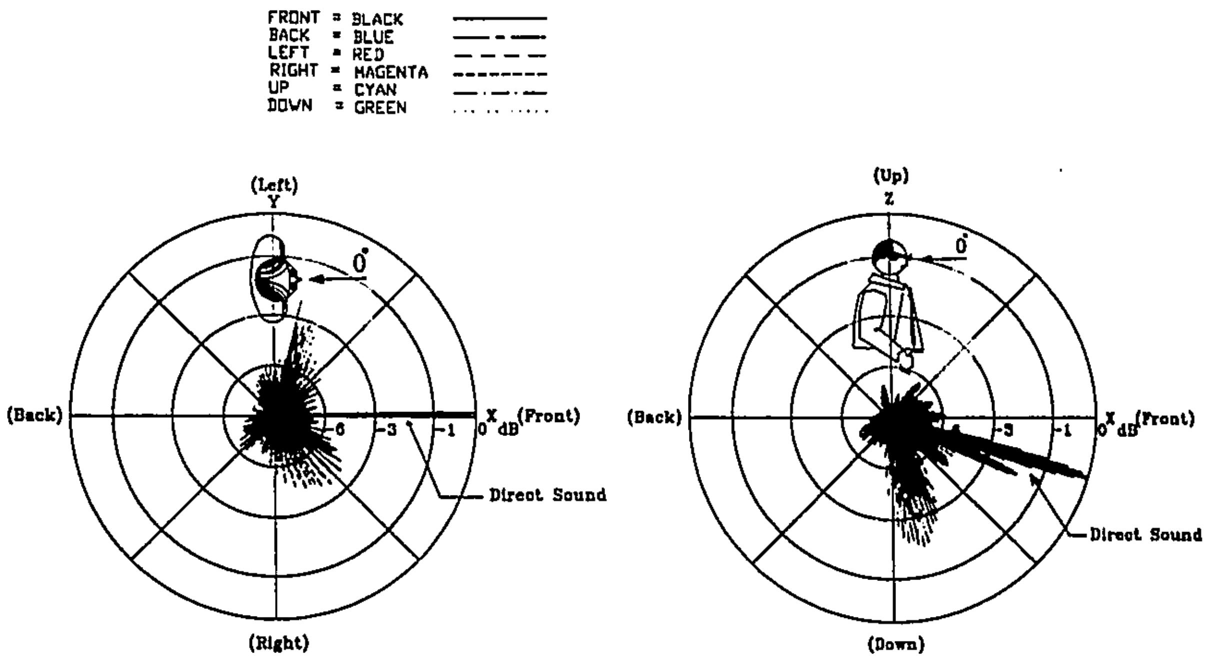


Fig. 4.5 Sound field directivity patterns viewed in different planes with respect to the listener for the signal of Figure 4.4. (Time interval : 0-80 msec)

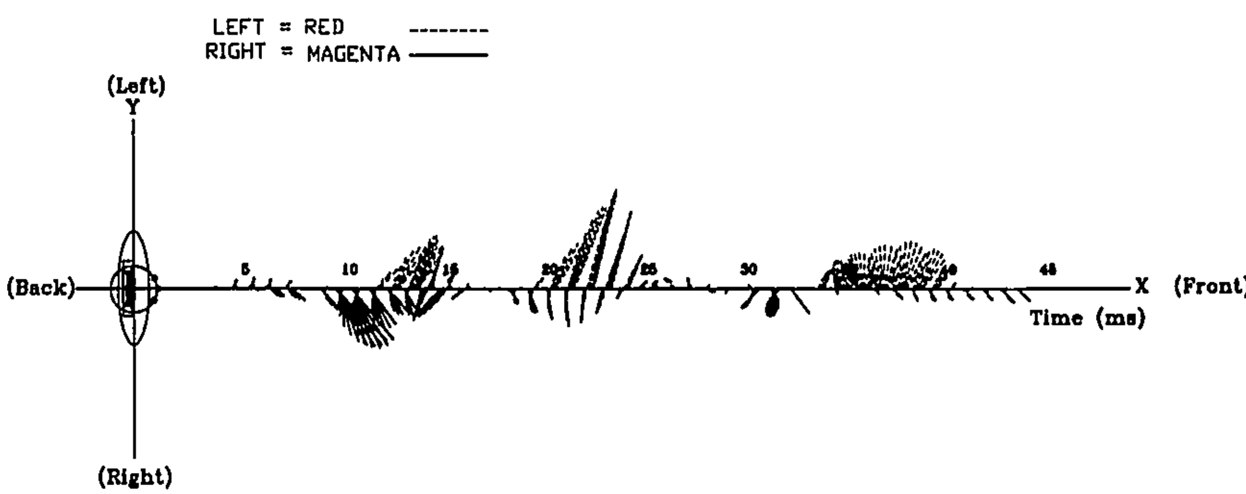


Fig. 4.6 Directional and temporal structure of left and right reflections in the horizontal plane for the signal of Figure 4.4.

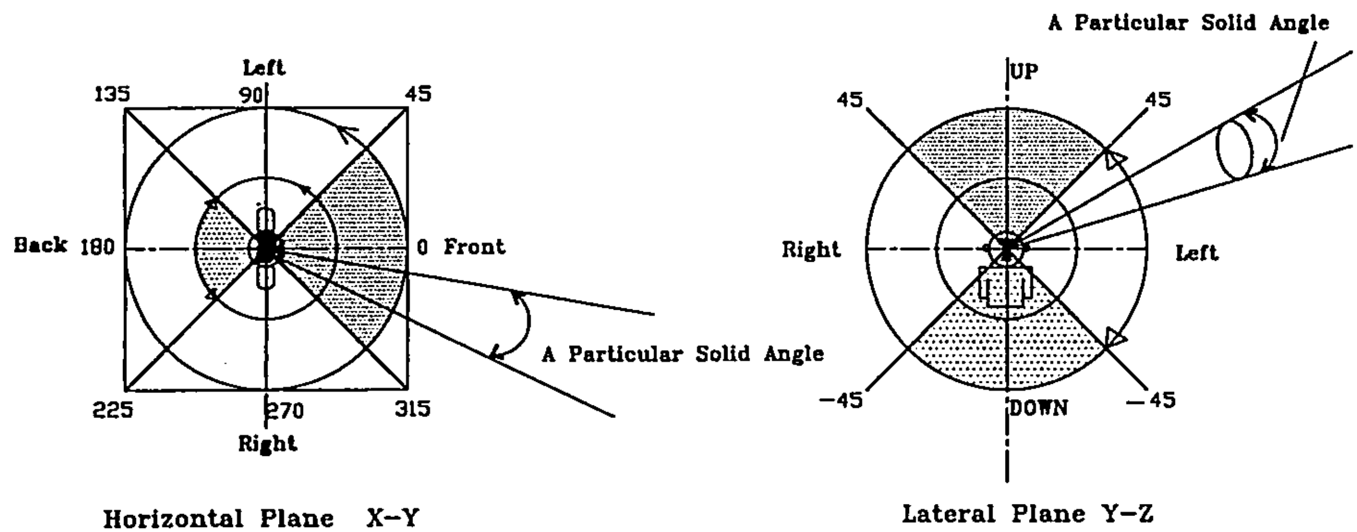


Fig. 4.7 Azimuth and elevation limits for the principal directions; Front, Back, Right, Left, Up and Down with respect to a listener in both the horizontal and lateral planes.

The directional information is identified in six principal directions with respect to the listener; these are front, back, right, left, up and down as shown in Fig. 4.7; thus the contribution of received sound energy in each can be separately examined. Further, since listener preference has been positively correlated with the presence of binaurally dissimilar early lateral reflections[52], it is useful to determine whether the sound reflections contain significant un-correlated lateral components; this can be readily examined by cross-correlating right and left arriving early sound reflections.

Time and directional information together with related room-acoustic indicators will also allow one to attribute cause to effect with respect to the influence of spatial design details such as proscenium, cantilevered or recessed balconies, and facia as in the case of a concert hall or vaults, arches and pillars in a church, as they may effect the arriving early reflections sequence at listeners' locations in their vicinity. However, it must be accepted that a net energy flow in a given direction will be indicated and that some circumstances arise which might cause erroneous directional interpretation, for example two angular symmetric vectors of equal amplitude occurring at the same instant of time will be resolved to a single resultant along the axis of symmetry. Indications of these occurrence can be given by comparing the temporal displays of pressure and unsigned intensity, that is, by examining an instantaneous pressure-intensity index.

4.4 VALIDATION OF THE MEASUREMENT METHOD

Known reflective surface locations and sound source position have been employed to test the efficacy of the measurement method to detect directional components. The method is found particularly useful for identifying the direction of specular reflections in the early time period. In a small semi-anechoic chamber a reflective panel (0.6 x 0.6 m) was placed over an absorptive floor surface (absorbtion coefficient > 0.95 for frequencies > 100 Hz), i.e case 1, as illustrated in Fig. 4.8 shows the test room configuration and the experimental setup. A short broad band m-sequence with the order of 12 and 13 were then used as the excitation signal fed to an isotropic loudspeaker located 1 meter from the floor. The measurements were taken using a 1/2" microphone probe mounted on the three-axis probe holder shown in Fig. 3.2 in three successive cartesian orientations before and after the placement of the reflective panel. The captured impulse responses filtered at mid-frequencies (500 and 1000 Hz) were then processed to yield measured 3-D intensity vectors. Fig. 4.9 shows the reflections in 3-D and 2-D illustrations. As can be seen from the different graphical views sound reflections from the floor direction where the panel was placed are evident shortly after the direct sound. The direct sound was also isolated to identify the source location and as shown in Fig. 4.9(c) it correctly indicated the location of the isotropic loudspeaker. The experiment was repeated with the reflective panel suspended from the absorptive ceiling of the test room (i.e. case 2 : Fig. 4.8), and then the panel was also fixed on a side wall location (i.e. case 3 : Fig. 4.8). Examining the 3-D intensity vectors versus time at

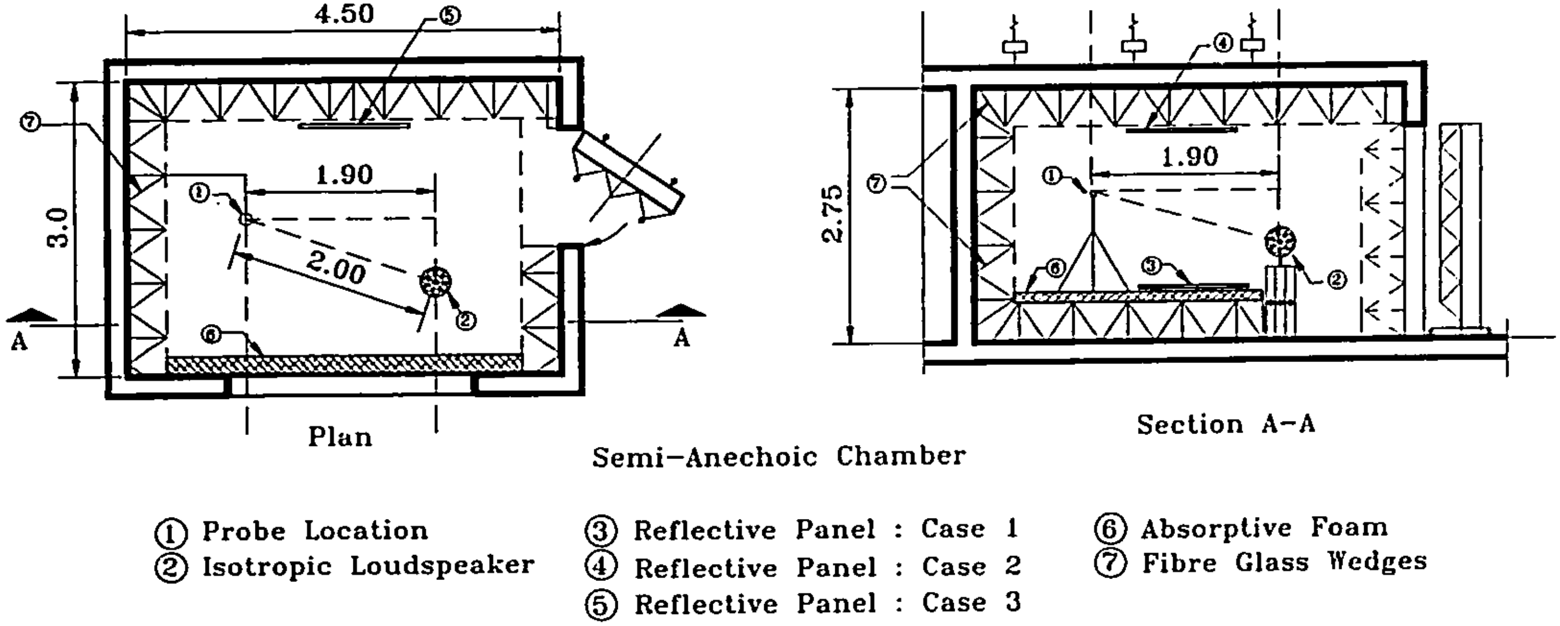


Fig. 4.8 The test room configuration and experimental setup for validating the measurement method to detect directional components.

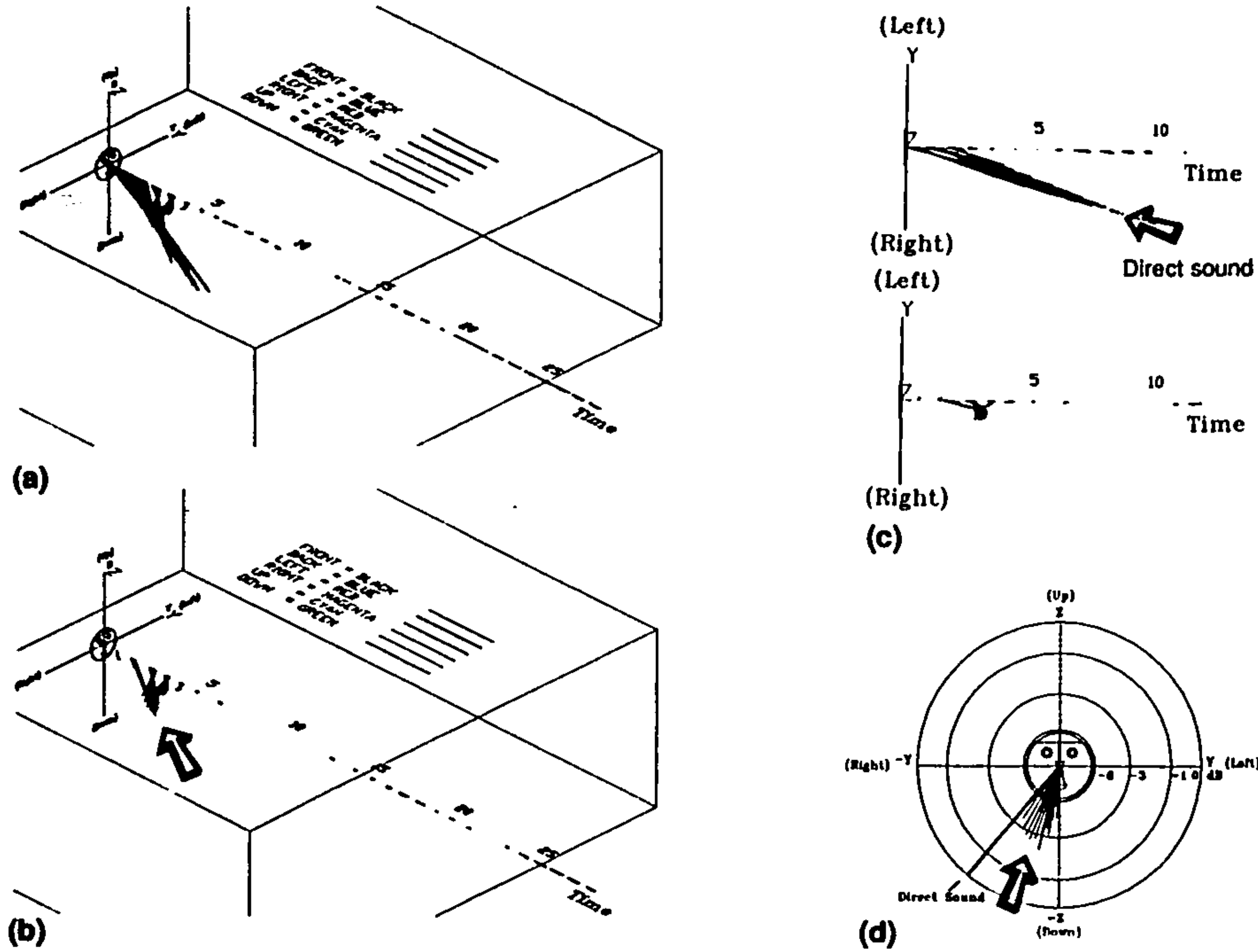


Fig. 4.9 3-D illustration of temporal and directional information of direct sound and floor reflections at 500 Hz, (a) 3-D illustration of received intensity vectors after reflective panel placement, (b) 3-D illustration of temporal and directional structure of reflections from floor in isolation, (c) Direct sound and reflections from floor in the horizontal plane vs time, and (d) directivity pattern of floor reflections in the lateral plane.

the relevant frequencies in each case showed a bundle of reflections with different magnitudes indicating the direction of the panel. In all cases, the intensity level has been normalized to the maximal direct sound for comparison between cases.

4.5 PRECISION AND LIMITATIONS OF THE MEASUREMENT METHOD

Microphone sensitivity, the phase mismatching between the measuring channels, and the directional characteristics of the probe itself will influence the accuracy of direction sensing. In 3-D transient sound intensity measurements these errors are more difficult to accommodate because the resulting intensity vector will include a combination of such errors encountered in the successive orthogonal measurements.

The channel phase mismatch causes a definable distortion of the probe directional characteristics. The direction for minimum sensitivity is deflected by an angle ψ from the plane perpendicular to the probe as given by[136] :

$$\psi = \arcsin\left(\frac{\varphi}{k\Delta r}\right) \tag{4.3}$$

where,

φ = microphone phase mismatch, radians

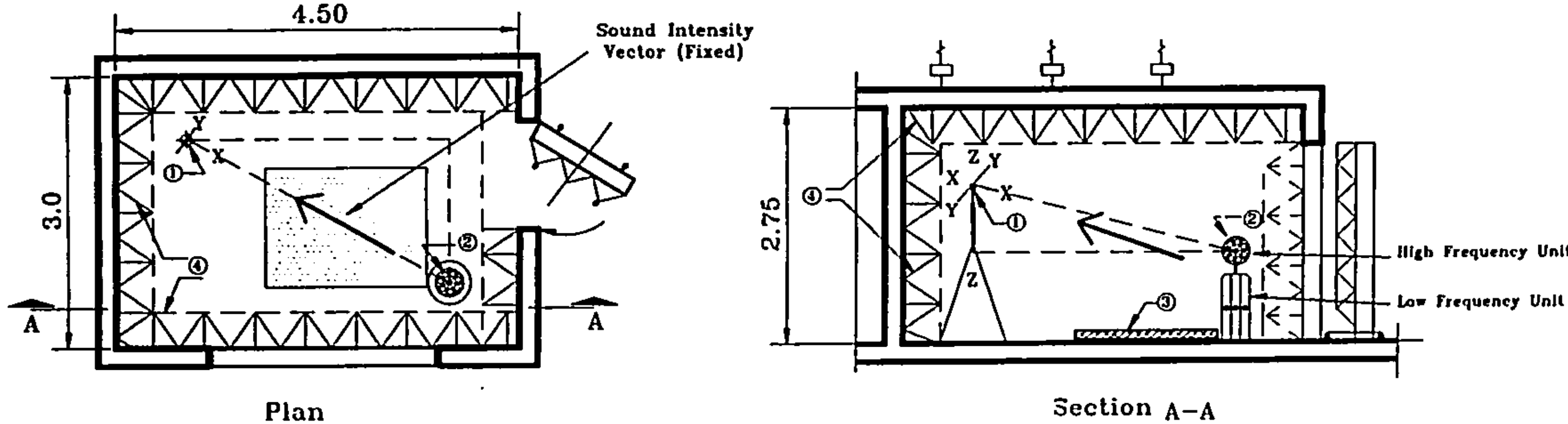
k = wave number (i.e. $2\pi f/c$, c = sound speed, m/s)

Δr = microphone spacing, m

For example, a 0.3° phase mismatch results in an angular direction error of approximately 2.6° or 11° at 125 Hz for microphones' spacing of 50 and 12 mm respectively. The directional reactivity[139] of each of the three orthogonal measuring directions relative to the direction of the true intensity vector is also of concern. In the present application, the true intensity vector incidence angles relative to the orientation of the probe are random and unknown and it is useful to identify the mean absolute value of the direction sensing errors together with their lower and upper bounds over several probe orientations versus frequency.

A measurement was conducted in a small semi-anechoic chamber with a fixed source-receiver position; the true intensity vector is fixed but the orientation of the probe in relation to it was varied. Fig. 4.10(a) depicts the experimental measurements' setup while Fig. 4.10(b) shows the different orientations and rotational direction of the probe coordinates relative to the true intensity vector. The purpose is to investigate the accuracy of measurement method directional detection with respect to the previously mentioned error source.

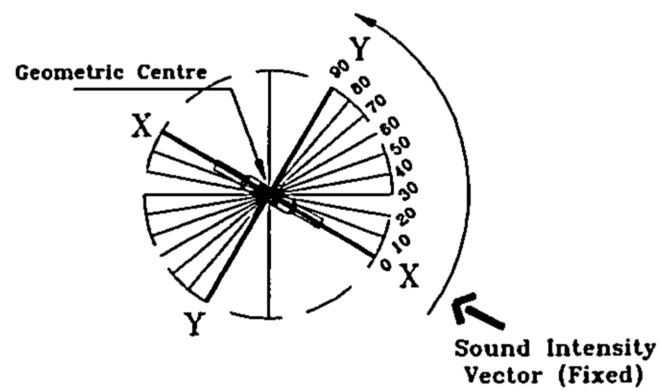
In each orthogonal direction +X-X, +Y-Y and +Z-Z (Case 1), impulse responses using short *m-sequence* signals were captured using a 1/2" sound intensity probe with a face to face configuration mounted in the holder described in Fig. 3.2; then the measurement was repeated for the same receiver position but with the probe reversed by 180°; i.e. -X+X, -Y+Y, -Z+Z (Case 2). The acquired data was then processed to yield the sound direction at octave frequencies from 500 to 8000 Hz



Semi-Anechoic Chamber

① Probe Location ② Isotropic Loudspeaker ③ Absorptive Foam ④ Fibre Glass Wedges

(a) Experimental setup.



(b) Orientations of the microphones' probe coordinates relative to the true intensity vector.

Fig. 4.10 The test room configuration and the experimental setup for investigating the directional detection accuracy, a) Experimental setup, and b) Orientations of the microphones' probe coordinates relative to the true intensity vector.

with a 12 mm probe spacing and from 250 to 1000 Hz with a 50 mm spacing. Measurements were conducted for different probe orientations starting from 0° where the probe axis coincides with the direct sound, to 90° in 10° steps (i.e. 10 probe orientations). The indicated angles of incidence, the azimuth (θ) and elevation (ϕ), of the maximum direct sound was isolated and investigated for all receiver orientations before and after probe reversal; the indicated direction was also calculated from both direct and reversed measurements. Fig. 4.11 depicts the terminology of the error indicators which will be used to investigate the method's accuracy.

Figs. 4.12 to 4.18 show the received normalized (re. maximum) intensity vector after compensating for the rotational angle of the probe system in the horizontal plane XY; and in the vertical plane XZ; part (*a*) in each figure presents direct measurements while part (*b*) shows the direction when calculated from both direct and reversed measurements. Ideally, all measured vectors over all probe orientations should indicate the same magnitude and direction (i.e. all vectors should coincide); this was not the case particularly in the XY plane. As can be seen in Fig. 4.12 at 8 kHz (one octave band frequency) the measured vectors are received in the XY plane with different magnitude over a directional range, *R* (i.e angular difference between vectors lower and upper bounds) of around 20.3° with a standard deviation, *STD* of $\pm 6.4^{\circ}$ about the mean direction. The mean positive azimuth error $E_{\theta,m+}$ is found to approach 4.1° while the minimum negative azimuth error $E_{\theta,m-}$ was -6.1°. The mean absolute azimuth error $E_{\theta,abs}$ was 4.9°.

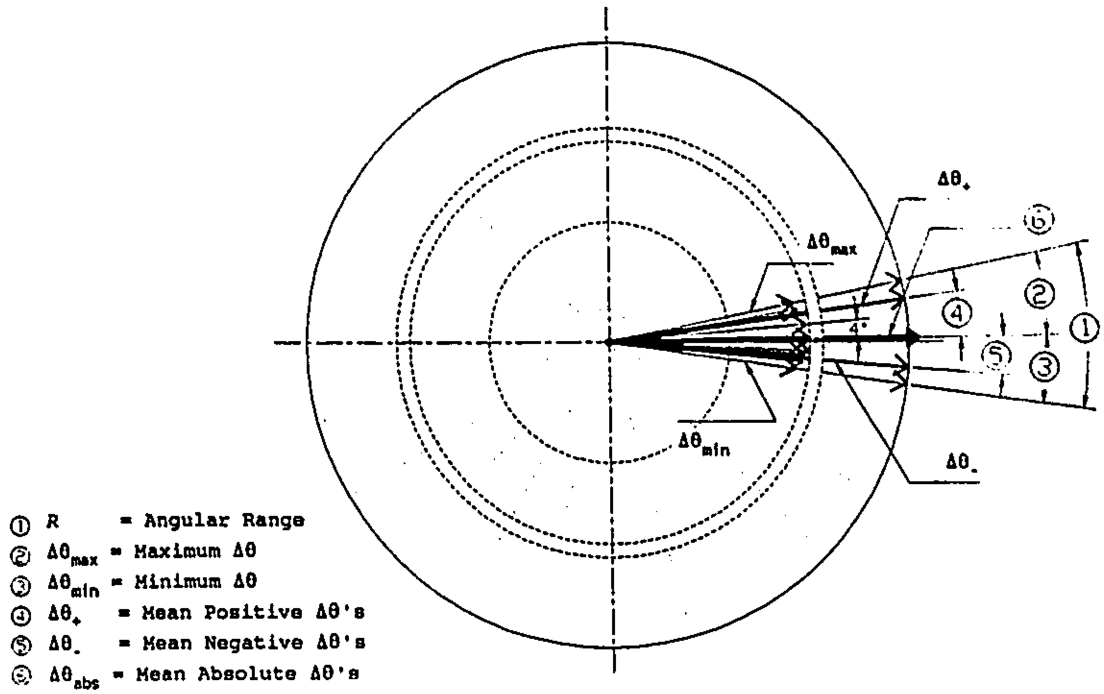


Fig. 4.11 Illustration of the terminology employed to describe the errors of the direction detection.

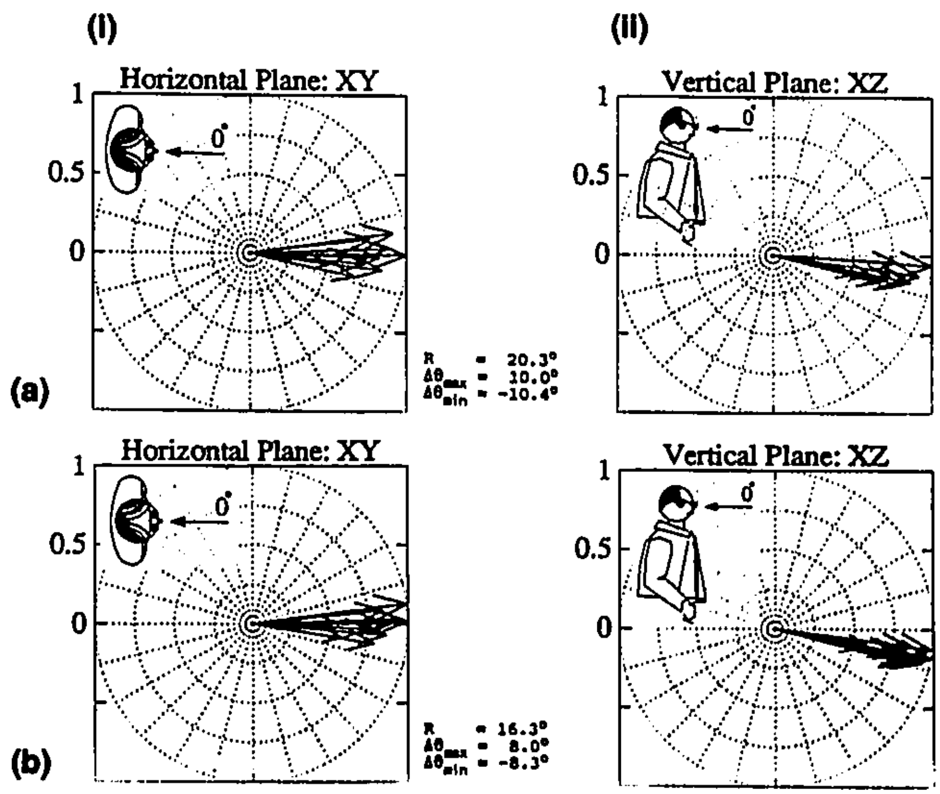


Fig. 4.12 The normalized (re. maximum) sound intensity vectors received over all probe orientations at 8 kHz octave band with a 12 mm microphones' spacer in the

These indicative parameters improved to 16° *R*; ±5.5° *STD* and $E_{\theta,m+}$ 3.6°, $E_{\theta,m-}$ -5.4°, $E_{\theta,abs}$ of 4.3° upon probe reversal. In the XZ plane the directional range *R* was around 11°, ±5.4° *STD*, and $E_{\phi,m+}$ 5°, $E_{\phi,m-}$ -2.2°, $E_{\phi,abs}$ of 3°; the order of magnitude of these parameters improved to approximately a value of 6.8°, ±2.3°, 1.5°, -2.2° and 1.7° respectively upon probe reversal. The wide directional range and large deviations even for probe reversal corrected results in both planes can be explained by the inadequacy of both microphone size and probe spacing for this particular frequency (usually 1/4" microphones with a 6 mm spacing would be required).

Both magnitude and directional error ranges are quite improved for normal measurements at 4000 Hz as shown in Fig 4.13 (in the XY plane: ±12° *R*, ±4.3° *STD* and a value of 3.8° for $E_{\theta,m+}$, $E_{\theta,m-}$, and $E_{\theta,abs}$; while in the XZ plane: ±2.4° *R*, ±1° *STD* and $E_{\phi,m+}$ 1°, $E_{\phi,m-}$ -0.7°, $E_{\phi,abs}$ of 0.9°). The error indicators upon probe reversal improved in a similar manner as at 8 kHz but with less magnitude. At 2000 Hz the intensity vectors (shown in Fig. 4.14) in XY plane: 6.5° *R*, ±2.3° *STD*, $E_{\theta,abs}$ of 2°; in the XZ plane: 3° *R*, 0.9±1° *STD*, and $E_{\phi,abs}$ of 0.6°. In case of 1000 Hz octave band, Fig. 4.15, (XY plane: 2.8° *R*, ±1° *STD* and $E_{\theta,abs}$ of 0.9; while in the XZ plane: ±4.6° *R*, ±1.4° *STD* and, and $E_{\phi,abs}$ of 1°) and as can be observed error ranges and deviations are decreasing with decreasing frequencies to 1 kHz but starts to increase at 500 Hz as shown in Fig 4.16(a) (when the same 12 mm probe spacing is in use) to a range of 22.7°, ±8.2° *STD* and $E_{\theta,abs}$ of 6.8° in the XY plane; 16.8° *R* and ±6.2° *STD* and 5.6° $E_{\phi,abs}$ in the XZ plane. When

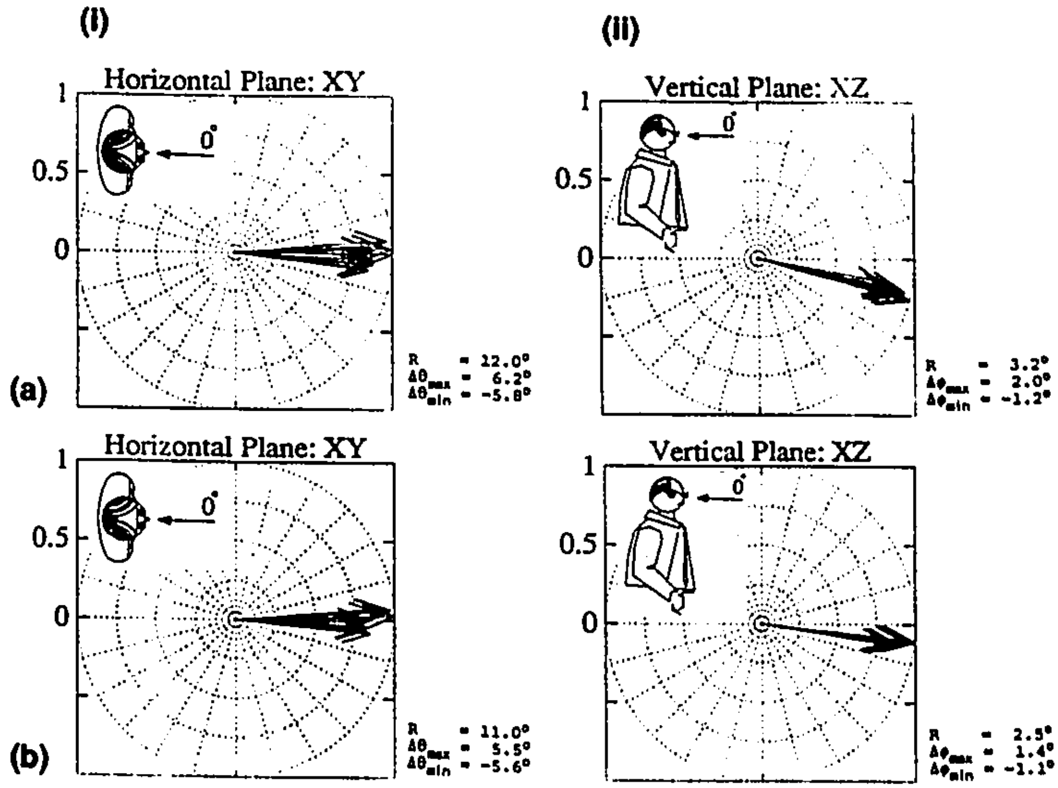


Fig. 4.13 Same as Figure 4.12, at 4 kHz octave band

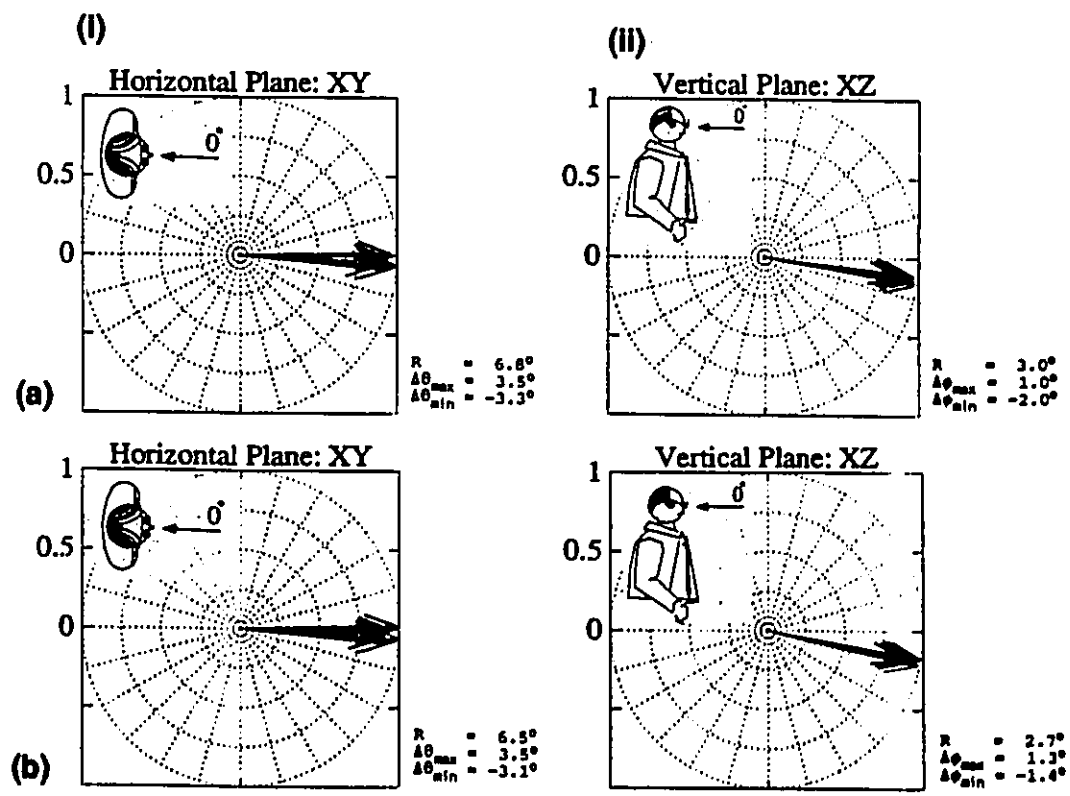


Fig. 4.14 Same as Figure 4.12, at 2 kHz octave band

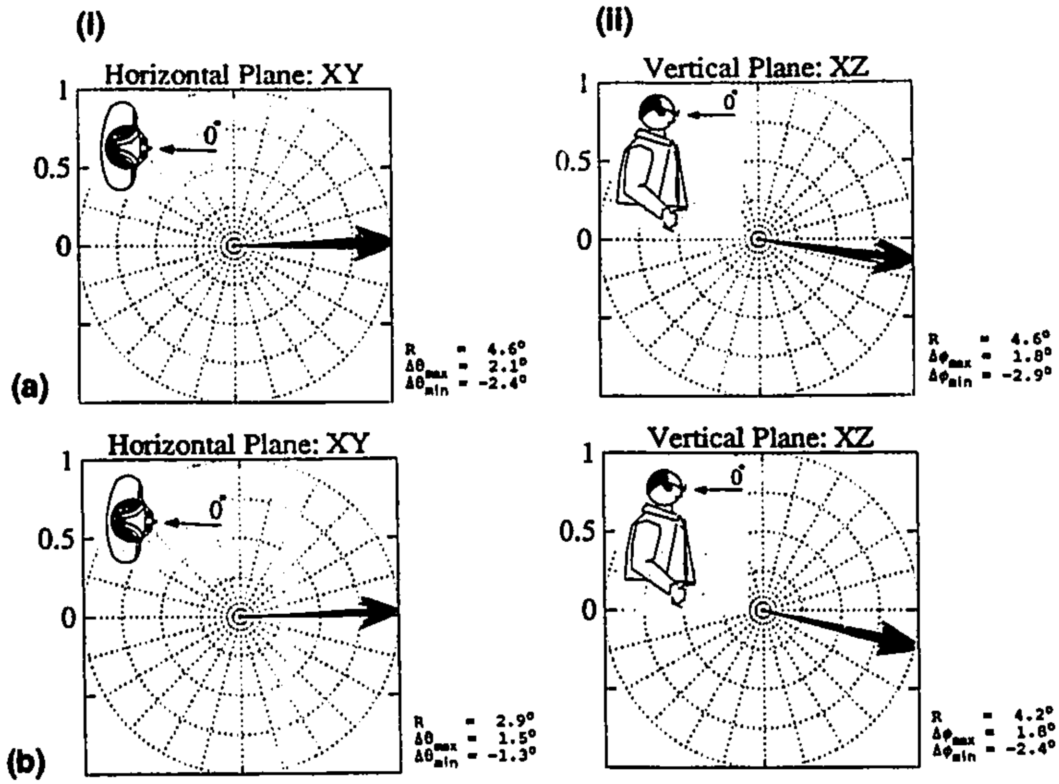
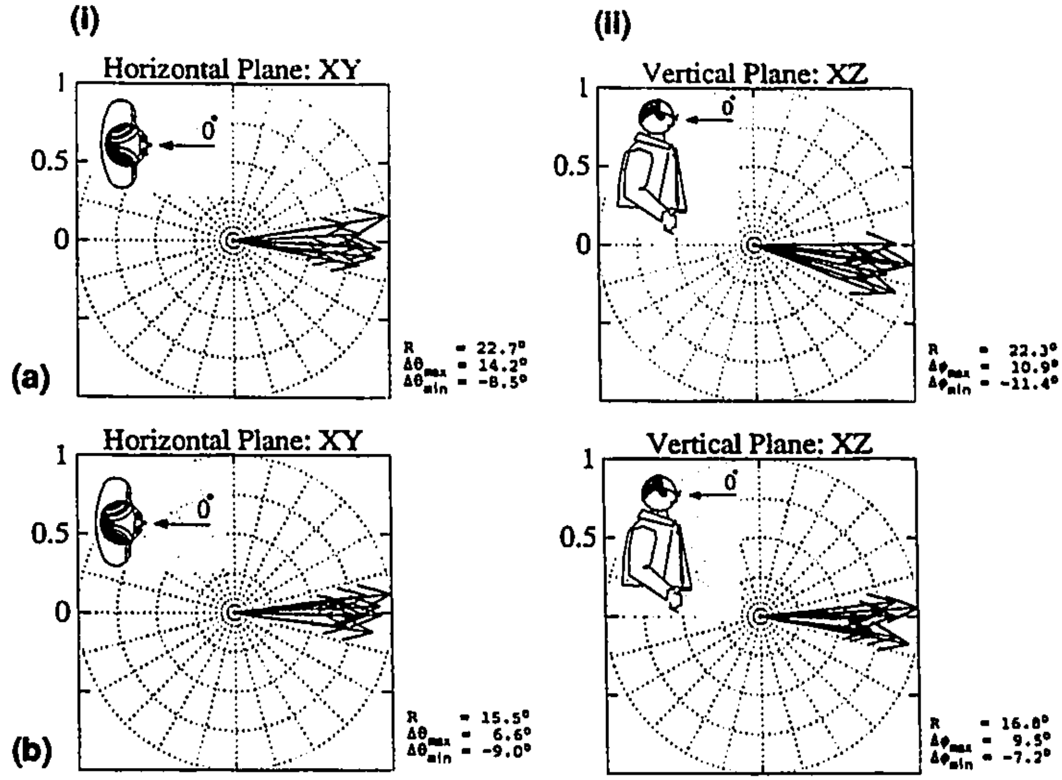
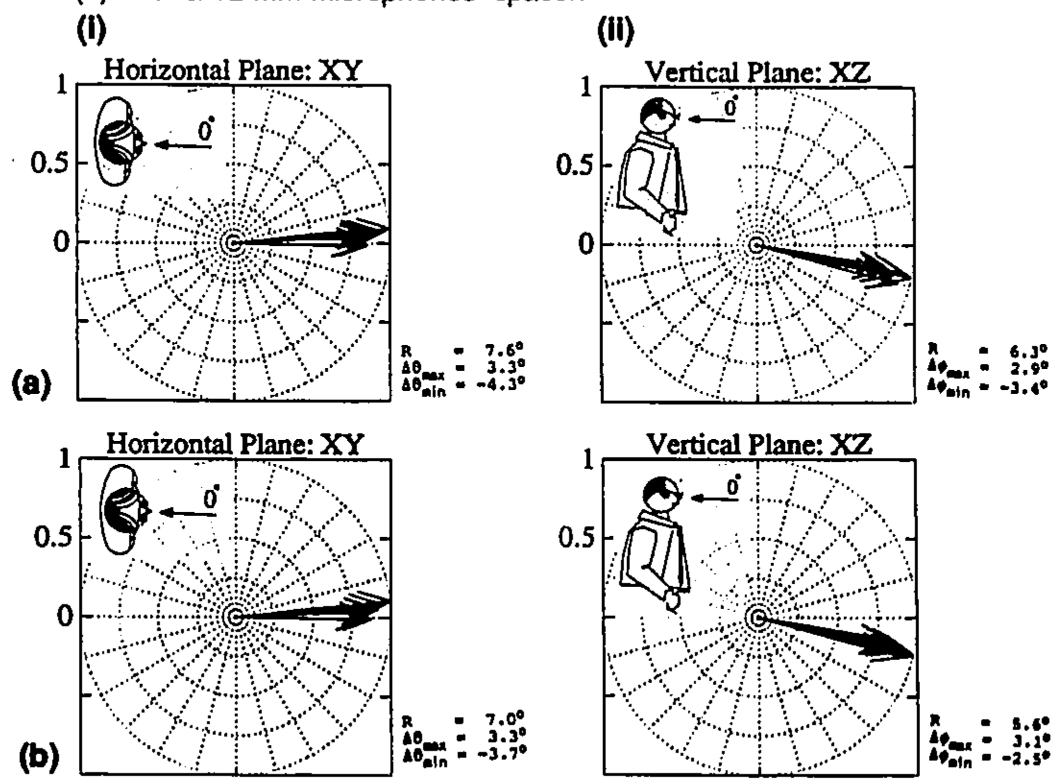


Fig. 4.15 Same as Figure 4.12, at 1 kHz octave band



(a) With a 12 mm microphones' spacer.



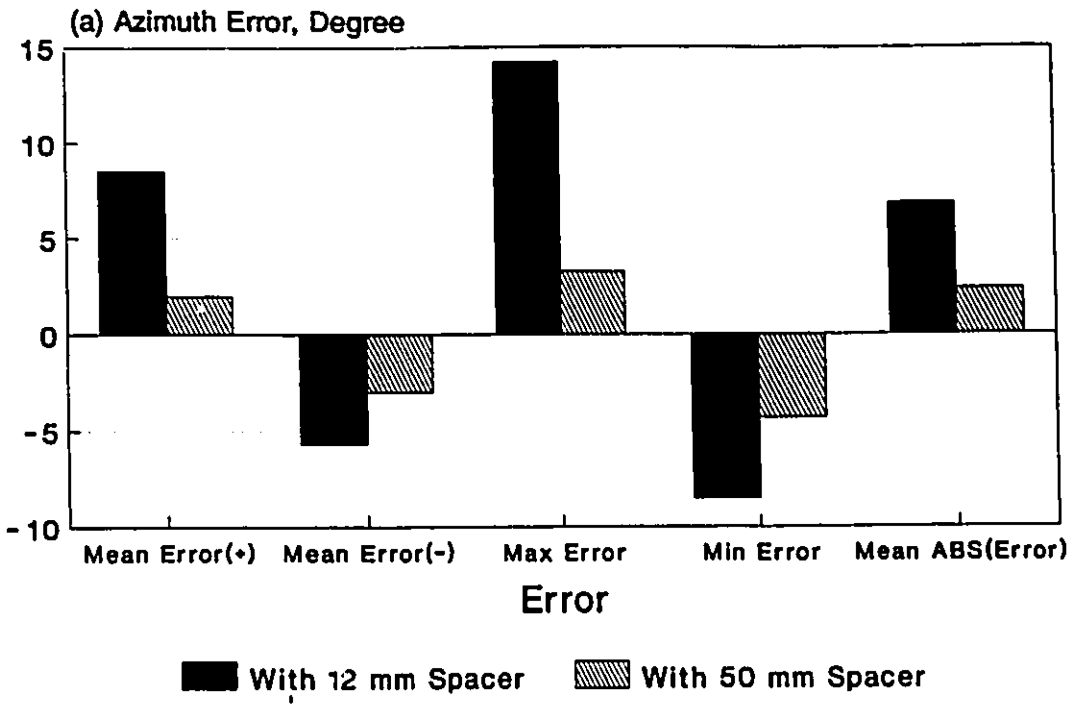
(b) With a 50 mm microphones' spacer.

Fig. 4.16 Same as Figure 4.12, at 500 Hz octave band, (a) With a 12 mm microphones' spacer, and (b) With a 50 mm microphones' spacer.

switching to a probe spacer of 50 mm the error range and deviation parameters at 500 Hz are drastically improved by a factor of 2 to 3 in both planes i.e. to the range of ±7.6°, ±2.9° *STD* and 3.2° $E_{\theta,abs}$ in the XY plane while it improved to 6.2°, *R* ±1.8° *STD* and 2.4° $E_{\phi,abs}$ in the XZ plane. These trends can be readily observed when comparing the intensity vectors shown Fig. 4.17(a) to those in Fig 4.17(b). The differences of the directional error indicative parameters calculated at 500 Hz with a 12 mm and 50 mm probe spacer can be seen in the bar charts shown in Fig 4.17(a) and (b). It is therefore recommended and will be adopted in the system measurement procedures to use a 50 mm probe spacer instead of a 12 mm spacer for mid-frequencies, particularly at 500 Hz and below for investigating the sound field directional characteristics in real hall measurements.

At low frequency such as 250 Hz and probe spacer of 50 mm the error and deviations of incident sound vectors (shown in Fig 4.18) increased to a range of 12.5°, ±4° *STD* and 3.2° $E_{\theta,abs}$ in the XY plane and to 9.9° *R* and ±2.8° *STD* and 2° $E_{\phi,abs}$ in the XZ plane. The error bounds at lower octave band frequencies of 63 and 125 Hz were not investigated but are expected to increase compared to those found at 250 Hz, however with a 100 mm spacer the resulting directional errors could be reduced by an order of magnitude as results proved for mid-frequencies.

To summarize the results of directional error investigation, Fig. 4.19(a,b) and Fig. 4.20(a,b) show the mean positive and negative azimuth errors, lower and upper



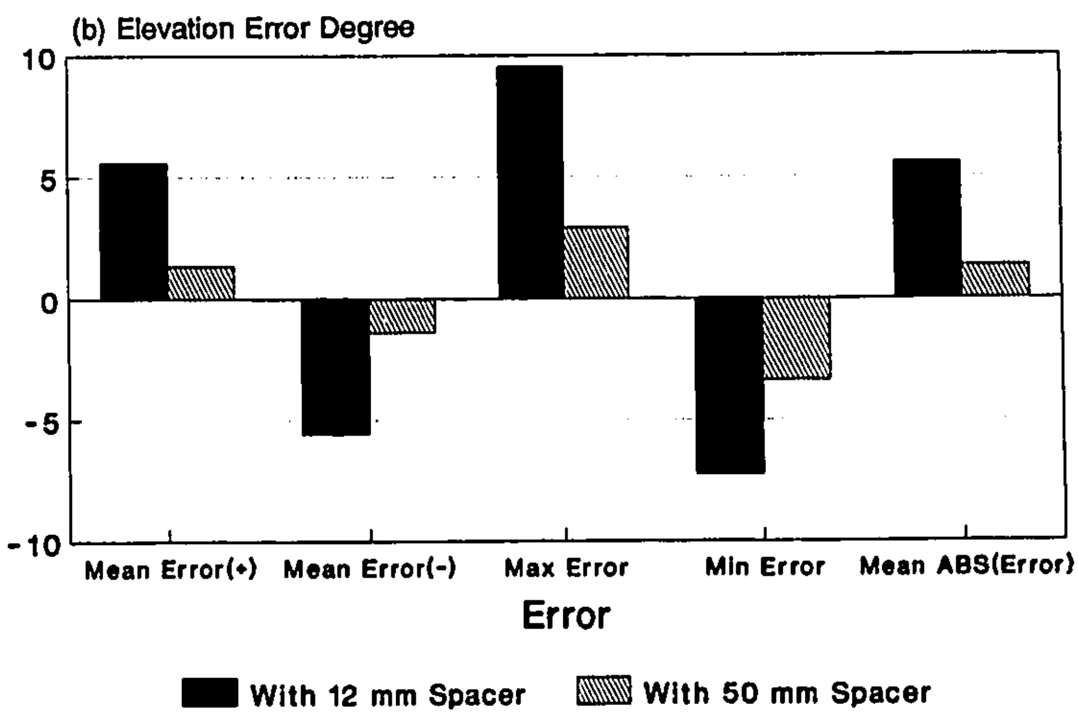
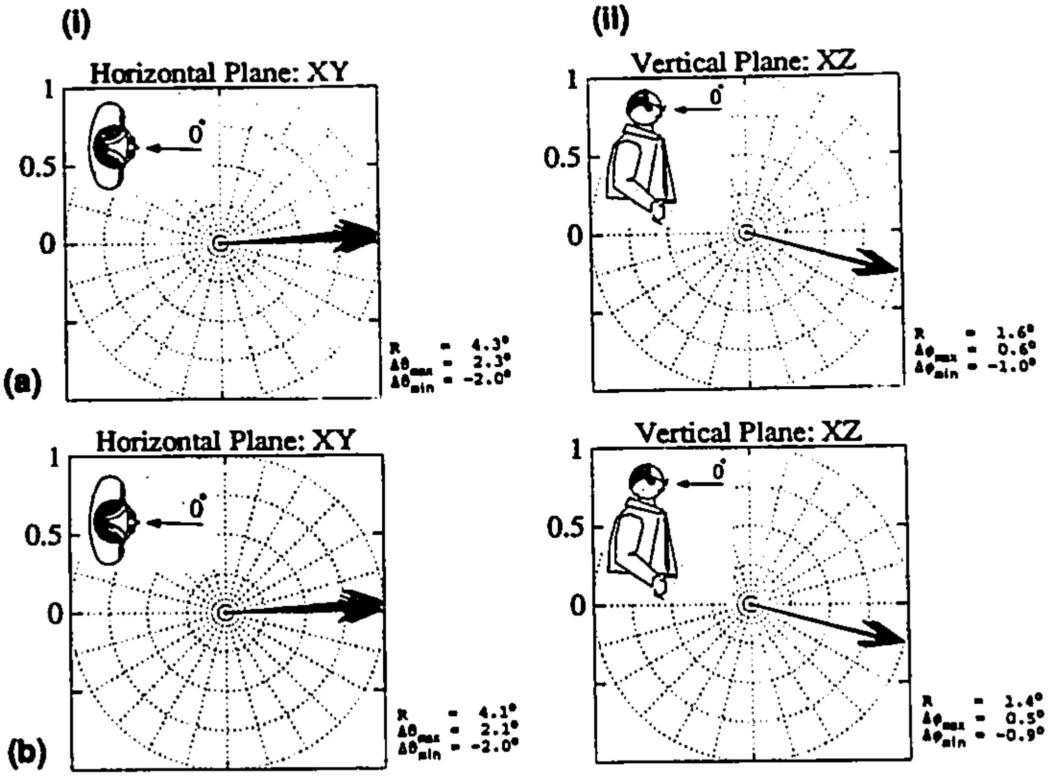
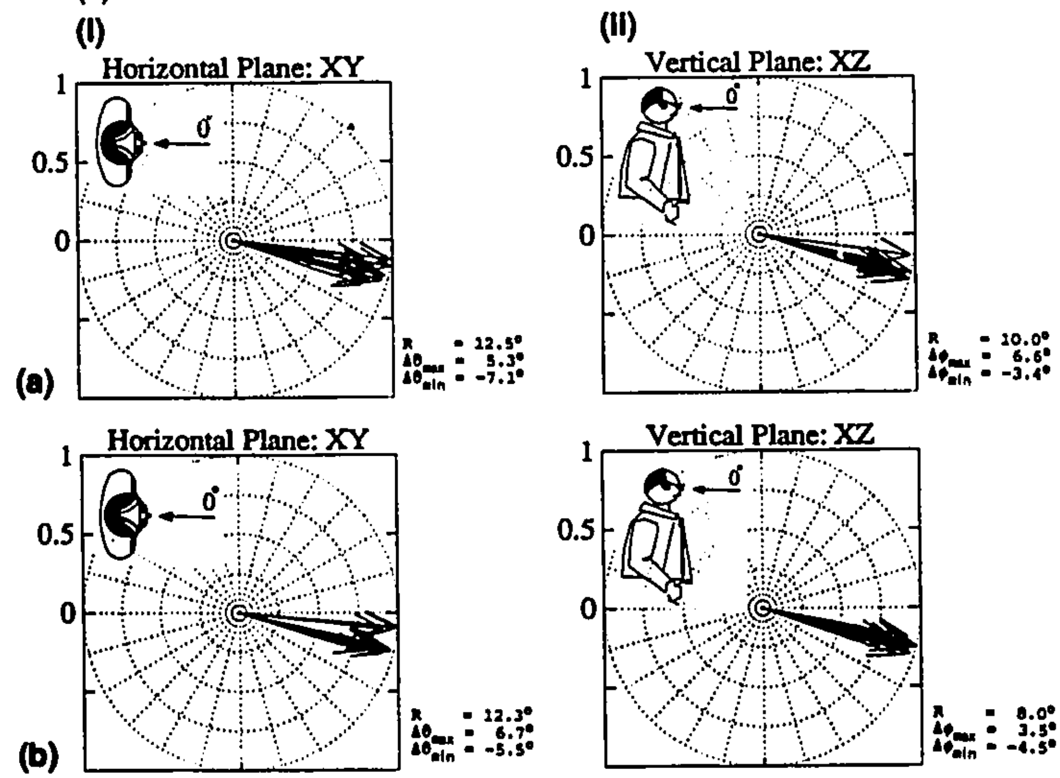


Fig. 4.17 Differences of directional error at 500 Hz, with 12 mm and 50 mm microphones' spacing, (a) Azimuth errors, and (b) Elevation errors.



(a) at 1 kHz octave band.

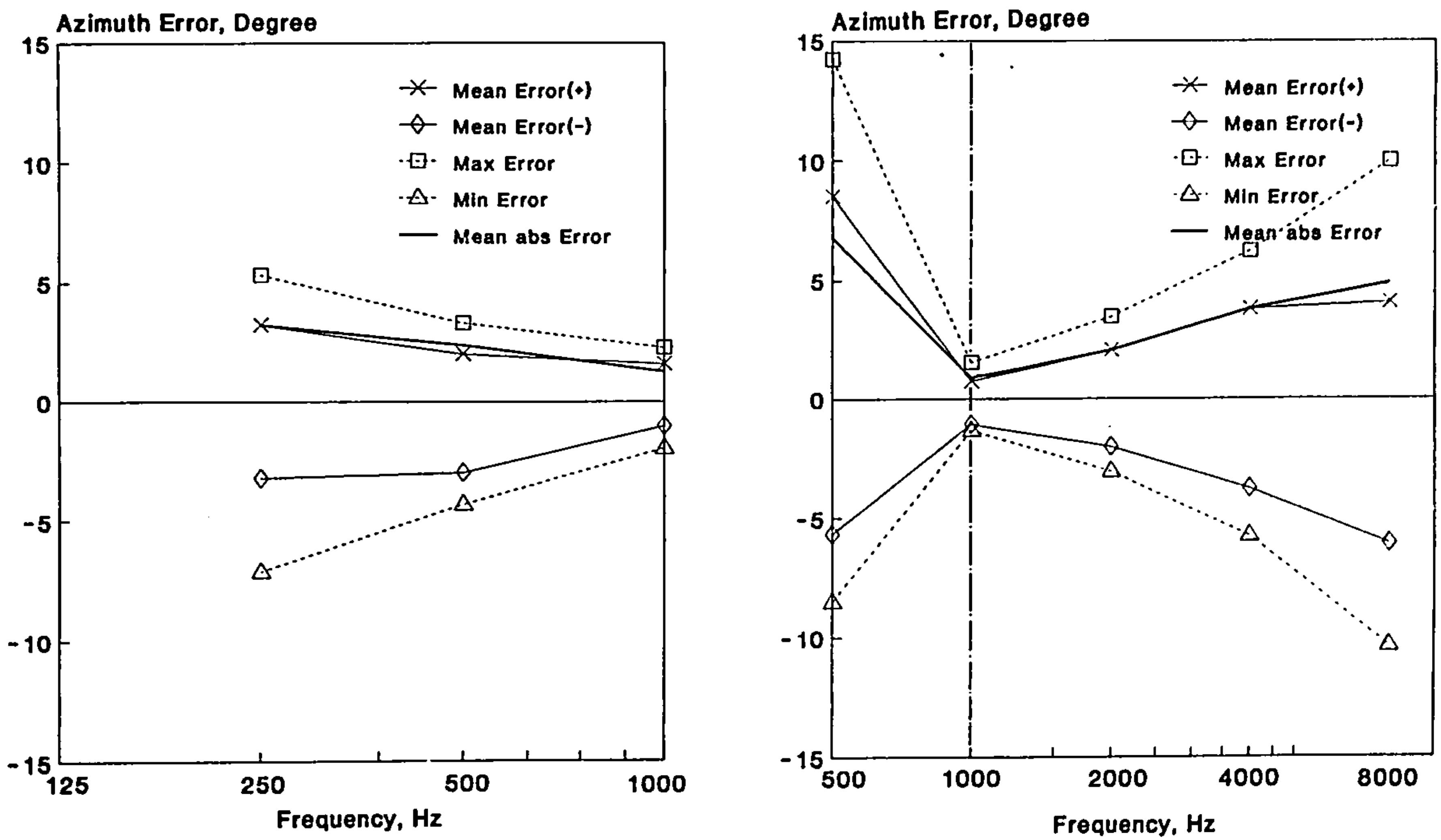


(b) at 250 Hz octave band.

Fig. 4.18 Same as Figure 4.12, (a) at 1 kHz, and (b) at 250 Hz octave bands with a 50 mm microphones' spacer.

direction sensing errors and the mean absolute error, over all recorded orientations of the probe, versus frequency, in the horizontal and vertical planes, for 12 mm (500 Hz to 8 kHz) and 50 mm probe spacing (250 Hz to 1 kHz). The direction angle errors at 4 and 8 kHz are high due to the inadequacy of both probe spacing and microphone size for these particular frequencies; the angle error also decreases with decreasing frequency when the proper probe spacer is used but increasing at lower frequencies. It was also found that the maximum angular range difference in the horizontal plane upon probe reversal were 0.6^o, 0.8^o and 1^o at 500 Hz, 1 and 4 kHz respectively so that measurements without probe reversal may be judged as reasonably accurate. The microphone types employed here were of the first generation intensity probe type; significant improvements in reduced phase error effects can be anticipated using more recent designs with a consequent improvement in directional accuracy.

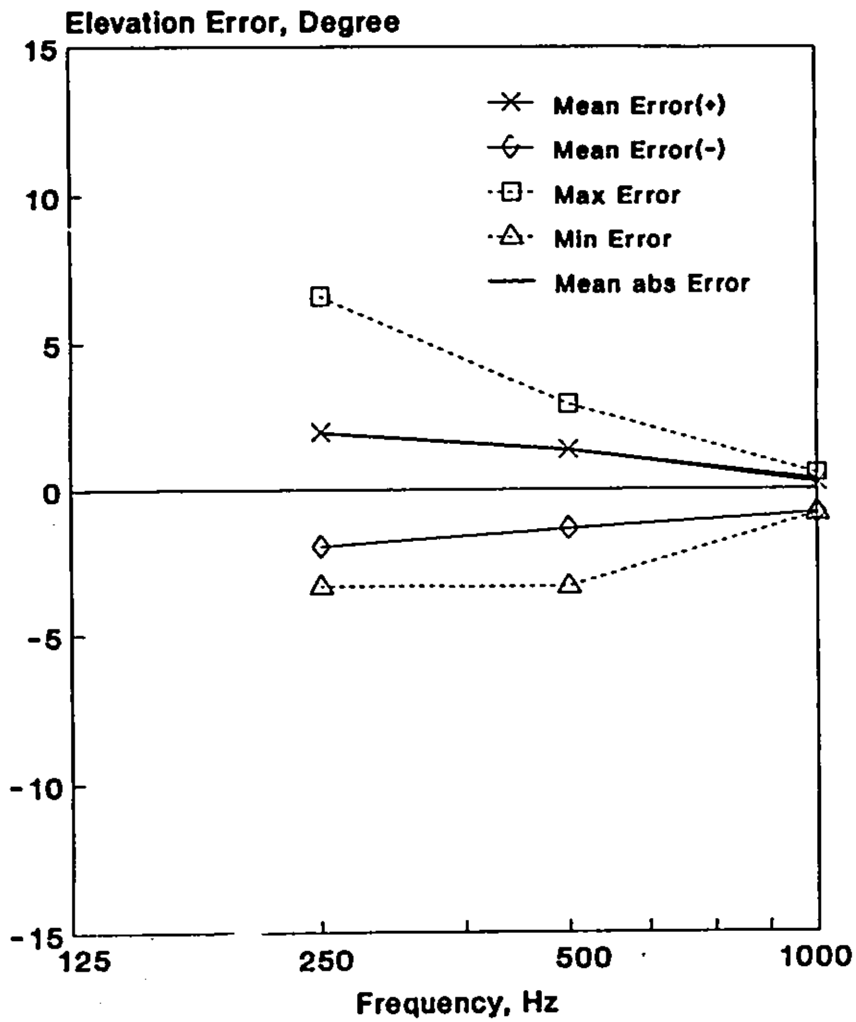
In practical measurement situations the issue of accuracy is relative both to the distance of the listener location to the boundary surfaces and to the size of the surface details under study. In large halls this distance for example can correspond to a transient time of 50 ms or more. With the present uncorrected maximum directional accuracy of the system i.e. 5.3^o at 250 Hz; the dimensions of interior design details should be greater than approximately 1.6 metres at a distance of 17 metres to correctly be detected; naturally reflecting objects to wavelength parameters must also be satisfied (preferably twice the wavelength, i.e. 2 x 1.37= 2.7 m, at 250 Hz). Architectural details such as a cantilevered balcony or a wall reflective panel will usually dominate in their vicinity and their influence or lack thereof at a listener position should be readily detected.



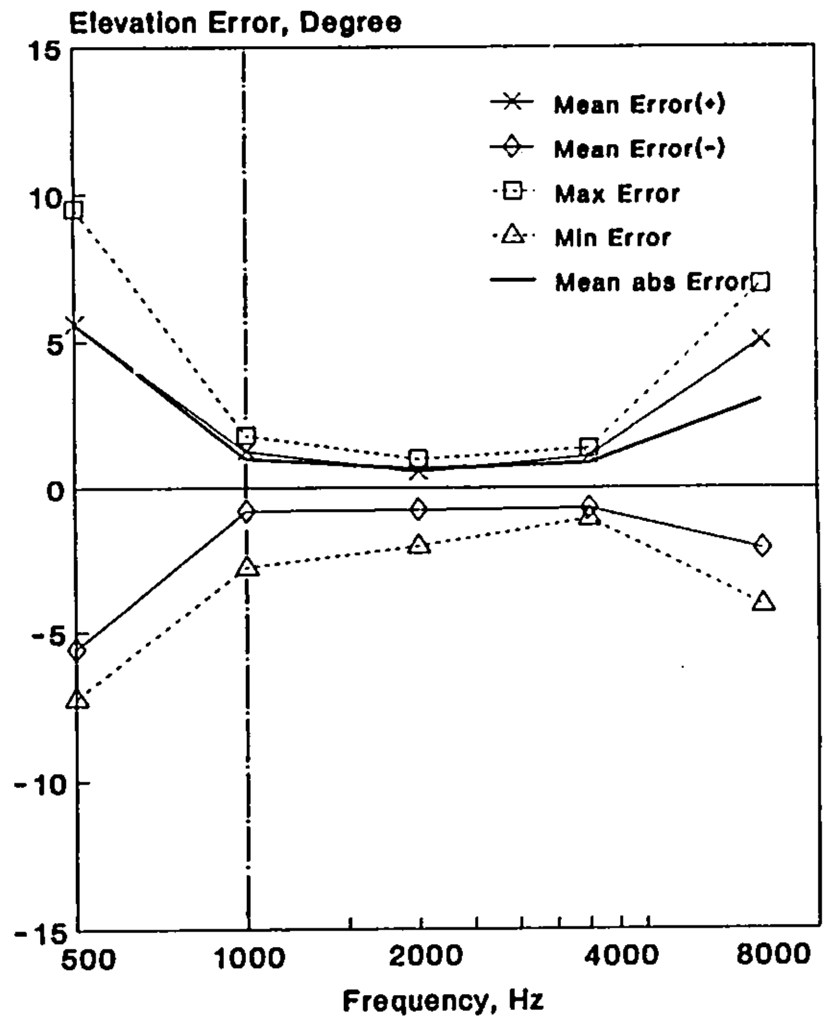
(a) With 50 mm microphones' spacing (250-1000 Hz).

(b) With 12 mm microphones' spacing (500 Hz-8 kHz).

Fig. 4.19 Directional azimuth error of the measuring system, versus frequency, (a) With 50 mm microphones' spacing (250-1000 Hz), and (b) With 12 mm microphones' spacing (500 Hz-8 kHz).



(a) With 50 mm microphones' spacing (250-1000 Hz).



(b) With 12 mm microphones' spacing (500 Hz-8 kHz).

Fig. 4.20 Directional elevation error of the measuring system, versus frequency, (a) With 50 mm microphones' spacing (250-1000 Hz), and (b) With 12 mm microphones' spacing (500 Hz-8 kHz).

CHAPTER 5

PROSPECTIVE QUANTIFIERS OF SOUND DIRECTIONAL DISTRIBUTION AND DIFFUSENESS

5.1 INTRODUCTION

Recently, Barron[144] commented that the largest gap in the objective description of concert hall acoustics appears to be the lack of a measure relating to subjective diffuseness or spatial distribution of the reverberant sound. Even the onset time for the diffuse sound field is debatable and a number of authors have proposed various possibilities. The objective of this chapter is to review known measurement methods and existing objective descriptors for quantifying sound field diffuseness in an enclosure. Their concepts, advantages, shortcomings and potential use are addressed, then prospective descriptors or quantifiers are proposed utilizing sound spatial information. Each descriptor is derived and discussed via experimental measurements of directional information in different sound fields.

5.2 DIFFUSE SOUND DEFINITION AND PROPERTIES

A widely accepted definition for sound diffuseness[3,5,27,145] states that a sound field is referred to as "*diffuse*" when the amplitude of the incident waves are uniformly distributed over all possible directions of incidence i.e. they have equal probability of being incident from all directions and subsequently an equal probability of impinging on the boundary surfaces of the enclosure at any angle. In addition the phase of these arriving waves should be randomly distributed and therefore, their combined effect can be determined by adding energies. In decaying sound fields these conditions should be fulfilled at each moment of the decay process or over short time intervals compared with the decay process duration.

5.3 IMPORTANCE OF DIFFUSE SOUND QUANTIFICATION

Sound field diffuseness is considered a crucial condition for the validity of the decay process in an enclosure and subsequently, most of the contemporary room-acoustic indicators are derived on this assumption. In room-acoustics, adequately diffuse sound is desired for acoustical quality. Lack of diffusion may occur due to either the characteristics of the enclosure geometry (e.g. parallel walls, shape proportions, and dimensions) and/or non-uniform distribution of boundary surfaces absorption. Traditional theory for a diffuse space divides the total sound into two components, the direct and reflected sound; the reflected sound is usually subdivided into early and late parts or regimes, the later arriving sound is then assumed to be diffuse. Simple diffuse-field theory predicts relatively constant sound pressure levels with increasing source-receiver distance under steady-state conditions and the decay of energy will then follow an exponential law. Measurements in existing halls show that the field is unlikely to be diffuse particularly in the late reflections period. Decay curves are neither exponential nor independent of the receiver position; for large distances from the source however they do approach classical decay theory. Barron[14] developed a revised theory to better explain the variations of sound levels in concert halls and Hodgson[146] reviewed knowledge about the accuracy and applicability of diffuse-field theory with respect to some acoustical parameters.

Recently, Soulodre and Bradley[147] undertook a subjective study on the influence of late arriving energy on spatial impression in concert halls and found that listener envelopment is produced by late arriving energy, not by early reflection alone; and it is affected by the level and arrival time; surprisingly, since late reverberation is usually considered detrimental to other subjective impressions such as music clarity and definition. The temporal and directional characteristics of the late reverberant reflections required to achieve a certain degree of envelopment however are still unclear.

Since late reflections associated with reverberation affect the subjective judgment of envelopment, and diffusion is closely related to reverberation, many arguments are raised. For example to what extent should the sound be diffuse and how could this be judged or quantified....?. Moreover since diffuse sound conditions are also necessary for many acoustical tests such as absorption measurements in reverberation rooms and transmission loss tests, should qualifying indicators be reconsidered..?. Methods of enforcing sufficient diffusion in a particular room can neither be decided upon nor their success or failure judged without an objective method and description of diffuseness degree.

5.4 WHERE AND WHEN DIFFUSE SOUND OCCURS

Where: One should remember that the application of statistical reverberation theory is only valid at a distance far from the sound source, namely a distance that avoids the dominance of the direct sound. This parameter is called the reverberation distance and can be expressed[3,5] by $R_h = 0.057.\sqrt{(V.Q/RT)}$, m, where V, the space volume, m^3, Q is the sound source directivity factor and RT is the space reverberation time in seconds.

When: The onset time at which it is of interest to judge diffuse sound is debatable and a number of authors with different concerns have attempted to define it. Kleiner *et al.*[148] (and references therein) in their study of the audibility of individual characteristics in reverberant tails in a concert hall comprehensively reviewed the existing objective onset time of the diffuse field. In summary this onset time was found to significantly vary among authors, namely being at 50, 80, 100, 100-150 or 160 ms after the arrival of the direct sound. Others were not concerned about defining a single number time limit but instead with the events that must occur before the diffuse state starts to take place, e.g. a time corresponding to the occurrence of 10-20 reflection orders, the last second order reflections or when the reflections density reaches approximately 2000 reflections per second. Based on the later, a rule-of-thumb for the start time of statistical reverberation was proposed[3], and is expressed by $T_{st} = 2.\sqrt{V}$, ms, where V is the space volume in m^3.

5.5 KNOWN MEASUREMENT METHODS AND OBJECTIVE DESCRIPTORS

5.5.1 Temporal Diffusion

Sound diffusion has been quantified from the pressure impulse response with respect to time by the "Temporal Diffusion", Δ proposed by Kuttruff [reported in reference[3]].

$$\Delta = \frac{\Phi_{(0)}}{\Phi_{max(\tau \neq 0)}} \qquad (5.1)$$

where,

$$\Phi_{(\tau)} = \int_0^\infty P(t) \, . \, P(t+\tau) \, dt$$

It characterizes the randomness of the impulse response by the ratio of the maximum value of the auto-correlation function of the impulse response, $P(t)$ at zero lag to the maximum value outside the origin. The greater the value of Δ the more temporally random the received reflections. The measure is a useful indication particularly when applied to the early reflection sequence in the bandwidth of interest. It could also be utilized to detect periodic pulses such as single or flutter echoes.

5.5.2 Spatial Diffusion

A very crude measure of spatial diffusion of a sound field is to check the spatial variance of the steady-state sound pressure level at different positions in the room excited by random noise; and is commonly used for defining diffuse field conditions in a reverberation chamber typically for the purpose of sound power testing. This method accounts for the amplitude gradient only, besides, for acceptable accuracy it should be evaluated in very narrow bandwidth (i.e. <= third octave bands).

Another objective descriptor, the directional diffusion index, depends on a knowledge of the sound field directional distribution. Sound directional distribution is characterized by the angles describing incidence (θ and ϕ) at an instant in time. By exciting the room with a stationary sound these parameters can be measured by scanning all directions with a directional microphone of high resolution then the degree of diffuseness at a receiver point can be quantified by the diffusion index proposed by Thiele[3,5,27]:

$$\Theta = 1 - (\mu/\mu_o) \qquad (5.2)$$

where,

$$\mu = \frac{1}{\Omega\langle \hat{I}\rangle} \int |\hat{I} - \langle \hat{I}\rangle| \, d\Omega$$

$\langle \hat{I} \rangle$ = average intensity, w/m^2,

$\hat{I}$ = incoming intensity, w/m^2,

Ω = solid angle of interest, and

μ_o = μ measured in free field with the same microphone

The index ranges from zero when the field is unidirectional to unity when reflections are equally distributed over all directions and the sound field is diffuse. However, the accuracy of the index is dependent on the microphone directional sensitivity which might modify the true directional characteristics of the sound field, therefore comparisons between workers employing different microphones are not possible. Furthermore, the directional distribution details and diffusion temporal dependence (i.e. when it occurred) can not be inferred, besides it accounts for all reflections irrespective of their delay time from the direct sound. The field can be considered sufficiently diffuse if the index at various points in the enclosure is not smaller than 0.6 [reference[27]]. Sound diffuseness can be also quantified by a graphical method which starts with the construction of the measured polar directional distribution against the characteristics of the used directional microphone itself and a unit circle representing an ideal diffuse field. The greater the area difference between the measured sound spatial distribution and the microphone directional characteristics, the more diffuse the sound is. Furduev and Chen-Tun[27] proposed an expression, based on measuring the normalized area differences, that varies from zero to one where a diffuse sound field exists.

An indirect but reliable measure is to calculate the correlation coefficient between the steady-state sound pressure signals at different locations in the room expressed by[3,5] :

$$\psi = \overline{P_1 P_2} / (\overline{P_1^2} . \overline{P_2^2})^{1/2} \tag{5.3}$$

The dependence of the correlation coefficient on the distance gives a general indication of the uniformity of the directional distribution in the room under investigation and takes into account the phase incoherence between the incident waves but since it is dependent on the distance of the measurement position, detailed comparisons between different listener positions can not be carried out.

Coherence between sound pressure and particle-velocity has been shown to reflect the nature of the sound field[149]. Theoretical diffuse field quantification by the two microphone intensity technique has been proposed by Gerges[150,151], where the coherence function between the acoustic pressure, *p(t)* and particle velocity, v*(t)* is employed as a quantitative indicator of the sound field diffuseness in a reverberant field and is defined by[150] :

$$\gamma^2(f) = |G_{pu}|^2 / (G_{pp} \cdot G_{uu}) \qquad (5.4)$$

where, G_{pu} is the cross spectrum of the pressure, *p(t)* and particle velocity, v*(t)* signals and G_{pp} and G_{uu} are the auto-spectrum of both respectively. A random error is associated with the calculations of such coherence functions and therefore the number of averages must be increased for low values[134].

5.6 PROSPECTIVE QUANTIFIERS

5.6.1 Visual Examination of Time-Segmented Directional Characteristics

Basically, the 3-D measurement method allows sound directional distribution in a given solid angle of interest and a desired time interval to be isolated and visually examined from different views for spatial uniformity of incident sound reflections. If the sound is fully diffuse, the received intensity vectors tend to be equally distributed in both magnitude and angle of incidence and therefore form a smooth round envelope with no significant irregularities or sudden dips otherwise irregularities of their level and coherence of directional incidence dominates. Fig. 5.1 shows an example of sound intensity vectors measured in a reverberant field for the time period 5 to 80 ms excluding the direct sound. Six principal directions* are shown and as can be seen in the figure parts **b** and **c** the reflections exhibit directional irregularities to the left compared to the right where the directional reflections are more or less uniform in magnitude (this was probably due to the influence of a stationary diffuser and rotating vane located to the right of the measurement point). In addition one should not expect as mentioned before to have a fully diffuse field realized over all directions in such a short period of time but it is likely to occur in the late part of the impulse response. Fig. 5.2 shows another example where the directivity patterns (in dB scale) in the horizontal plane are observed in a sliding time window of 50 ms in sequence, the transition from a directive sound in the first time window to a partially diffuse sound in the third

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* See Fig. 4.7, p. 4.11.

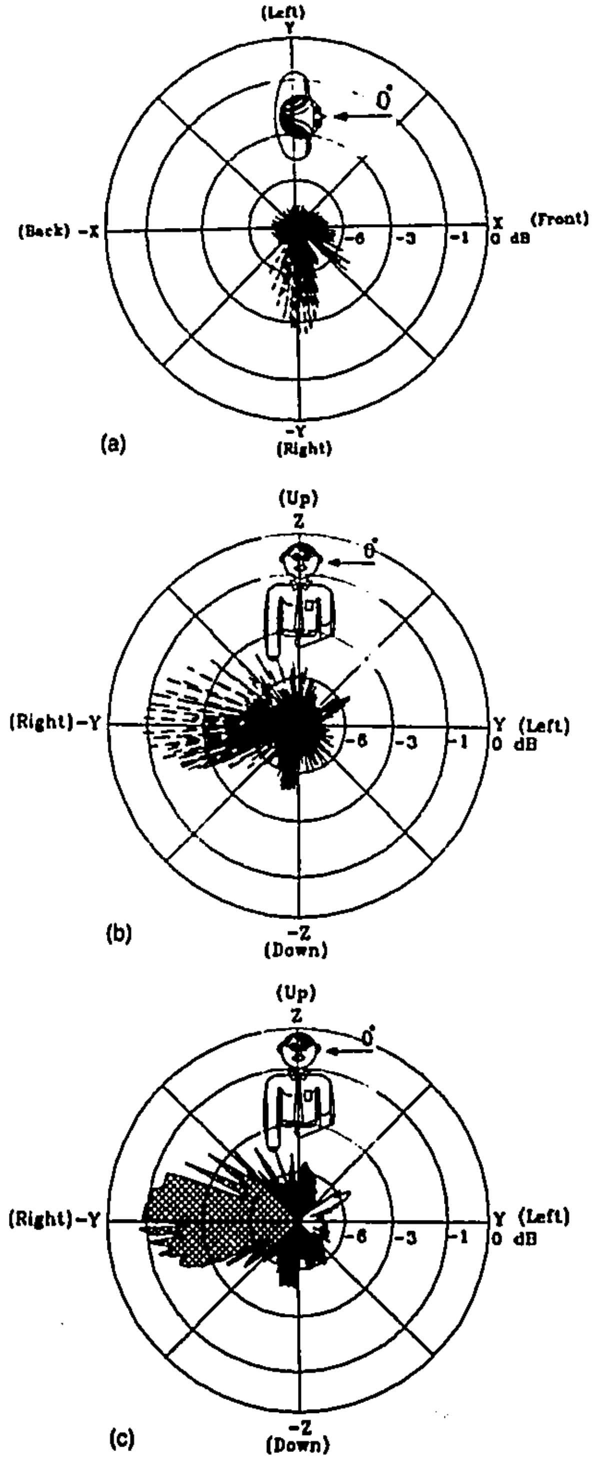


Fig. 5.1 An example of sound intensity level vectors at 500 Hz measured in a reverberant field for the time interval 5-80 msec in the, (a) Horizontal plane, (b) Lateral plane, and (c) envelope of the received reflections in the lateral plane. [Note: Sound intensity vectors are normalized with respect to the maximum direct sound intensity]

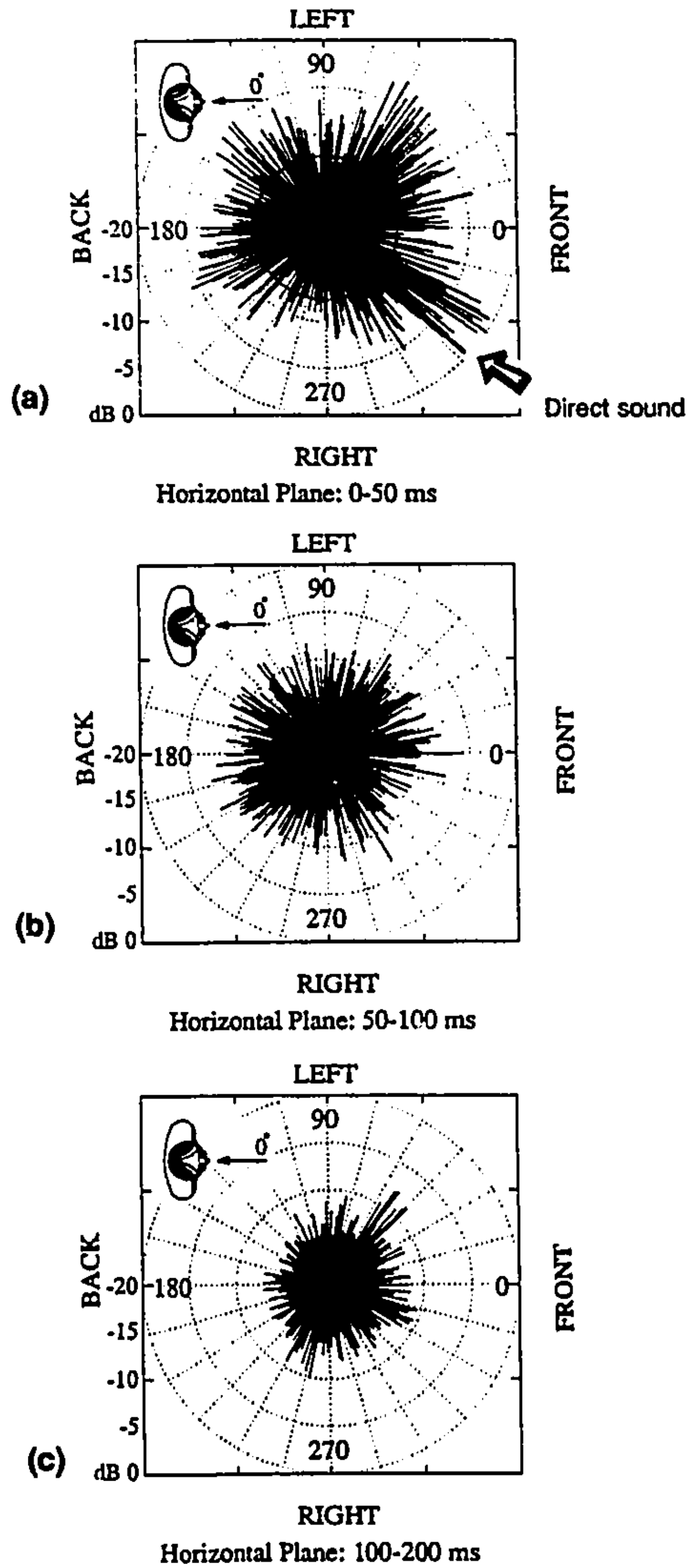


Fig. 5.2 An example of the directivity patterns for a measurement in reverberant field at 2 kHz, shown in the horizontal plane, within a sliding time window of (a) 0-50 msec, (b) 50-100 msec, and (c) 100-200 msec. (dB scale: 0 to -20 dB)

time window can be readily visualized and distinguished. A simple number indicator to represent each time segment can be expressed by the standard deviation of the magnitude of received reflections, over the specified time window.

5.6.2 Uniform Spatial Distribution

Based on a sound diffuseness definition of uniform spatial distribution and considering the projection of the incident sound intensity in only one plane, together with the assumption that all received vectors after the direct sound are more or less equal in magnitude, then the total number of vectors (*N*) should be equally distributed over 360° in a manner as shown hypothetically in Fig. 5.3(a). Each vector forms an angle equal to *n.(360/N)* with the direct sound where n is the order number of the incident reflection (*1,2,3,4,...N*). However, should the same number of vectors be received at the observation point but distributed over a very narrow angular range as shown if Fig. 5.3(b), one may then express the angular deviations of the directive case to the uniformly distributed case by the formulae:

$$D_\theta = 1 - \left(\int_{n=1}^{N-1} |\theta_m(n) - \theta_t(n)| \Big/ \int_{n=1}^{N-1} \theta_t(n) \right) \qquad (5.5)$$

where,

$\theta_m(n)$ = the n^{th} measured angle of the intensity vector to the direct sound.

$\theta_t(n)$ = the n^{th} theoretical angle of the intensity vector to the direct sound and is equal to $n.(\theta_T/N)$, θ_T being the total angle of assessment, and

N = total number of the received reflections in the chosen time window.

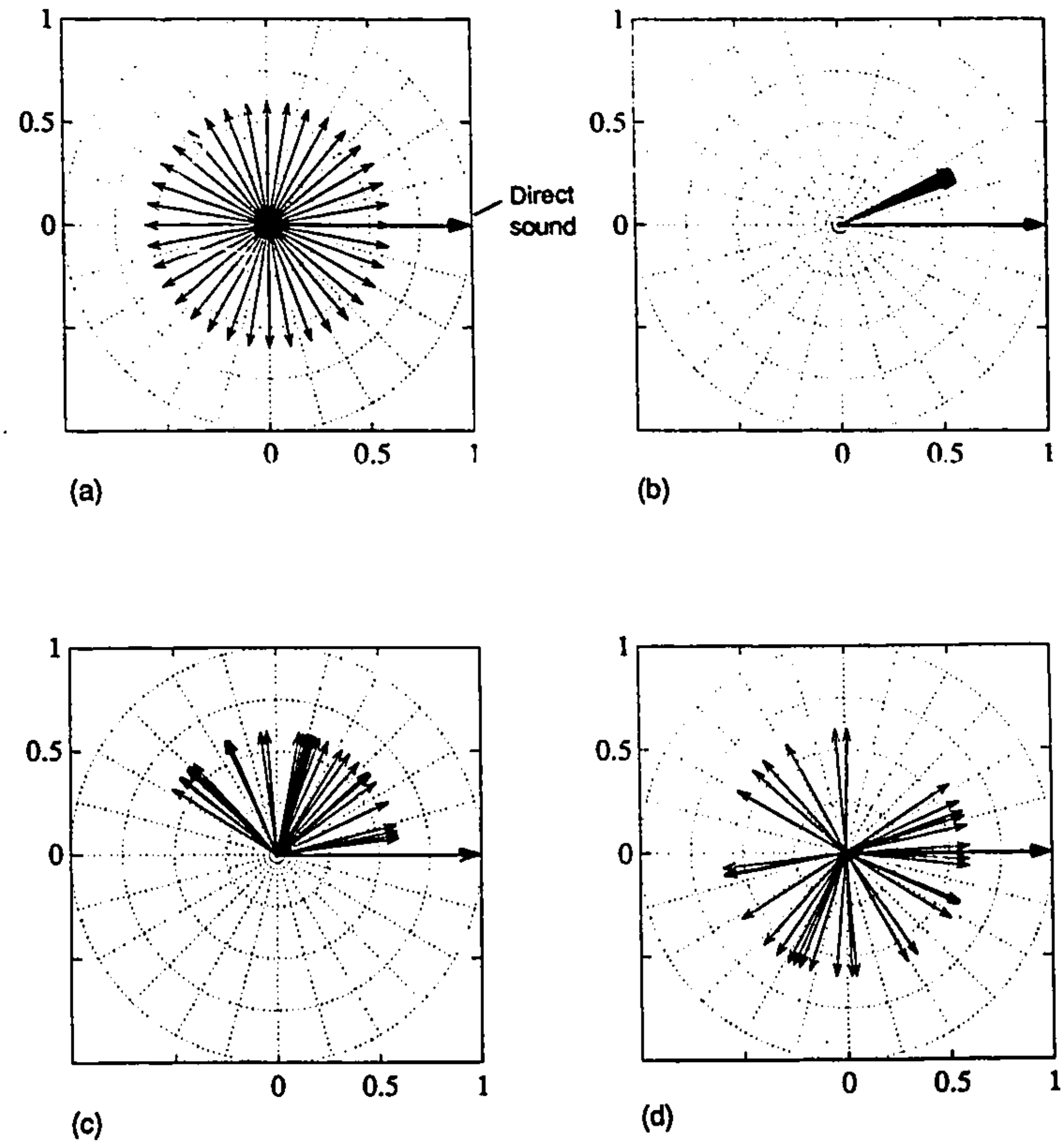


Fig. 5.3 Hypothetical intensity vectors distribution in the horizontal plane, (a) Uniformly distributed, $D_\theta = 1.0$, (b) Unidirectional, $D_\theta = 0.13$, (c) Randomly distributed in a narrow azimuth angle, $D_\theta = 0.3$, and (d) Randomly distributed over the 360° azimuth angle, $D_\theta = 0.4$.

The value of the second term of equation (5.5) will range between zero and unity and as the measured azimuth angle distribution approaches the theoretical uniform distribution its value reaches zero. In order to arrange a D_θ that increases with the degree of uniform sound distribution, the second term is subtracted from unity. The expression can be used to check sound uniformity in both the horizontal and vertical planes, namely θ and ϕ.

5.6.3 Net Sound Energy Flow

Generally speaking, an ideal diffuse sound field exists when the energy flow at a given position is the same in all directions for all arrival times, hence there is no acoustic net energy flow and the instantaneous sound intensity is zero. To quantify the sound field diffusion with respect to acoustic net energy flow, a "Directional Diffusion", *DD* is proposed :

$$DD = \frac{1}{\Delta t} \int_{t}^{t+\Delta t} \vec{I}(t)\, dt \Big/ \int_{0}^{T} |\, \vec{I}(t)\, |\, dt \qquad (5.6)$$

where, the numerator is the mean energy flow, w/m^2 in a given direction, that is the magnitude of intensity which would result if all intensity vector components on a given directional axis, received within the time period t to Δt, were then divided by the time period over which the assessment is considered. The denominator involving $|I(t)|$ is a measure of the total energy passing through the measurement point over the total impulse response period *T*. The resulting sign of *DD* indicates the direction of the net energy flow. Normalizing the resulting time-dependent values with the absolute maximum will yield values for the component index between ±1. If DD_x, DD_y and DD_z are the cartesian component Directional Diffusion calculated from equation(5.7), then Spatial Diffusion (*SD*) is given by:

$$SD = (DD_x^2 + DD_y^2 + DD_z^2)^{1/2} \qquad (5.6)$$

DD and/or *SD* if close to zero indicates that overall intensity vectors are evenly distributed directionally and in magnitude about the measuring point, suggesting that the sound field is diffuse. High *DD* or *SD* values imply a highly unidirectional sound field. In practice the time of start should be chosen to represent the end of the direct sound in the impulse response record, typically 5 ms; the window Δt may be fixed for variable t, for example it could be the early reflection time of the sound intensity vectors from 5 to 50 or 80 ms or the late part from 80 to 200 ms or 500 ms. *DD* and *SD* give a sense of net energy flow compared to the total energy received; and its value decreases with greater sound diffusion. The instantaneous sound intensity can be windowed by a successively sliding rectangular window of some interval (suggested between 1 and 5 ms) to obtain successive values of *DD* and hence *SD* from the end of the direct sound; this would indicate the change of the mean energy flow with time. An example is shown in Fig. 5.4 for a measurement in a reverberation chamber. It shows both the components of the directional diffusion, and the *SD* values for early reflections and late periods and as can be seen even after the first 100 ms the sound energy still exhibits some directivity as indicated by peaks. Naturally, Δt can also be chosen to be the full record, that is $\Delta t \rightarrow T$, to yield a single record number value for *DD* and *SD* ; in

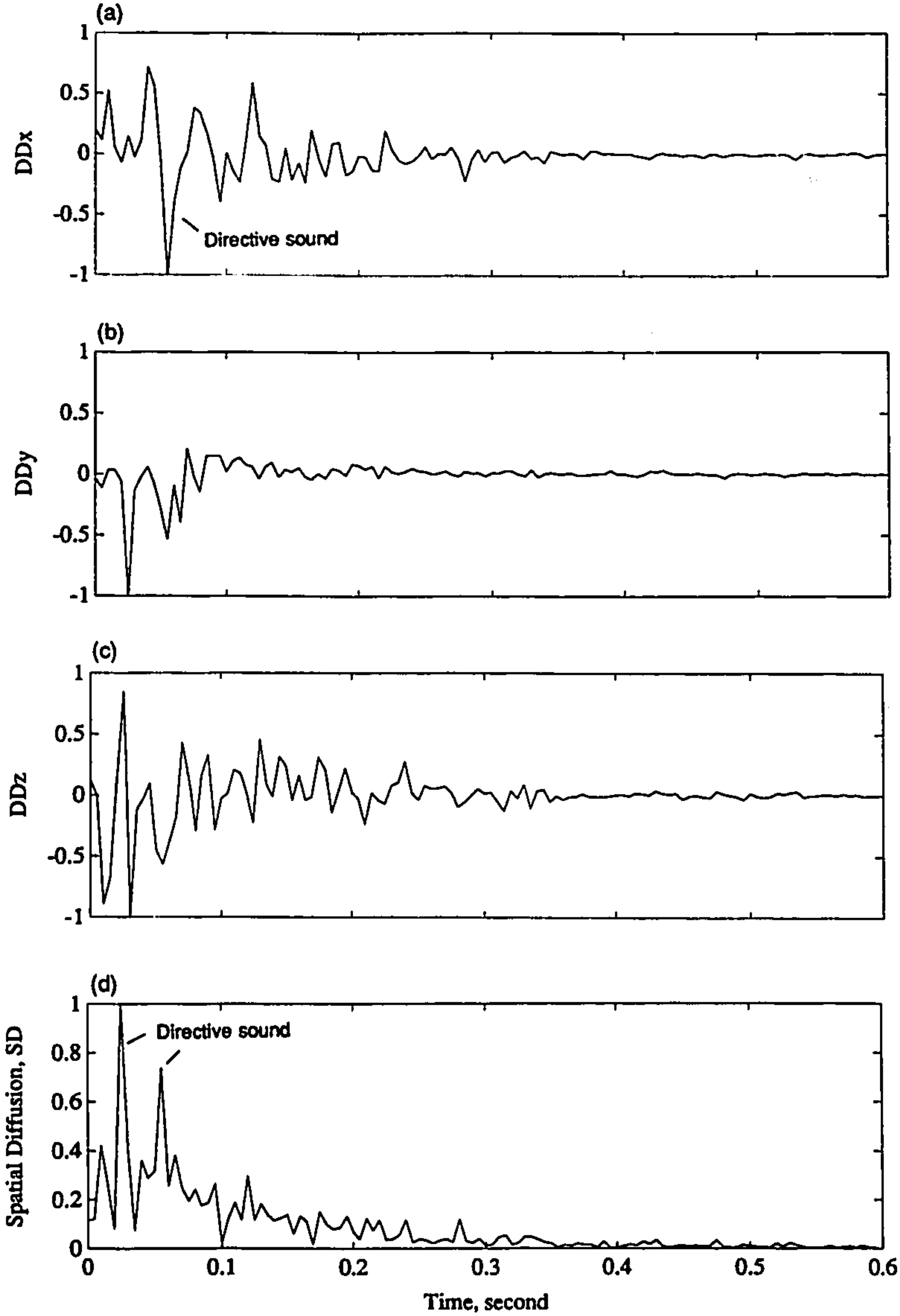


Fig. 5.4 Directional diffusion (*DD*) at 500 Hz vs time in a reverberant field: (a),(b),(c) Component directional diffusion DD_x, DD_y, DD_z, and (d) Spatial diffusion SD. [Note : A sliding time window of 5 msec is used]

this case some advantage is gained by expressing the denominator of equation (5.6) as a time average value so that perfect diffusion is signalled by zero and a unidirectional field is indicated by unity.

5.6.4 Directional Sound Decays

Diffusion of the sound field can also be viewed as the sound being isotropic in all directions so that, in the case of uniform spatial distribution of absorptive materials and equal probability of incidence, the sound decay in all directions should exhibit the same decay rate. Deviations of a directional decay component from the others indicates a lack of spatial homogeneity. This can be verified by examining the decay curves of sound intensity components compared to each other. Fig. 5.5 illustrates sound intensity decay curves for component direction intensities in a reverberant field and a small room at 500 Hz. The decay curves are obtained applying the "*Schroeder* " backward integration to the absolute values of the intensity response which expresses the received sound energy It can be seen that the instantaneous intensity component decays are close to each other in the case of a reverberant field and the slopes are similar compared to the case of a small room, the decay in the Y direction is sagging at some periods of time while the Z direction sound intensity decay deviates in manner which indicates unequal energy distribution; this can be attributed to absorptive suspended ceiling tiles and a variety of surfaces treatment and fittings in the test room. Directional sound intensity decay curves are found interesting and therefore their interpretational

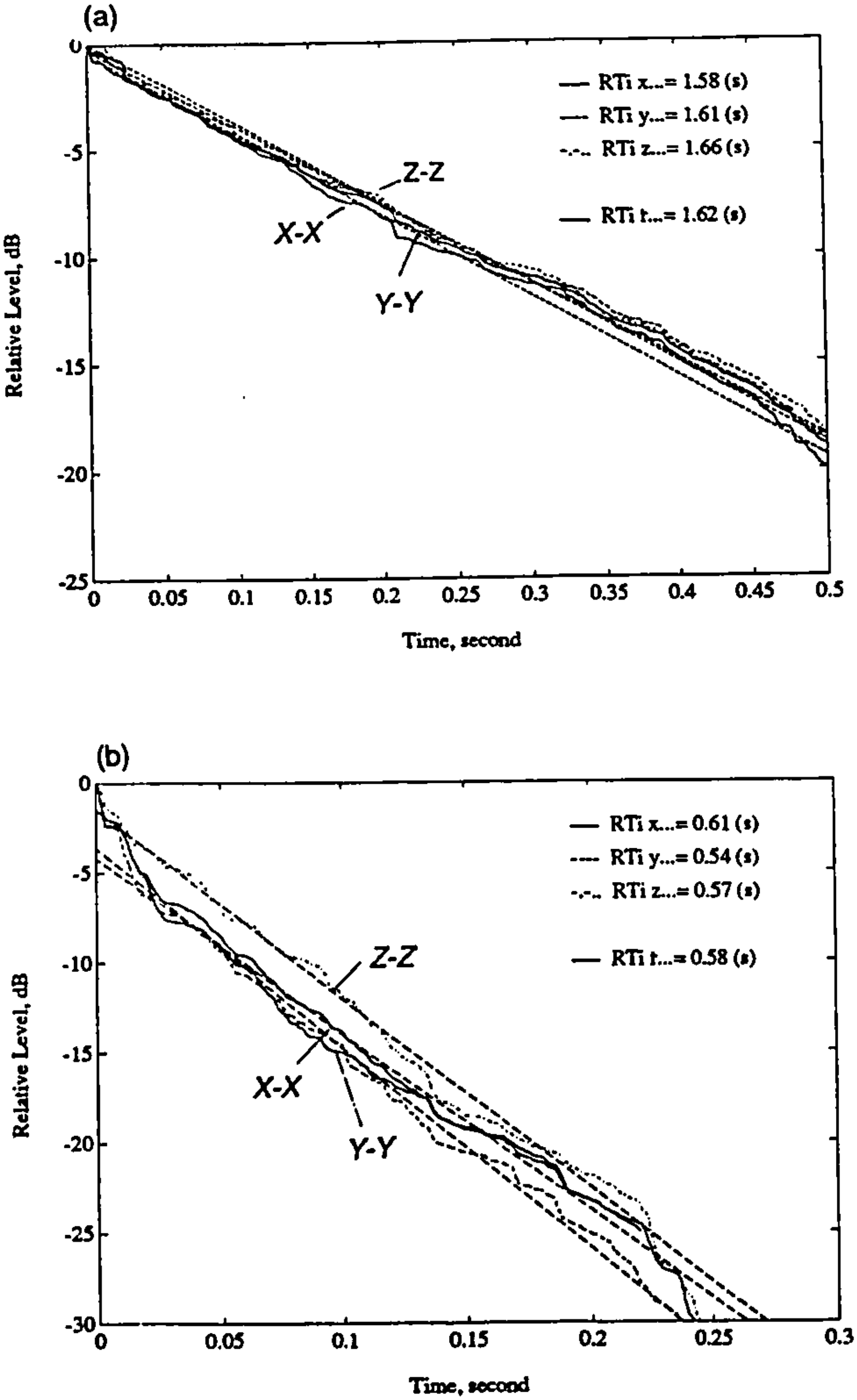


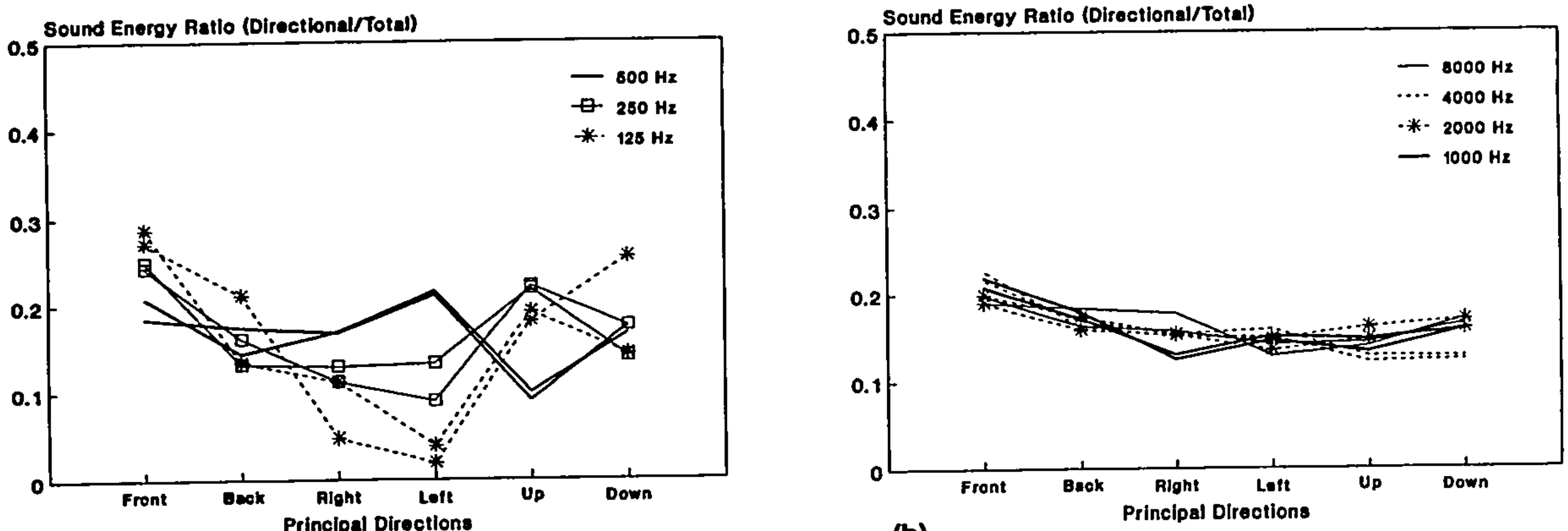
Fig. 5.5 Orthogonal component "sound intensity" decay curves at 500 Hz in, (a) Reverberant field, and (b) Small live room.(--- Linear regression fit)

characteristics will be investigated further in the analysis of the measurements in real halls (Chapter 6). It might be noted in passing that this particular measure could also be achieved by employing a standard figure of eight microphone in each cardinal orientation as used in *LEF* evaluation, that is, its measurement does not in fact involve sophisticated instrumentation.

5.6.5 Balanced Spatial-Frequency Sound Energy

A general indication of the sound waves homogeneity may be inferred from examining received spatial sound energy ratios at frequencies of interest and in six principle directions with respect to the listener (front, back, right, left, up and down). The sound energy ratio is defined as the total sound energy received from a particular direction to the total sound energy from all directions but excluding the direct sound. Fig. 5.6 shows a comparison of sound energy ratios received in six directions for measurements in a reverberant field. As can be seen in Fig. 5.6(b) the incident sound energy ratios at 1 to 8 kHz are almost equal in the reverberant environment compared to those at 125 to 500 Hz and shown in Fig. 6.5(a). The sound field appears to be partially diffuse at 500 Hz while it is not so at 125-250 Hz; this is likely due to the small size of the test room and its governing cutoff frequency.

Further work is required to determine which of these prospective measures might better serve as a quality indicator.



(a) (b)

Fig. 5.6 Comparison of sound energy ratios received in six principal directions for measurements in a reverberant field (i.e. reverberation chamber), a) at (125-500 Hz), and b) at (1-8 kHz),.

CHAPTER 6

DIRECTIONAL INFORMATION IN REVERBERANT SPACES: EXAMPLE APPLICATIONS OF THE SYSTEM AND MEASUREMENT METHOD

6.1 INTRODUCTION

This chapter describes the measurement and analysis of sound field directional characteristics within three contrasting large reverberant spaces; a variable concert hall, a chapel and a large church. The developed measurement system based on three-dimensional sound intensity measurements has been employed to assess sound quality and to obtain the directional characteristics of the sound fields. The purpose now is to show the applicability of the measurement method and its potential.

Reverberation time has been measured based on both pressure impulse response and three-dimensional transient sound intensity; similarities and differences are explored. 3-D transient sound intensity decay curves are also employed to investigate sound diffusion and example results of sound directional information are presented. The overall emphasis is to present detailed directional information with respect to the listener and to detect the effect of surrounding interior architectural features.

6.2 EXAMPLE APPLICATIONS: DESCRIPTIONS AND SOUND QUALITY ASSESSMENT

In this section a description of the geometry and spatial characteristics of each enclosure (measurement site) is presented followed by a summary of measurement results. The measurement procedure, conditions, and a more involved assessment of sound quality using contemporary room-acoustic indicators are reported in Appendices A-IV, A-V, and A-VI. Attention will be confined here to an assessment of directional characteristics and its related issues. The directional assessment was hitherto not possible to perform and as described here, will assist the reader in appreciating the advantages of its use. In order to avoid unnecessary complex evaluation, given the great increase in available information, an approach to a comprehensive acoustic evaluation with different levels of detail is also described.

In general the objectives of the present applications can be stated as follows:

- To characterize the acoustical properties of the space using state of the art room-acoustic indicators in order to classify the enclosure and to compare the criteria with reported results of other halls. In practice this objective is usually met by conducting similar measurements and typical analysis as provided in Appendices A-IV, A-V, and A-VI, though generally by monaural measurement equipment.

- To investigate the effect of the interior architectural features on the sound field directional characteristics.
- To employ the spatial information of sound for diagnostic purposes.

6.2.1 "*LOYOLA*" Concert Hall : The Hall Description

Room-acoustic measurements have been undertaken to assess the effectiveness of the adjustable side wall panels on the acoustic quality of the "***LOYOLA***" concert hall (Concordia University). The panels can be mechanically rotated to serve as either absorptive (denoted "OFF") or reflective (denoted "ON") surfaces and are intended to modify reverberance when required. Comments have also been received concerning the acoustic goodness at some particular locations.

The "***LOYOLA***" concert hall is described as follows:

- ***Hall shape*** : Classical shoe-box shaped hall, with steeply raked seating.
- ***Dimensions*** : 16.40 m (width) x 32.80 m (length) x 9.0 m (average height).
- ***Volume*** : 4800-5000 m^3 approximately.
- ***Capacity*** : 600 seats (main hall) + 20 seats (lounge).
- ***Special features*** : Side walls are equipped with mechanically rotating horizontal strip panels with one side reflective and the other absorptive surfaces.

Fig. 6.1(a) and (b) show the concert hall plan and longitudinal section with approximate dimensions. The selected measurement and source locations are each indicated by numerated letter (**R**) and (**S**) respectively and will subsequently be used as a reference for measurement results. Fig. 6.2 and Fig. 6.3 show interior views of the concert hall and its architectural design features. A summary of the selected measurement positions together with related comments is provided in **Table 6.1**.

The hall responses include two sets of measurements for two different conditions or configurations of the hall. These will be referred to in the following text and graphs as side panels "*OFF*", with the adjustable panels exposing their absorptive side, and side panels "*ON*" with the same source location (**S**) but the adjustable panels exposing their reflective side.

In general, measured mean mid-frequency (500-1000 Hz) values for both hall configurations show small appreciable audible change with respect to the audience impression* when the adjustable side wall panels are "*ON*" or "*OFF*". It is therefore evident that the adjustable side wall panels which are designed to enhance reverberation when required actually produce small audible changes to the acoustical quality of sound field in the hall. No significant change was found in the room-acoustic indicators, see **Table 6.2**, with the exception of C_{80} at low frequencies, when side wall panels were "*ON*" or "*OFF*", (that is whether the

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* See Appendix A-IV.

panels exposed a reflective or non-reflective surface to the audience floor).

The measured mean mid-frequency values of investigated room-acoustic indicators are less or close to the acceptable values**. Thus, considering the hall volume, the acoustical conditions of the hall for ensemble and orchestral music is judged as reasonably good. A detailed description of the objective indicators and measurements upon which **Table 6.2** is based are presented in Appendix A-IV.

** See Chapter 2, Table 2.1 and 2.2, and Sections 2.2.2-2.2.6.

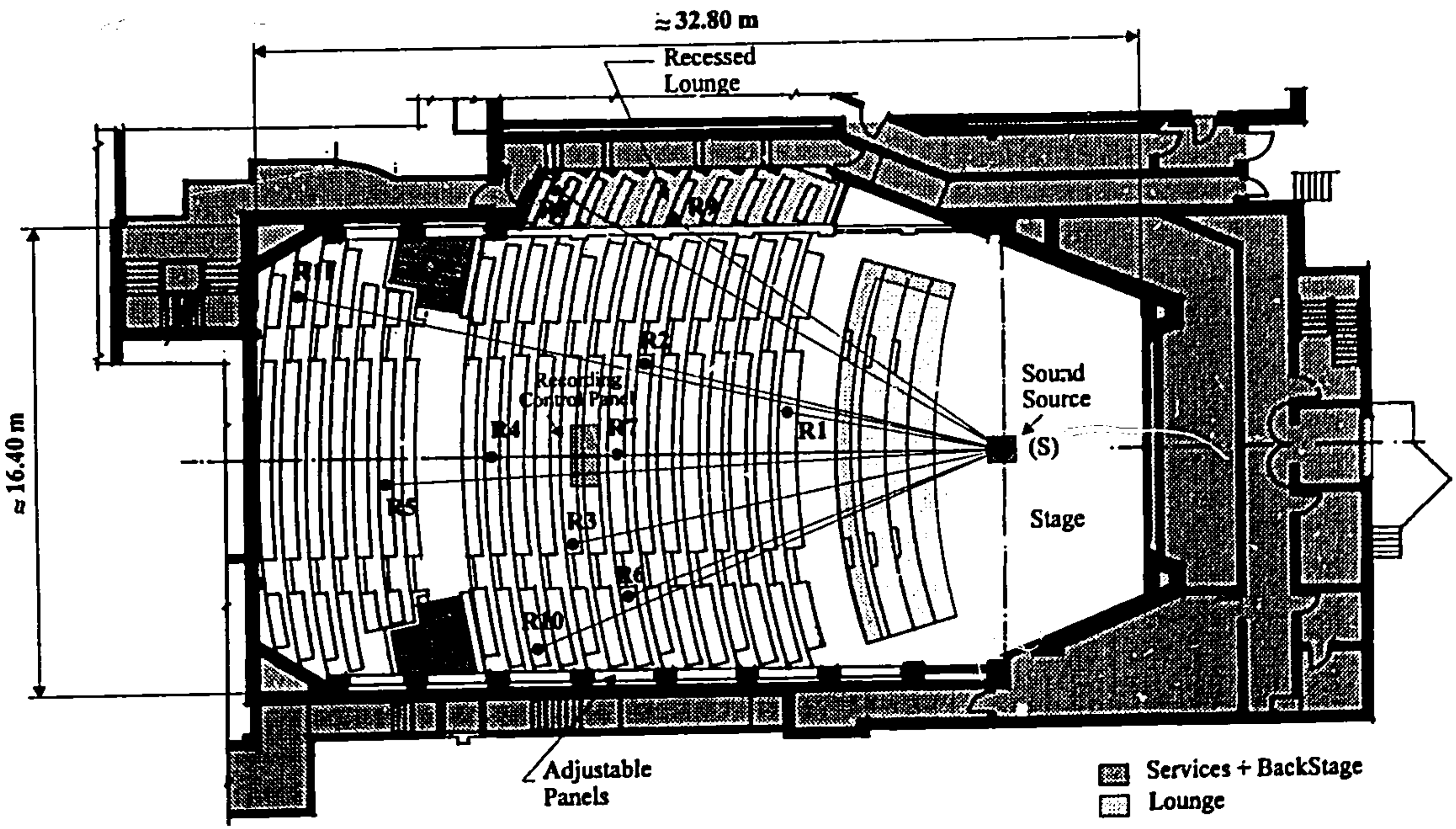


Fig. 6.1-a The **"LOYOLA"** concert hall, architectural plan showing the spatial and seating organization. The locations of the selected measurement and source positions together with their symbol references are also shown.

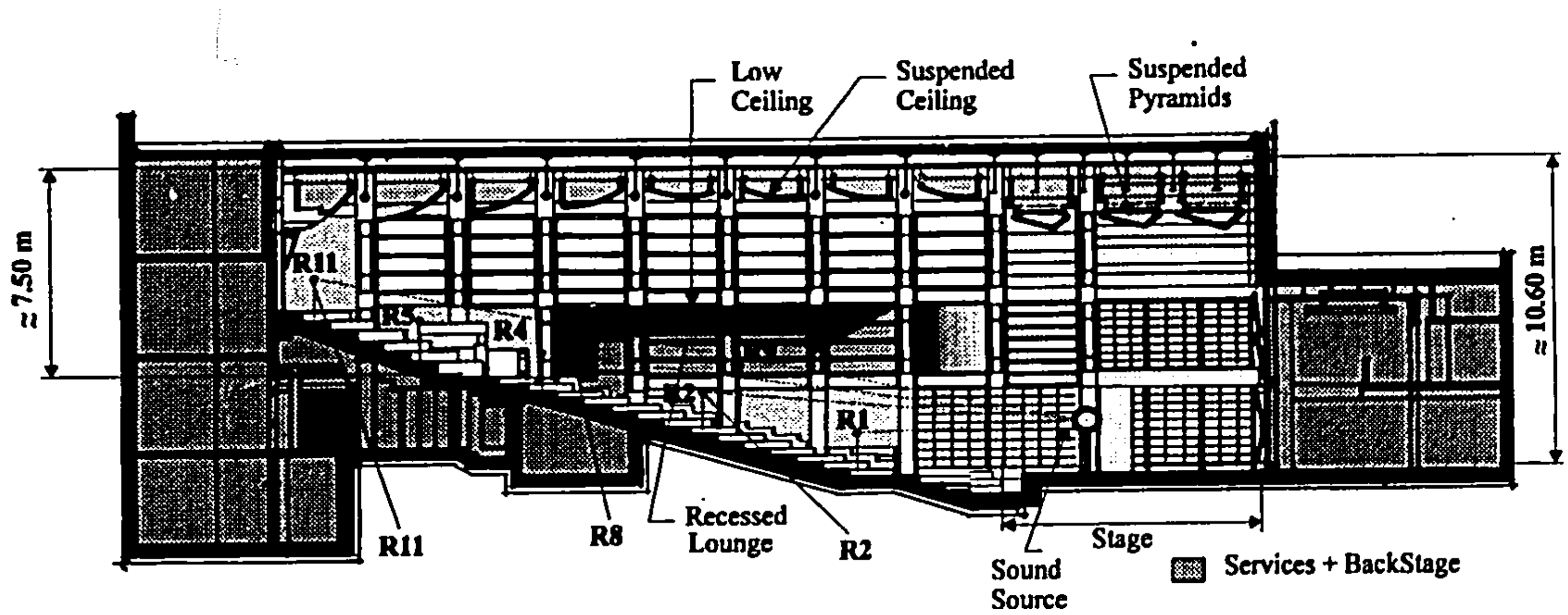
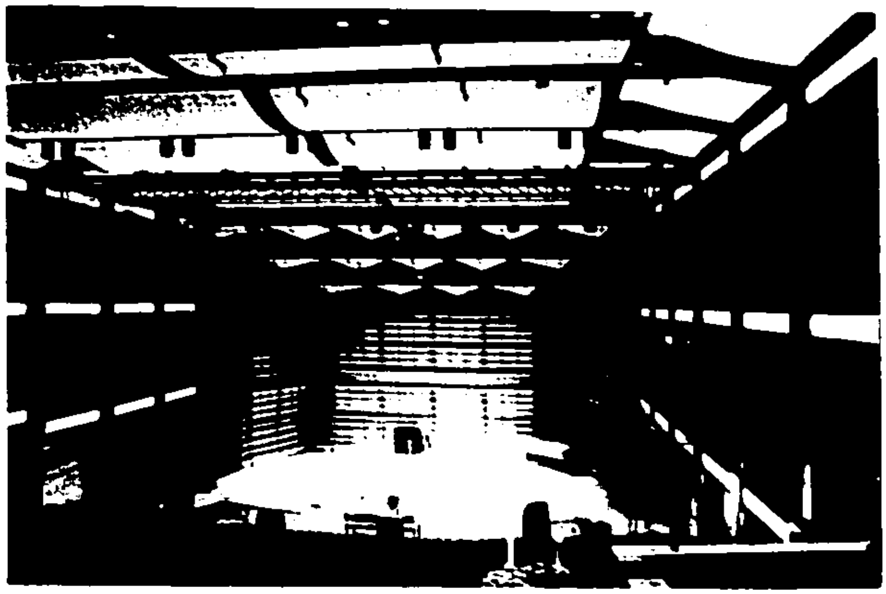


Fig. 6.1-b The **"LOYOLA"** concert hall, longitudinal section showing ceiling height, configuration, and interior design features.

(a)



(b)

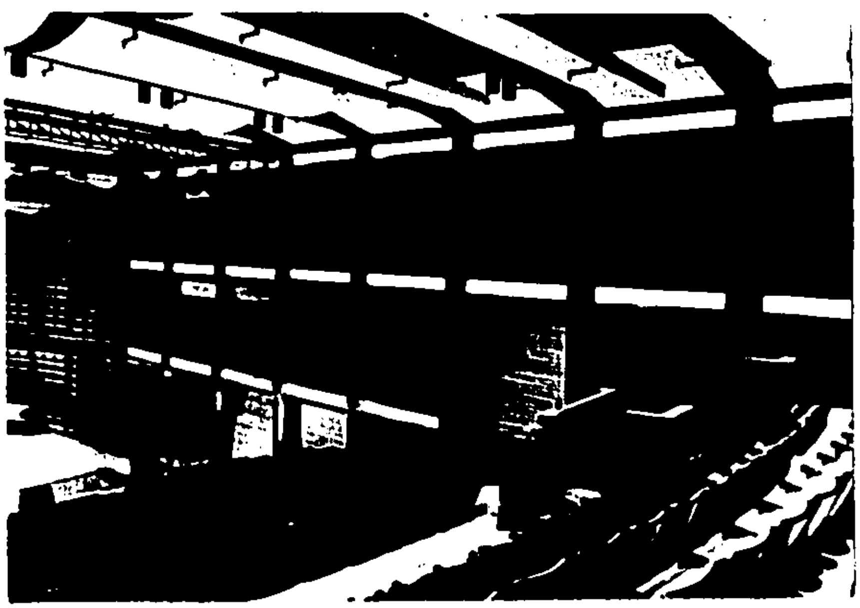


Fig. 6.2 The "LOYOLA" concert hall, interior views showing the architectural design features, looking at (a) The stage, and (b) The side wall adjustable panels.

(a)

(b)

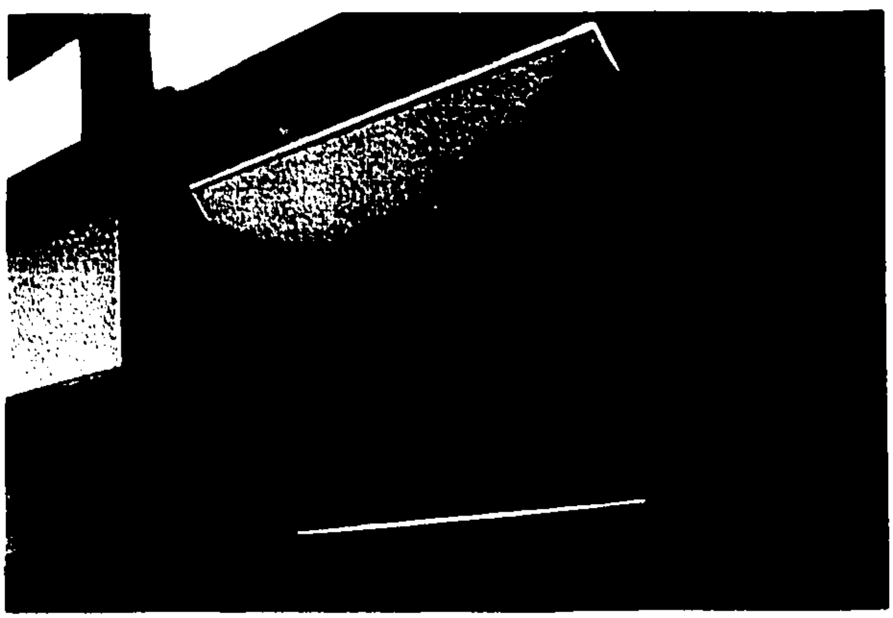


Fig. 6.3 The "LOYOLA" concert hall, interior views looking at (a) The rear seating zone, showing back wall treatment, and (b) The side wall adjustable panels detail.

TABLE 6.1 "LOYOLA" concert hall : Description of the selected receiver positions.

Seat Location Symbol	Distance from Source (S), m	Position Comments
R1	5.96	First seating row, the nearst to the sound source
R2	11.68	Middle seating zone, in the front of side recessed lounge
R3	14.35	Middle seating zone
R4	16.89	End of middle seating zone
R5	20.92	Middle of back seating zone
R6	12.93	Side seating area
R7	12.27	Centre of the middle seating zone in front of a recording control panel
R8	18.10	In the side recessed lounge, last seating row adjacent to lounge back wall
R9	14.17	In the side recessed lounge
R10	16.93	Middle seating area close to a side wall
R11	25.13	The farthest location, rear seating area, close to the back wall

TABLE 6.2 A summary of the contemporary single-number objective indicators evaluated for the "LOYOLA" concert hall , side wall reflective panels "ON". (Space average values)

Position Values

STD

Space Average

			Room -Acoustic Indicators								
			EDT	RT	D_{50}	R	C_{80}	T_s	G	LEF	SNR
		FREQ., Hz	Sec.	Sec.	Ratio	dB	dB	Sec.	dB	Ratio	dB
	1	63	0.80	1.30	0.60	-1.8	6.90	0.060	2.20	0.10	
	2	125	1.03	1.36	0.41	1.6	1.78	0.093	3.97	0.13	
	3	250	1.24	1.43	0.29	3.9	-0.74	0.117	7.49	0.15	
LOW-FREQ.	●	Mean	1.02	1.36	0.43	1.2	2.65	0.090	4.55	0.13	
	4	500	1.22	1.16	0.34	2.9	0.35	0.106	8.58	0.16	0.37
	5	1000	1.14	1.16	0.39	1.9	1.43	0.098	6.97	0.20	1.27
MID-FREQ.	●	Mean	1.18	1.16	0.37	2.4	0.89	0.102	7.78	0.18	1.64
	6	2000	1.19	1.07	0.44	1.0	1.80	0.091	6.46	0.24	2.03
	7	4000	1.16	1.11	0.43	1.2	1.93	0.089	2.76	0.26	
	8	8000	—	—	—	—	—	—	—	—	
HIGH-FREQ.	●	Mean	1.18	1.14	0.44	1.1	1.87	0.09	4.61	0.25	
				Value		%AL		Rating			%I.S.
	●	STI		0.57		7.8		FAIR			95
	●	RASTI		0.56		8.2		FAIR			

Tonal Colour	1.05	Bass Ratio	0.85	Treble Ratio	0.71	

Note: % I . S. = % Intelligible Syllables

6.2.2 "*LOYOLA*" Chapel : The Space Description

The "*LOYOLA*" chapel enclosure is described in terms of the following aspects:

- ***Chapel shape*** : Classical box-shaped hall having a high barrel vault ceiling. A back balcony exists where an organ is installed.
- ***Dimensions*** : 16.5 (width) x 28.5 (length) x 12.50 (maximum, height).
- ***Volume*** : Approximately, 6600 m^3 .
- ***Capacity*** : Approximately, 200-225 seats (main hall) + 30-40 seats (balcony).
- ***Furniture*** : Pews and kneeling benches made of polished hardwood on a horizontal floor and arranged parallel to side walls as shown in Fig. 6.4(a)
- ***Special features*** :
  - The chapel ceiling is a parallel high barrel vault (see Fig. 6.4(b)).
  - Side walls and ceiling surfaces are covered with polished wooden strip panels.

Fig. 6.4(a) and (b) show the chapel's architectural plan, longitudinal and traverse sections with approximate dimensions. Fig. 6.5 shows interior perspectives views illustrating the chapel architectural design features.

Since the chapel enclosure design is symmetric, eight measurements locations have been selected on one side of the space with one location in the balcony.

These seat locations cover different congregation seating zones and are shown in Fig 6.4(a); each is indicated by a numbered letters **(P)** and **(B)** according to the measurement order and will be subsequently used to identify measurement results in the following text and figures. A summary of the selected measurement positions together with their overall distances from the sound source and related comments is provided in **Table 6.3**.

The mid-frequency space average of the measured objective room-acoustic indicators as shown in **Table 6.4** are found to approach acceptable values and in consequence the acoustical conditions of the chapel for both speech and ensemble music can be judged good when the chapel is empty or occupancy is low. However the acoustical quality is expected to degrade to some extent when the chapel is fully occupied. Table 6.4 displays a summary of the contemporary single-number objective indicators evaluated for the "***LOYOLA***" chapel. A detailed analysis is presented in Appendix A-VI.

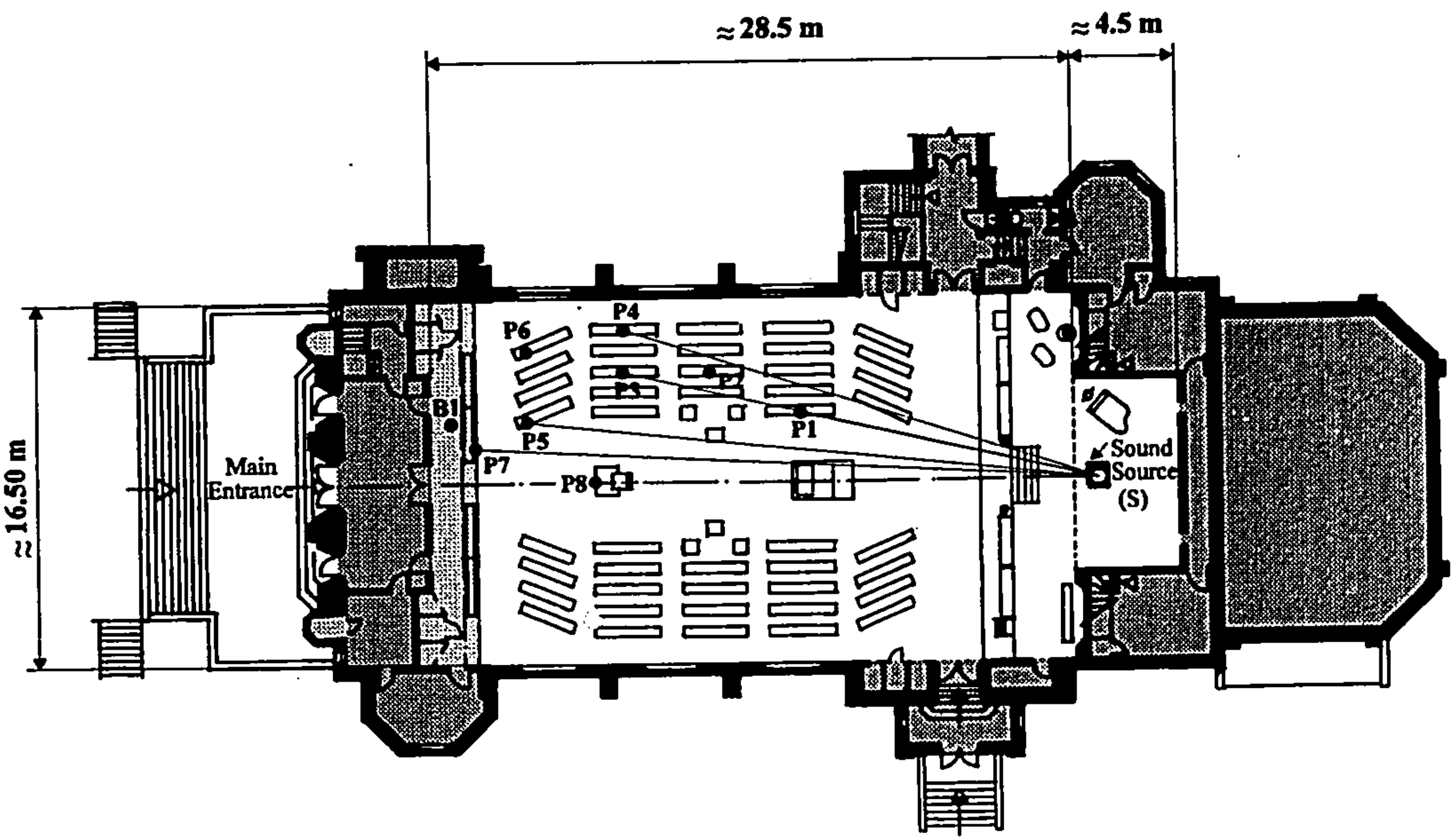


Fig. 6.4-a The "**LOYOLA**" chapel, architectural plan showing the spatial and seating organization. The locations of the selected measurement and source positions together with their symbol references are also shown.

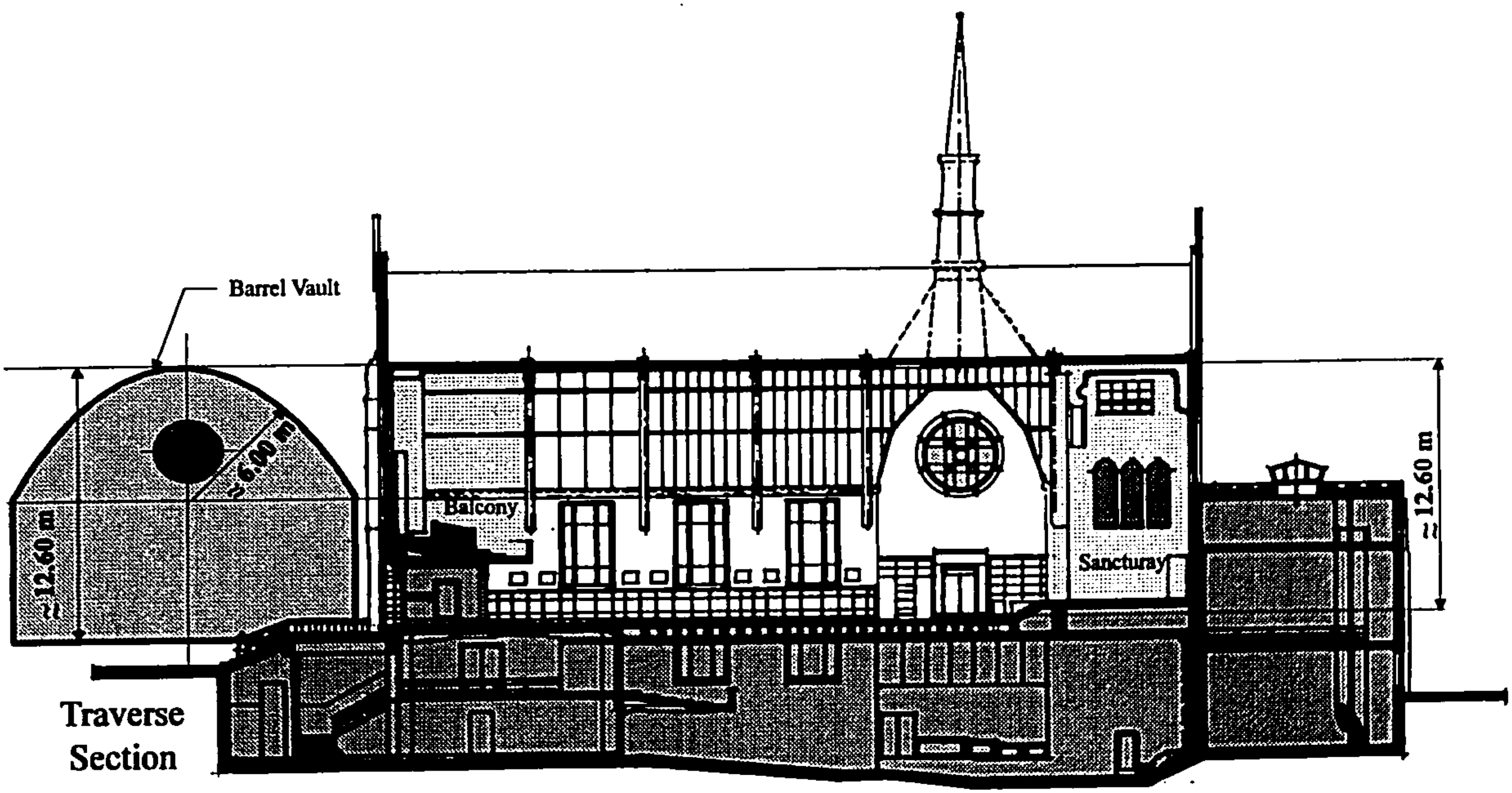
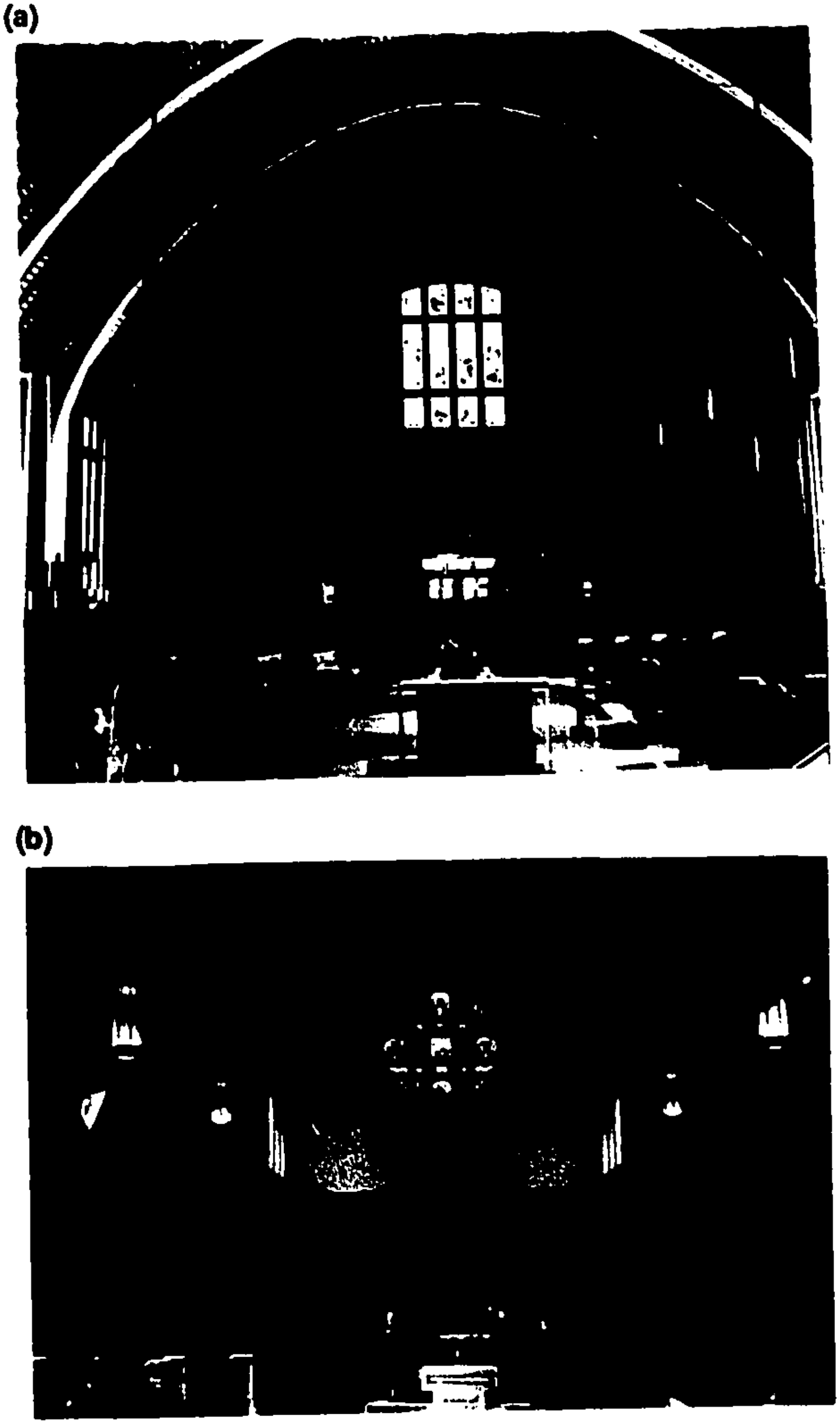


Fig. 6.4-b The "**LOYOLA**" chapel, longitudinal and traverse sections showing ceiling height, configuration, and interior design features.

(a)



(b)

Fig. 6.5 The "LOYOLA" chapel, interior views (a) Looking at the balcony, showing the barrel vault configuration and surrounding interior design features, (b) Looking at the sancturay.

TABLE 6.3 "LOYOLA" chapel : Description of the selected receiver positions.

Seat Location Symbol	Distance from Source (S), m	Position Comments
P1	10.93	Front congregation seating, nearest to the sound source
P2	14.86	Near a side Wall, middle seating zone
P3	18.37	Middle seating zone
P4	18.08	Middle seating zone
P5	21.07	Back seating zone,near a side wall
P6	22.64	Located in the far corner of the chapel space
P7	25.53	Rear seating zone, under the balcony edge
P8	20.69	Centre line of the chapel hall, a speaker bench
B1	26.91	In the balcony, first seating row

TABLE 6.4 A summary of the contemporary single-number objective indicators evaluated for the "LOYOLA" chapel. (Space average values)

Position Values
STD
Space Average

		Room -Acoustic Indicators								
		EDT	RT	D_{50}	R	C_{80}	T_s	G	LEF	SNR_{sa}
	FREQ., Hz	Sec.	Sec.	Ratio	dB	dB	Sec.	dB	Ratio	dB
1	63	1.96	2.78	0.42	1.40	1.46	0.150	2.20	0.20	
2	125	1.66	2.68	0.48	0.35	2.07	0.122	5.55	0.26	
3	250	1.93	2.25	0.24	5.00	-2.23	0.172	8.73	0.25	
LOW-FREQ. ●	Mean	1.85	2.57	0.38	2.13	0.44	0.148	5.49	0.24	
4	500	2.10	2.24	0.22	5.50	-3.31	0.189	9.22	0.22	-3.77
5	1000	2.11	2.15	0.23	5.25	-2.40	0.182	6.24	0.20	-2.90
MID-FREQ. ●	Mean	2.11	2.20	0.23	5.37	-2.86	0.186	7.73	0.21	-3.34
6	2000	1.74	1.75	0.31	3.47	-0.22	0.132	5.07	0.19	-0.53
7	4000	1.25	1.25	0.38	2.12	1.25	0.098	-1.53	0.15	
8	8000	—	—	—	—	—	—	—	—	
HIGH-FREQ. ●	Mean	1.49	1.49	0.35	2.78	0.52	0.115	1.77	0.17	
			Value		%AL		Rating			%I.S.
●	STI		0.48		12.7		FAIR			92
●	RASTI		0.45		14.9		FAIR			

Tonal Colour	1.05	Bass Ratio	0.85	Treble Ratio	0.71	

Note: % I. S. = % Intelligible Syllables

6.2.3 "*ST. LOUIS DE FRANCE*" Church : The Church Description

Room-acoustic measurements have been undertaken to characterize the acoustical properties of "*ST. LOUIS DE FRANCE*" church, in Montréal (constructed in 1935). The opportunity was taken to test the effectiveness of the electroacoustical sound system installed to improve the congregation's speech intelligibility both for normal-hearing and hearing-impaired persons in the presence of the church's long reverberation time which is in direct conflict with satisfactory speech perception. The church may be described as follows:

- ***Church shape*** : Classical cross-shaped hall composed of main central nave with high pointed-vault ceiling and two narrow side aisles with low flat ceiling. A back balcony exists where an organ is installed.
- ***Dimensions*** : 22.8 m (width) x 59.0 m (length) x 22.4 m (maximum height).
- ***Volume*** : Approximately, 27,500 - 30,000 m^3.
- ***Capacity*** : Approximately, 800-850 seats (main hall) + 40 seats (balcony).
- ***Furniture*** : Pews and kneeling benches made of polished hardwood on a horizontal floor.
- ***Special features*** : - The central nave ceiling is a parallel high pointed vault (Fig. 6.6(b)).

- Side walls and ceiling surfaces are covered with a layer of white-painted cork panels of 2.5 cm thickness.
- The church is equipped with an electroacoustical sound-reinforcement system with a total of 6 distributed linear-array loudspeakers, (i.e. loudspeaker columns, see Fig. 6.6(b)).

Fig. 6.6(a) and (b) show the church's architectural plan and longitudinal section with approximate dimensions. Fig. 6.7 and Fig. 6.8 show perspectives of the interior architectural design features.

Eight measurement locations have been selected on one side of the church's centre axis of symmetry. These seat locations cover different congregation seating zones and are shown in Fig. 6.6(a); each is indicated by a numerated letter (**L**) according to the measurement order and will be subsequently used to identify measurement results. A summary of the selected measurements positions together with their overall distances from the sound source and related comments is provided in **Table 6.5**.

The mean mid-frequency measured room-acoustic indicators, **Table 6.6**, are found to deviate from the optimal acceptable values and in consequence the acoustical conditions of the church for both speech and ensemble music are judged to be unsatisfactory when the church is empty and/or occupancy is low. However the acoustical quality is expected to improve to the fair rating when the church is fully occupied. **Table 6.6** displays a summary of the contemporary single-number objective indicators evaluated for the "***ST. LOUIS DE FRANCE*** " church. A detailed analysis is given in Appendix A-VI.

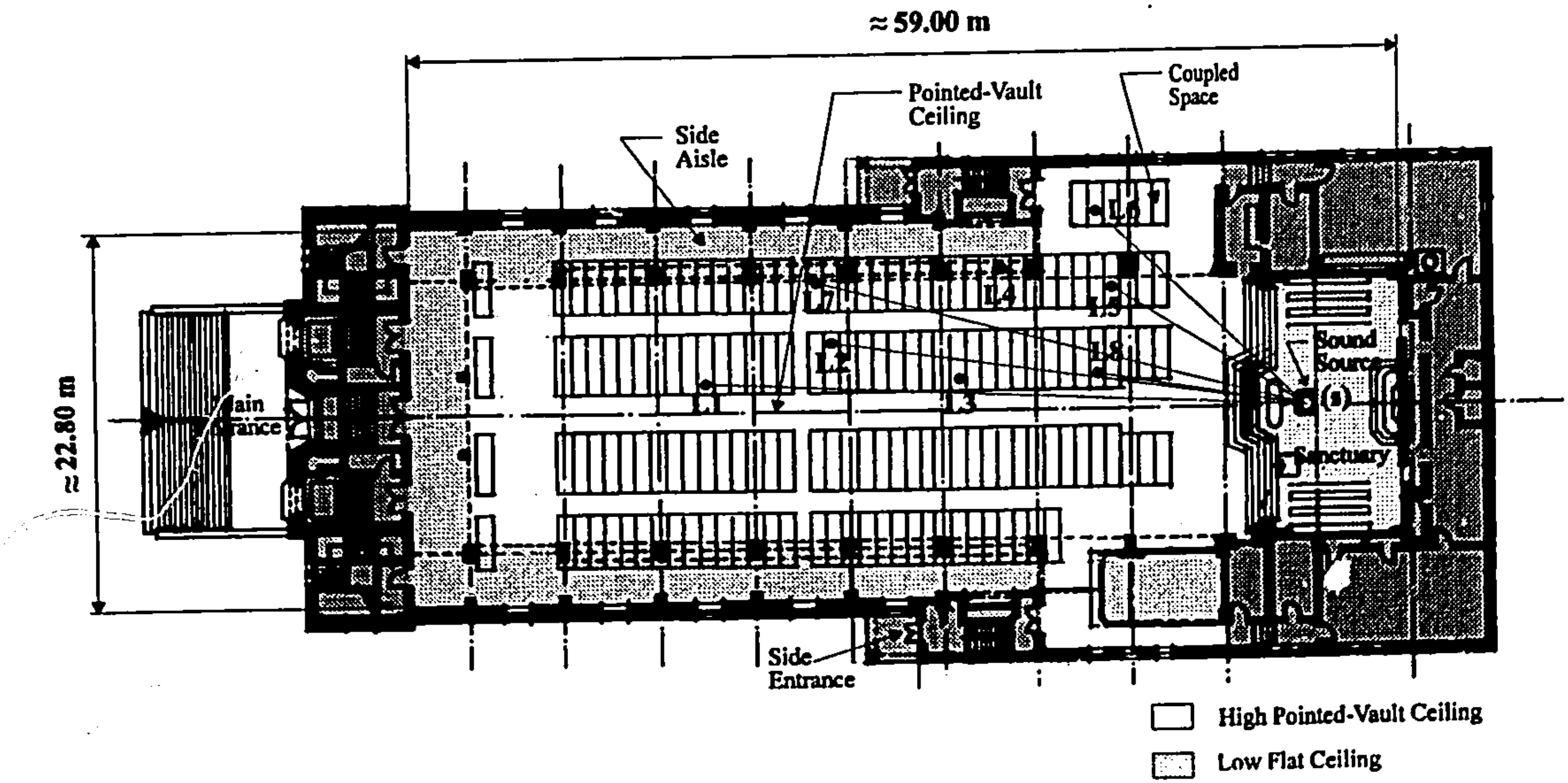


Fig. 6.6-a The "**ST. LOUIS DE FRANCE**" church, architectural plan showing the spatial organization. The locations of the selected measurement and source positions together with their symbol references are also shown.

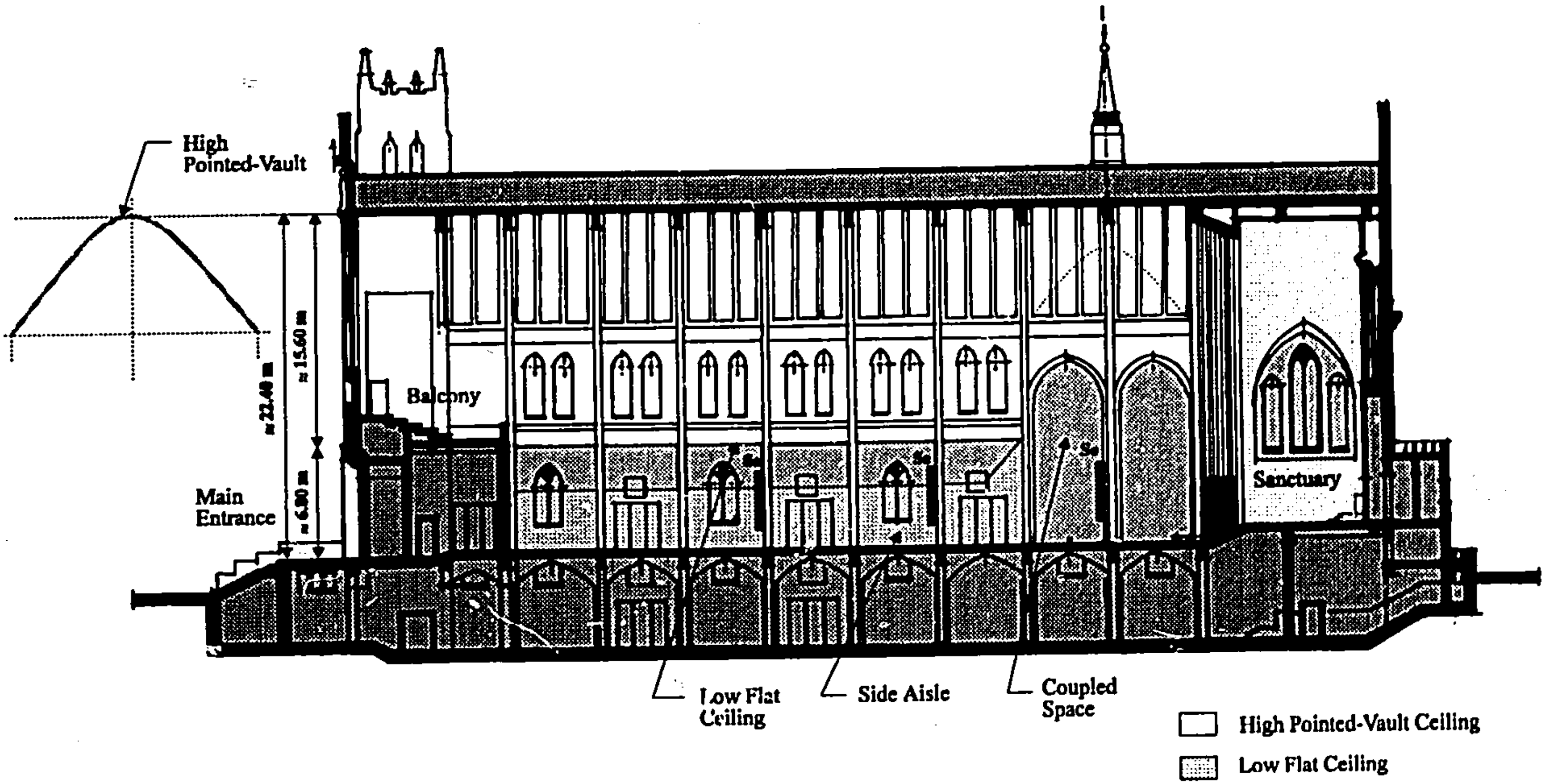


Fig. 6.6-b The **"ST. LOUIS DE FRANCE"** church, longitudinal and traverse sections showing ceiling height, configuration, and locations of line-array loudspeakers of the electroacoustical sound-reinforcement system.

Fig. 6.7 The "ST. LOUIS DE FRANCE" church, interior view looking at the sanctuary, showing the pointed-vault configuration. and locations of the sound system loudspeakers.

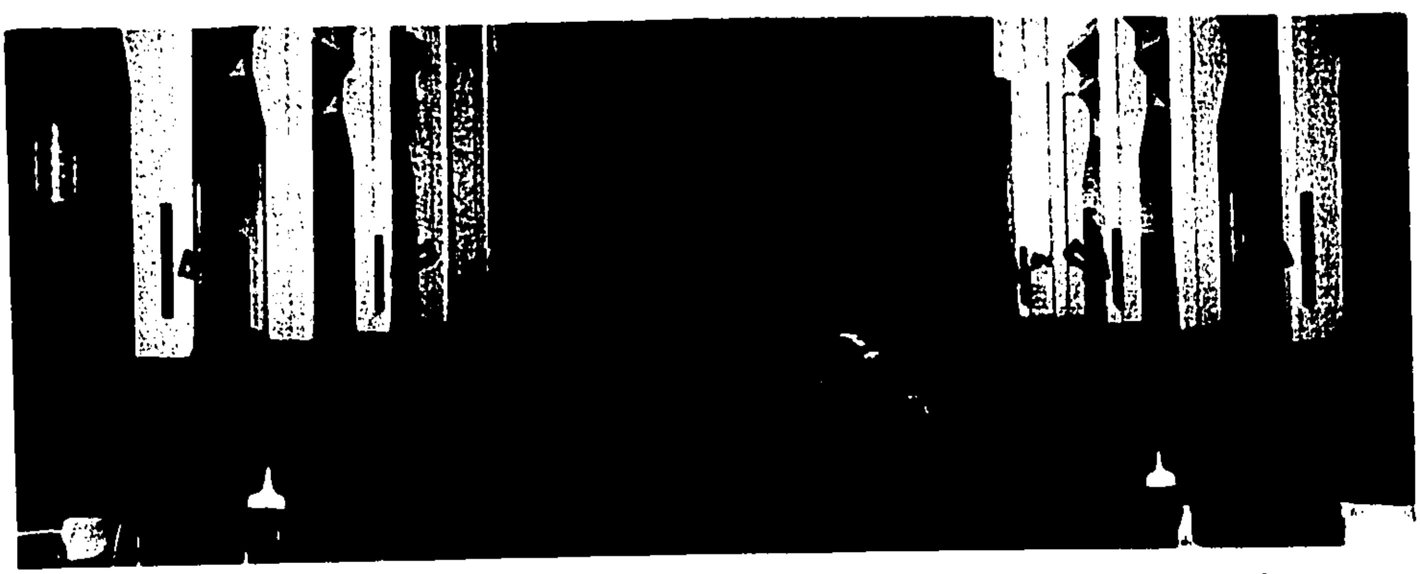


Fig. 6.8 The "ST. LOUIS DE FRANCE" church, interior view looking at the sanctuary, showing main central nave and the two side aisles, with low flat ceiling. The locations of the sound system loudspeakers are also shown (indicated by the arrows).

TABLE 6.5 "ST. LOUIS DE FRANCE" church : Description of the selected receiver positions.

Seat Location Symbol	Distance from Source (S), m	Position Comments
L1	32.48	Central nave under barrel vault, rear congregation seating
L2	25.77	Central nave, middle seating zone
L3	18.54	Central nave, middle seating zone
L4	17.89	Under low flat ceiling of the side aisle, obstructed by a column
L5	11.63	Front seating zone, the nearest location to the sanctuary floor area
L6	15.40	Located in "coupled" space with a high traverse vault-shaped ceiling
L7	27.48	Rear seating zone, under a side aisle
L8	13.30	Near centre line of main nave, second location nearest to sound source

TABLE 6.6 A summary of the contemporary single-number objective indicators evaluated for the "ST. LOUIS DE FRANCE" church. (Space average values)

Position Values
STD
Space Average

			Room -Acoustic Indicators								
			EDT	RT	D_{50}	R	C_{80}	T_s	G	LEF_s	SNR_{50}
		FREQ. Hz	Sec.	Sec.	Ratio	dB	dB	Sec.	dB	Ratio	dB
	1	63	2.67	2.87	0.37	2.3	-0.21	0.169	-0.16	0.05	[illegible]
	2	125	2.40	2.69	0.38	2.1	-0.25	0.158	1.83	0.15	[illegible]
	3	250	3.58	4.01	0.10	9.5	-12.4	0.322	6.99	0.18	[illegible]
LOW-FREQ.	●	Mean	2.88	3.19	0.28	4.0	-4.29	0.216	2.88	0.13	
	4	500	4.21	4.69	0.12	8.7	-10.9	0.398	6.89	0.15	-11.8
	5	1000	4.61	4.99	0.10	9.5	-9.83	0.416	5.62	0.22	-10.5
MID-FREQ.	●	Mean	4.41	4.84	0.11	9.1	-10.4	0.407	6.26	0.19	-11.2
	6	2000	3.73	3.86	0.14	7.9	-6.92	0.314	3.68	0.20	-7.21
	7	4000	2.28	2.52	0.24	5.0	-3.52	0.185	-3.56	0.23	[illegible]
	8	8000	—	—	—	—	—	—	—	—	[illegible]
HIGH-FREQ.	●	Mean	3.00	3.19	0.19	6.3	5.22	0.249	0.06	0.22	[illegible]
				Value		%ALcons		Rating			%I.S.
	●	STI		0.36		24.2		POOR			70
	●	RASTI		0.30		33.6		POOR			

Tonal Colour	1.05	Bass Ratio	0.85	Treble Ratio	0.71	

Note: % I. S. = % Intelligible Syllables

6.3 SPATIAL INFORMATION OF SOUND FIELDS : MEASUREMENT RESULTS AND ANALYSIS

6.3.1 Levels of Sound Quality and Diagnostic Assessment

Obtaining spatial information of the sound field significantly increases the available data for further analysis. A break down of the process of sound quality assessment or acoustics problem investigation should help in handling and productively employing this data as opposed to overwhelming the user. Contemporary room-acoustic indicators can be considered the first level of available assessment. Single channel measurements of room impulse response would be the initial action followed by calculation of most subjectively relevant room-acoustic indicators. An example format of these indicators versus frequency bands usually of interest is shown in **Table 6.7** along with comment as to their use. Three sets of data can be obtained; position values, space-averaged values, and standard deviations of each indicator. Space average values of the "***LOYOLA***" concert hall with the side walls reflective panels "OFF" are shown as an example in **Table 6.7.** In practice, this information is used to characterize the hall or space under investigation by way of comparing the mid-frequency space-averaged results to widely accepted optimal values and then inferring a sound quality rating. Examining the single frequency band values (i.e. abnormal, discrepancies,..etc) of any one particular indicator or more might reveal an acoustical defect. Seat-to-seat variations, at this stage are also a beneficial avenue for defining sound quality homogeneity. Examination of position values would also help identifying unsatisfactory seating zones and

TABLE 6.7 Contemporary room -acoustic indicators. Example vaules (space average) are displayed for the "LOYOLA" concert hall with side wall reflective panels "OFF".

Level 1 Assessment

Position Values
STD
Space Average

	FREQ., Hz	EDT	RT	D_{50}	R	C_{80}	T_s	G	LEF	SNR
		Sec.	Sec.	Ratio	dB	dB	Sec.	dB	Ratio	dB
1	63	0.76	1.20	0.55	-0.9	3.3	0.096	1.30	0.05	
2	125	1.01	1.18	0.41	1.6	2.2	0.096	3.60	0.16	
3	250	1.16	1.32	0.29	3.9	-0.2	0.113	6.65	0.13	
LOW-FREQ. ●	Mean	0.97	1.23	0.41	1.5	1.8	0.093	3.90	0.11	
4	500	1.14	1.09	0.34	2.9	1.0	0.097	7.94	0.15	0.61
5	1000	1.06	1.09	0.39	1.9	2.1	0.091	6.24	0.17	1.64
MID-FREQ. ●	Mean	1.10	1.09	0.37	2.4	1.5	0.094	7.94	0.16	1.13
6	2000	1.13	1.03	0.44	1.0	2.20	0.085	5.50	0.18	2.31
7	4000	1.11	1.07	0.43	1.2	2.2	0.085	1.82	0.20	
8	8000	—	—	—	—	—	—	—	—	
HIGH-FREQ. ●	Mean	1.12	1.05	0.44	1.1	2.2	0.085	3.66	0.19	
			Value		% AL		Rating			% I. S.
●	STI		0.59		7.0		FAIR			96
●	RASTI		0.57		7.8		FAIR			

Tonal Colour	1.05	Bass Ratio	0.99	Treble Ratio	1.02

NOTE: % I. S. = % Intelligible Syllables

Use to characterize the hall, or space

Compare to widely accepted optimal values

Sound quality rating

Detect acoustical defects (if any)

NOTE: IACC can also be included , but Binaural measurements with a dummy head are required.

providing a first guess as to acoustical defects.

The more detailed levels of assessment start with measuring spatial information, however, this requires a measuring tool and methodology such as the 3-D transient sound intensity impulse technique pioneered here. A level 2 of assessment employs position results only and Table 6.8 displays both raw data and advanced single-number indicators to be calculated for further examination. **Table 6.8** displays the 3-D measurements at listener position **R11** in the concert hall. Position values excluding the direct sound and values for the first 80 msec result in two possible sets of data. Spatial information presented in directional energy ratios both in linear and dB scales can be used to detect directions of dominant received sound energy and check directional energy balance. Directional impulse responses may be also cross-correlated for dissimilarity quantification. New Room-acoustic indicators such as Front/Back (*F/B*) energy ratios can be correlated to other indicators in order to test its subjective relevance.

Based on the data of the stage 2 assessment, directivity can be viewed for visual examination if found necessary, either in linear-scaled polar plots (relative to maximum received sound) to highlight directional information, or in dB-scaled polar plots (for example the first 20 dB relative to the maximum received sound intensity) which are more relevant with respect to subjective response or to objective quantification for sound diffuseness.

TABLE 6.8 An example of directional sound energy ratios and Front/Back, (F/B) indicator vs frequency evaluated for position "R11" in the "LOYOLA" concert hall.

Level 2 Assessment

0-80 msec.

Position Values S-T (Total Energy)

	FREQ. Hz	Front	Back	Right	Left	Up	Down	$(F/B)_t$	$(F/B)_s$	$(F/B)_{80}$
		Ratio						dB	dB	dB
1	63	0.14	0.54	0.09	0.05	0.09	0.08	-5.6	-5.8	-1.4
2	125	0.32	0.07	0.28	0.11	0.16	0.06	6.5	6.5	2.8
3	250	0.43	0.13	0.14	0.05	0.18	0.05	5.1	5.1	8.2
LOW-FREQ. ●	Mean	0.30	0.25	0.17	0.07	0.14	0.06	2.0	1.9	3.2
4	500	0.32	0.18	0.24	0.08	0.14	0.04	3.0	2.6	2.8
5	1000	0.25	0.29	0.14	0.07	0.19	0.05	0.3	-0.5	-0.1
MID-FREQ. ●	Mean	0.29	0.24	0.19	0.08	0.17	0.05	1.7	1.1	1.4
6	2000	0.37	0.20	0.16	0.09	0.08	0.09	3.3	2.7	4.6
7	4000	0.31	0.19	0.15	0.15	0.10	0.09	2.8	2.1	3.5
8	8000	—	—	—	—	—	—	—	—	—
HIGH-FREQ. ●	Mean	0.34	0.20	0.16	0.12	0.09	0.09	3.1	2.4	4.1

Correlate to C_{80} or Other Indicators

Detect Dominant Sound Directions

Check energy balance

Check Right & Left Dissimilarity Cross-Correlation

NOTE : $(F/B)_t$ = Front/Back sound ratio, for the total impulse response.

$(F/B)_s$ = Front/Back sound ratio, excluding the direct sound.

$(F/B)_{80}$ = Front/Back sound ratio, for the first 80 msec.

A level 3 assessment might include examining and comparing the directional sound decay values for particular listener locations in order to check directional decay homogeneity or unexpected deviations. Plots of instantaneous directional decay curves could also be viewed. **Table 6.9** shows an example of a stage 3 data format for listener position **R11** in the "***LOYOLA***" concert hall.

A stage 4 assessment could involve quantifying sound diffuseness, initially one might investigate the time-segmented directional information presented in single-number indicators versus successive specified time windows (50 or 100 msec). Low, mid or high frequencies averages might then be a sufficient indication of direction-time transition. **Table 6.10** displays an example of a stage 4 result at position **R11** in the concert hall. Time-windowed directivity patterns may be necessary for visual examination. This stage of assessment may also include examining time arrival, magnitude and direction of discrete reflections. The isolation of sound in a particular solid angle of interest is possible. **Table 6.11** presents a summary of the four levels of assessment which might be needed for a comprehensive acoustic evaluation together with their areas of application. It should be remembered that prior to the present work just a level 1 type assessment was possible. The usefulness or the relative importance of each stage and related actions of analysis will be demonstrated in the following sections and later in the example applications.

TABLE 6.9 An example of sound intensity directional reverberation times vs frequency evaluated for listener position "R11" in the "LOYOLA" concert hall.

Level 3 Assessment

- ○ Space Average
- ○ STD
- ◉ Position Values

	Position Values	Directional Sound Intensity Decays			
		RTx	RTy	RTz	RTt
	FREQ., Hz	Sec.	Sec.	Sec.	Sec.
1	63	1.13	1.28	1.15	1.18
2	125	1.02	1.01	1.06	1.03
3	250	1.09	1.15	1.09	1.10
LOW-FREQ. ●	Mean	1.08	1.15	1.10	1.10
4	500	1.09	1.08	1.11	1.09
5	1000	1.08	1.08	1.08	1.08
MID-FREQ. ●	Mean	1.09	1.08	1.65	1.09
6	2000	1.10	1.08	1.09	1.09
7	4000	1.10	1.06	1.08	1.08
8	8000	—	—	—	—
HIGH-FREQ. ●	Mean	1.10	1.07	1.09	1.09

Compare

Use to check directional decay deviations

Plot directional decay curves if necessary

TABLE 6.10 An example of sound diffuseness quantification (D_A , D_θ) in successive time windows evaluated for a measurement at position " R11" in the "LOYOLA" concert hall.

Level 4 Assessment

		Position Values	Time-Segmented Directional Information								Notes
			0-50 ms		50-100		100-200		200-300		
		FREQ., Hz	D_A	D_θ	D_A	D_θ	D_A	D_θ	D_A	D_θ	
	1	63	5.4	0.24	3.5	0.15	3.4	0.80	1.2	0.60	
	2	125	4.6	0.16	3.9	0.41	4.1	0.76	2.8	0.45	
	3	250	4.8	0.37	3.9	0.31	3.3	0.76	2.2	0.55	
LOW-FREQ.	●	Mean	4.9	0.25	3.7	0.29	3.6	0.77	2.1	0.51	
	4	500	4.5	0.16	4.2	0.43	3.6	0.46	2.1	0.72	
	5	1000	4.1	0.24	3.8	0.31	2.4	0.72	1.3	0.76	
MID-FREQ.	●	Mean	4.3	0.20	4.0	0.37	3.0	0.59	1.7	0.74	
	6	2000	3.8	0.19	2.8	0.35	1.3	0.64	0.4	0.66	
	7	4000	4.3	0.30	3.8	0.30	2.6	0.62	1.4	0.64	
	8	8000	-	-	-	-	-	-	-	-	
HIGH-FREQ.	●	Mean	4.1	0.25	3.3	0.33	2.0	0.63	0.9	0.65	

Quantification of sound diffuseness

Examine direction-time transition

NOTE : D_A = Standard Deviation of sound intensity relative level within the specified time window.

D_θ = Calculated according to formulation (5.5) , page 5.12.

TABLE 6.11 A summary of the levels of assessment, actions, type of analysis, and expected end results needed for a comprehensive acoustics evaluation.

Scope	Level	Category	Content	Sets of Data
STATE OF THE ART	Level 1 Assessment	ACTION	● Measure Room Impulse Response (Single Channel)	SETS of DATA
		TYPE OF ANALYSIS	● Calculate Contemporary Room-Acoustic Indicators	● Location Values ● Space-Averaged Values ● Spatial Variation (STD)
			● Examine Impulse Responses for Echo Existence (Echo Criteria) ● Check Abnormal Values (Outside Widely Accepted Optimal Ranges) ● Examine Seat-to-Seat Variation (STD) (If necessary)	
		END RESULT	● Characterize the Sound Quality ● Compare the Space Sound Quality with Similar Spaces ● First indication of Acoustical Defects (If any)	
NEW AVENUES FOR DETAILED SOUND QUALITY ASSESSMENT AND DIAGNOSTICS	Level 2 Assessment	ACTION	● Measure 3-D Transient Sound Intensity Impulses (Dual Channel)	SETS of DATA
		TYPE OF ANALYSIS	● Calculate Spatial Information of the Sound Field	Location Values
			● Calculate and Examine Directional Energy Ratios vs Frequency ● Check Front/Back Energy Indicator ● Define Dominant Sound Directions ● Check Right & Left Dissimilarity ● Check Energy Balance	
	Level 3 Assessment	TYPE OF ANALYSIS	● Examine Linear-Scale Directional Information in relevant Planes (Horizontal, Lateral and Median) ● Highlight Dominant Sound Energy Directions ● Switch to dB-Scale Polar Plots to Examine Directivity Patterns ● Determine compliance with Directional Subjective Preference	
			● Examine Directional Reverberation Times	● Location Values ● Space-Averaged Values
			● Check Directional RT Variations ● Examine Directional Decay Curves (If necessary)	
	Level 4 Assessment	TYPE OF ANALYSIS	● Examine Directional Information versus Arrival Time (Time Domain)	● Arrival Time ● Direction Angles ● Relative Strength
			● Isolate Spatial Information of Sound for a Particular direction or Solid Angle	
			● Quantify Sound Diffuseness	● vs Time ● Time-Segmented ● Uniform Distribution
		END RESULT	● Cause and Effect Appreciation ● Detailed Diagnostics ● Quantification of the Sound Field Diffuseness	

6.3.2 Sound Intensity Decay Curves

Contemporary room-acoustics indicators allow an objective assessment of acoustical conditions in an enclosure to be made. Since the development of these indicators they have been used extensively to evaluate halls[10-13,18-21,23,152] with the dual purpose of interpreting their values and establishing conditions giving rise to good acoustical quality. Typically, the acoustic indicators are based on simple energy ratios approximated by the square of the impulse response sound pressure or the decay rates of sound pressure level.

While sound pressure has been considered the basic measure of what we hear, sound intensity is the actual measure of propagated sound power. In early room-acoustics evaluations sound energy was difficult if not impossible to measure, therefore sound pressure level decay records were used to infer quality parameters such as reverberation time. However, the theory of sound decay in enclosures always describes the decay as that of acoustic energy density (E) and not of the sound pressure. Stanzial *et al.*[153] experimentally investigated reverberation time evaluated using sound pressure and sound intensity and reported discrepancies. Reverberation time can be expressed by $T_s = \tau \ln (E_o/E_s)$ [3] where E_o is the initial energy density (i.e. energy per unit volume), E_s is the energy density at the threshold of hearing, and (τ) is a time constant related to the energy in the decay process. Simple addition is possible for energy, however pressure addition depends on the phase angle relationships of the incident sound waves and their subsequent squaring may result in higher values than the added energies.

The following examples report further analysis of the measurements presented in section 6.2, in particular the "***LOYOLA***" concert hall. Reverberation Time (*RT*) is also calculated utilizing the component sound intensity impulses (*X-X*, *Y-Y* and *Z-Z*) following the procedure described in Chapter 5, section 5.6.4.

Typical examples of resulting component sound intensity decays at different octave band frequencies and measurement positions **R1**, and **R11** are shown in Fig. 6.9. The orthogonal directional intensity decays are shown along with the reverberation times RT_{ix}, RT_{iy} and RT_{iz} evaluated from the slope of the best fit straight line over the range from -5 to -25 dB (i.e. RT_{20}). As can be observed for example from Fig. 6.9(a-i), the intensity level decay of the component sound intensity results in curves which are not as smooth as the decay curves usually obtained from sound pressure impulse responses using for example a "*Schroeder*" backward integration method[26,97]. The sound intensity decay curves can be characterized by a stair profile with different step heights and lengths. The heights of the resulting steps vary as a result of the different walls' surface absorption coefficients and the dependence of such coefficients on the angle of incidence. The length of the steps is probably related to the different free path lengths experienced by the same or superimposed bundles of sound reflections and hence depend on both the room dimensions compared to the wave length of the frequency of interest and the directional distribution of the reflected sound.

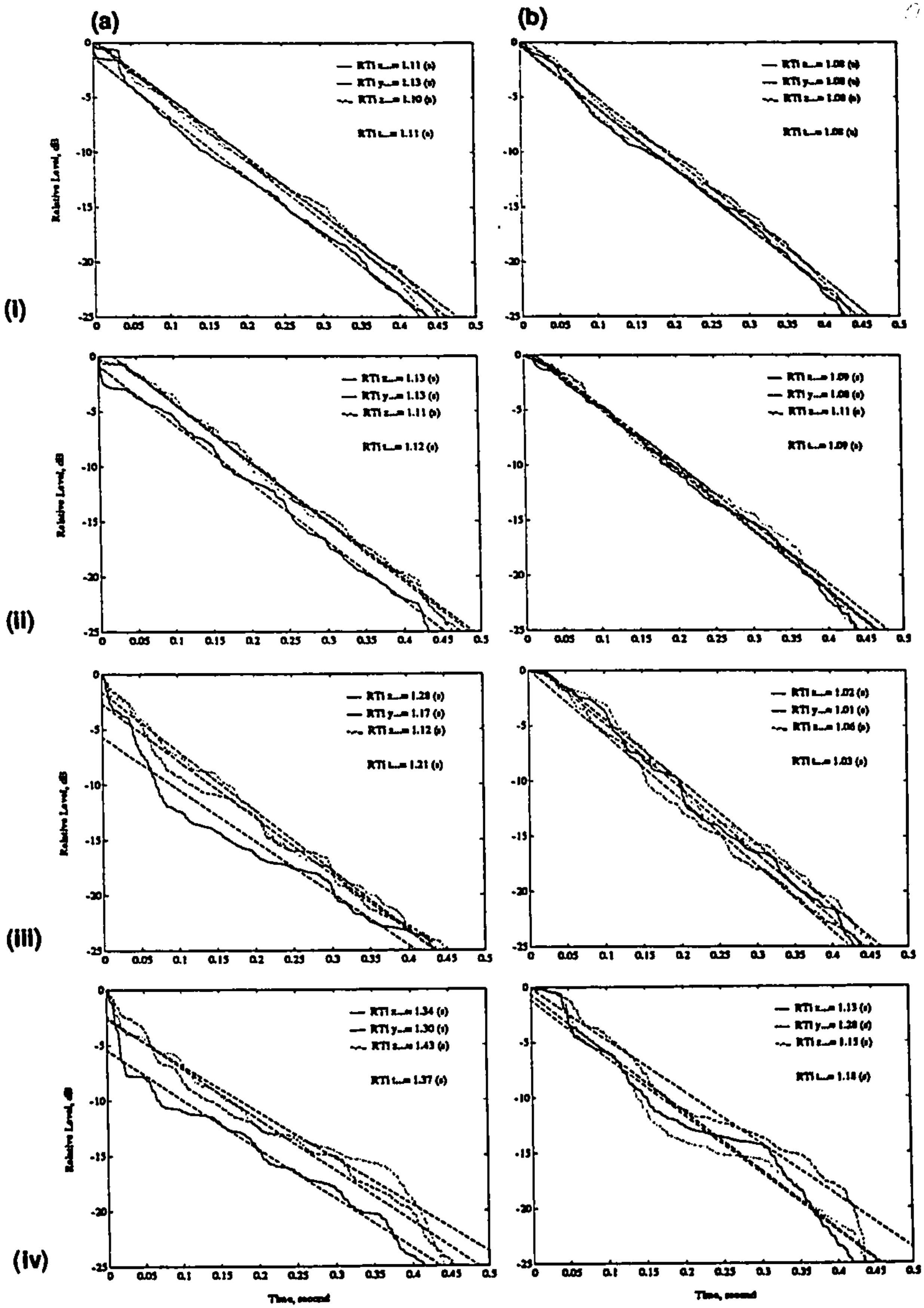


Fig. 6.9 Component sound intensity decay curves and reverberation times at listener positions, (a) **R1** and (b) **R11** at octave bands (i) 1 kHz, (ii) 500 Hz, (iii) 125 Hz, and (iv) 63 Hz, — X-X direction, --- Y-Y direction, -.-. Z-Z direction, and .... Regression line (from -5 to -25 dB).

The orthogonal components of sound intensity decay are useful as an indication of sound field diffusion. Diffusion if viewed as sound being isotropic in all directions and surface absorptions being uniformly distributed, means that the sound decay in all directions should exhibit the same decay rate. Deviations of a directional decay component implies lack of sound homogeneity, or, dominance of the acoustic energy in one direction. This can be verified by examining the decay curves of sound intensity components at different frequencies and positions. By comparing intensity decay curves at position **R1** (Fig. 6.9(a)) in the front seats of the "*LOYOLA*" concert hall to those at position **R11** (Fig. 6.9(b)) at the rear of the hall. At **R1** the instantaneous intensity component decay shapes are not similar and component *RT's* also deviate from each other. Directional decay abnormalities such as sagging in one direction or double-sloped decays in the other directions are also evident. The situation is quite different at position **R11**, where component decays are similar to each other and RT's almost identical. One can conclude that the sound at location **R1** is not as diffuse as at locations **R11**; although it is located 5.90 metre from tho sound source. This distance is greater than the "reverberation distance" rule of thumb (i.e. 3.6 m according to the $R_h = 0.057 \cdot \sqrt{(V.Q/RT)}$, see Chapter 5), and therefore it is considered diffuse. It should also be observed, that with decreasing frequency (e.g. 63 and 125 Hz) the decay steps' lengths increase, this probably occurs as the ratio between the wave length to the room dimension becomes smaller. The same trend is observed for measurements conducted for example at positions **P1** in the front seats and **B1** which is located in the rear balcony in the "***LOYOLA***" chapel.

6.3.3 Sound Quality Evaluation via Intensity-Based RT

The similarities and differences of reverberation time evaluated versus frequency using both sound pressure level decays (RT_p) and intensity (RT_I) are now of interest. Fig. 6.10 compares the spatial average of reverberation times (RT_p) determined from measurements at nine positions in the concert hall (Fig. 6.10) as calculated from the pressure impulse responses compared to corresponding values of (RT_I) utilizing sound intensity component impulses in the same decay range (i.e. RT_{20}) at the same locations. "*Schroeder*" backward integration is used in both, but applied to the absolute value of the intensity impulse to yield a comparative total energy assessment. Also depicted is the spatial average of Early Decay Times (EDT_p) (i.e. the slope of the best fit straight line over the range from 0 to -10 dB). It can be seen at low frequencies (125-250 Hz) that RT_p values diverge from both RT_I and EDT_p; over the full frequency range RT_I and EDT_p are almost similar and identical in trend.

The percentage of the absolute mean differences between RT_p and RT_I in the low frequency range is about 11% while it is 8% at mid-frequencies and around 4% at high frequencies. The difference at low frequencies can be attributed to the fact that at these frequencies, the resulting pressure of the received sound waves sensed by the microphone is more sensitive to phase relationships which then lead to higher processed values upon squaring. The absolute mean difference between EDT_p and RT_I is found to be about 4% at middle octave band frequencies and

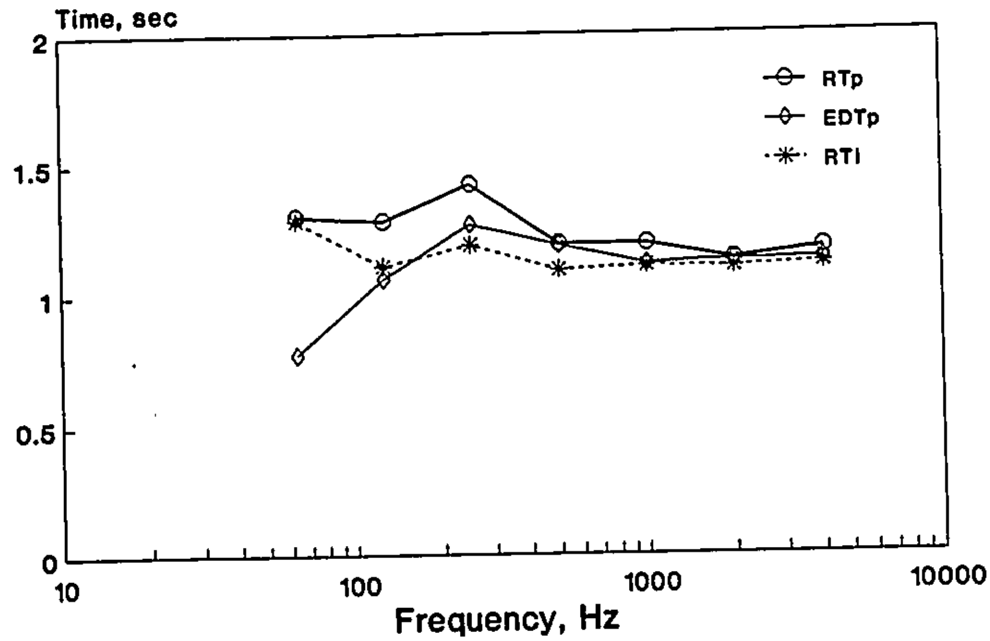


Fig. 6.10 Spatial average of Reverberation Time (RT_p), Early Decay Time (EDT_p) and (RT_l) evaluated for the "***LOYOLA***" concert hall.

approximately 2% at high frequencies (2, 4, and 8 kHz).

It can be suggested that to establish the acoustical characteristics of a room in terms of boundary surface absorption or reflective properties one must utilize the reverberation time observed from the decay of sound energy expressed by acoustic energy density or sound intensity. Reverberation time based on sound intensity measurements is also found to be similar to the early decay time particulary in the mid- and high-frequency range and it is interesting to note that EDT_p is reported to be subjectively more relevant than conventional reverberation time[37]. It should be noted that at high frequencies in all measurements, RT_p, EDT_p and RT_i converge to similar values, as might be expected.

6.3.4 Directional Information for Diagnostic Purposes

Since the development of newer room-acoustics indicators and with available measurement data from a large number of halls, there have been several attempts to correlate them to the overall geometric variables such as room width, height and side wall angles. Empirical predictive models had been developed by Gade[30,31] based on statistical analyses. Siebein *et al.*[60] and others[154,155] have reported investigations into statistical relations among the acoustical measures with the architectural features of the room but with more details included than earlier studies; however such models are only of use in the early design stage.

When an acoustical deficiency is encountered somewhere in a hall, the cause is not readily known or what should be changed, or to what extent before the required objective indicators are achieved. Therefore, visualizing the directional characteristics of the sound field and their magnitude at such locations will not only identify possible causes but also help to interpret the values of measured indicators in a more reliable manner.

To indicate such potential, examples of directivity patterns in actual sound fields are investigated. It should be noted that in the following figures sound field directivity will be shown either in linear-scaled polar plots (relative to maximum received sound) to highlight directional information, or in dB-scaled polar plots (i.e. the first 20 dB relative to the maximum received sound intensity) which are more relevant with respect to subjective response or to objectively quantify sound diffuseness.

Fig. 6.11 shows the directions of received sound at location **R2** in the "***LOYOLA***" concert hall in the horizontal, and lateral planes; this seat was reported by experienced listeners to be acoustically unsatisfactory. Looking at the directivity patterns of the direct and subsequent reflected sounds at 4 kHz (Fig. 11(a)), it can be seen that strong sound reflections are received from the listener left direction (about +60° from straight ahead within a vertical angle of +20°). These reflections are directed towards the opening to the left (with respect to the listener) side recessed lounge and its low ceiling. Less strong side reflections at 2 kHz are

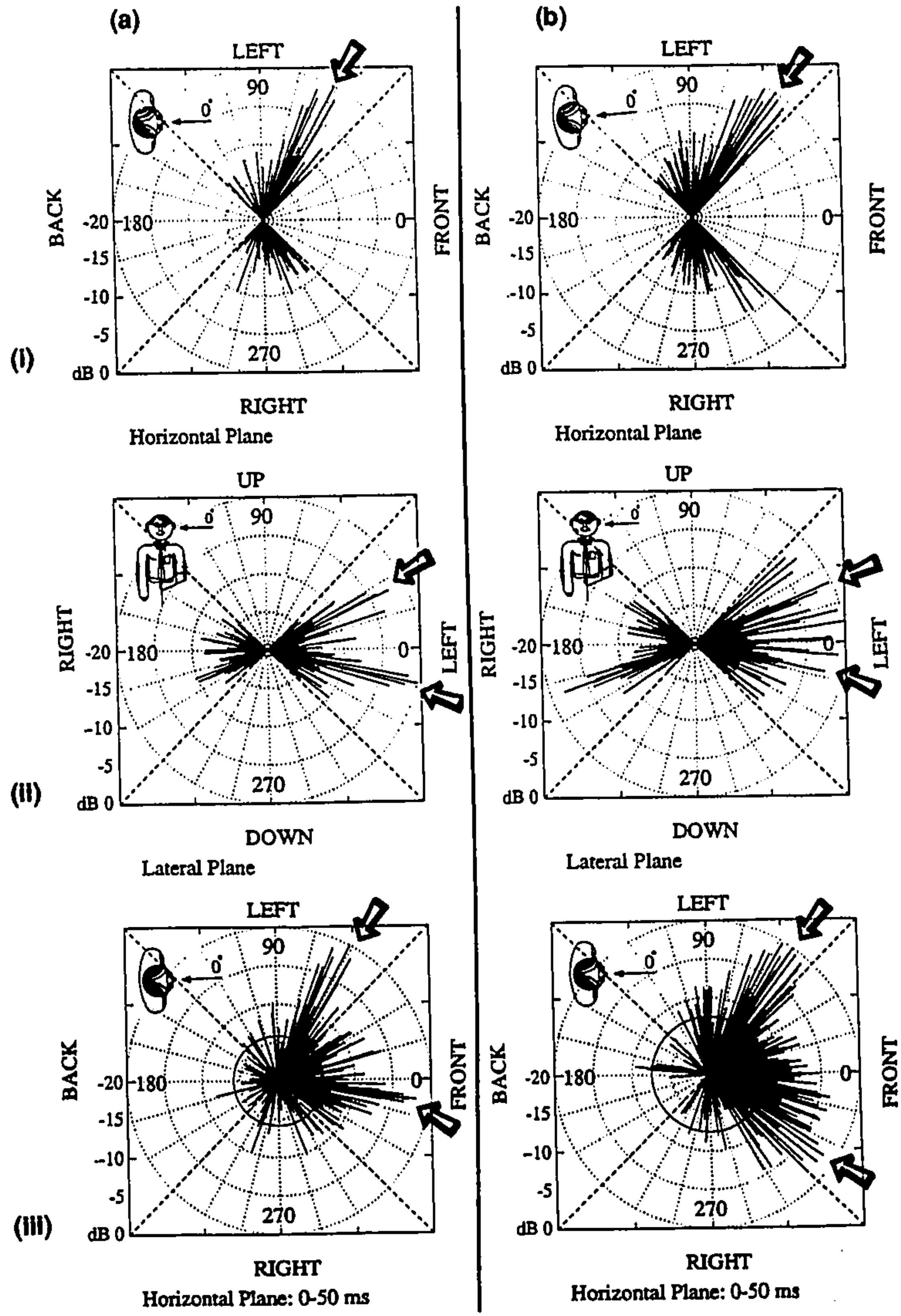


Fig. 6.11 Directivity patterns at listener position **R2**, (a) at 4 kHz and (b) at 1 kHz, (i) Horizontal plane, (ii) Lateral plane, and (iii) First 50 msec of received vectors from all directions.

also found. At high frequencies (2 and 4 kHz) the preferred arriving directions contributing most to subjective diffuseness has been found to be within 18° from straight ahead[77], while strong specular side reflections outside this angle cause unpleasant sound with an image shift of the sound source; thus the unsatisfactory seat assessment may be explained. Measured values of LEF_5 reported by both *RAMSoft-II* using a figure of eight microphone and *CBS-RAIMS* indicated high lateral energy fraction up to 0.31, but the figure of eight microphone would indicate the resulting energy along the lateral axis; whether the arriving energy is balanced from the left and right directions or received from only one side can not be inferred from that single-number indicator. At 1 kHz octave band shown in Fig. 6.11(b) successive strong left reflections dominate in the same solid angle as at 4 kHz, but at this frequency they are within an acceptable range of directional preference (+55° from straight ahead), thus this particular problem can be resolved by selective frequency absorption (2-4 kHz) in the recessed lounge. The subjective directional preferences versus octave frequency band are shown in Fig. 6.12.

At 500 Hz (see Fig. 6.13) most of the sound energy is received from the listener's front; stage ceiling suspended reflectors and hall ceiling reflectors cause strong acoustic rays in a + 45° direction above the horizontal plane and they are confined to a narrow angle in the lateral plane. It should be also noted that the front to back sound energy ratio at 500 Hz is high at this location.

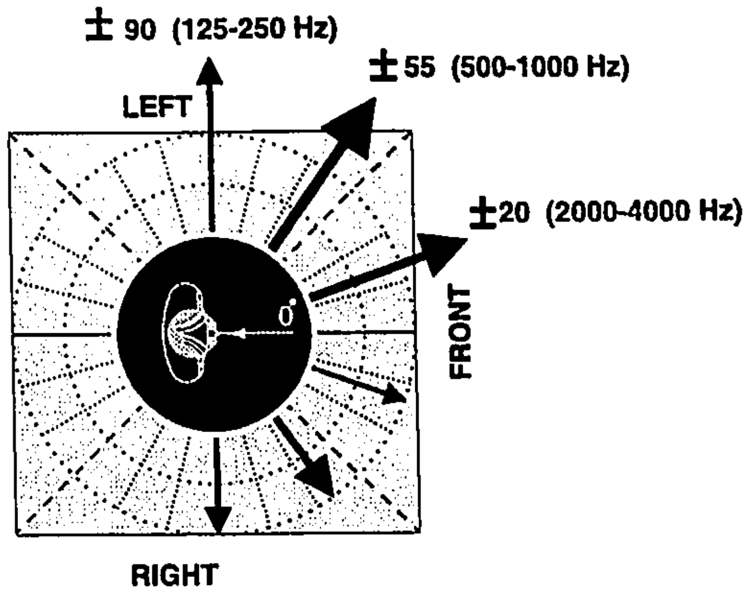


Fig. 6.12 The subjective directional preference versus octave frequency bands [inferred from reference[77]].

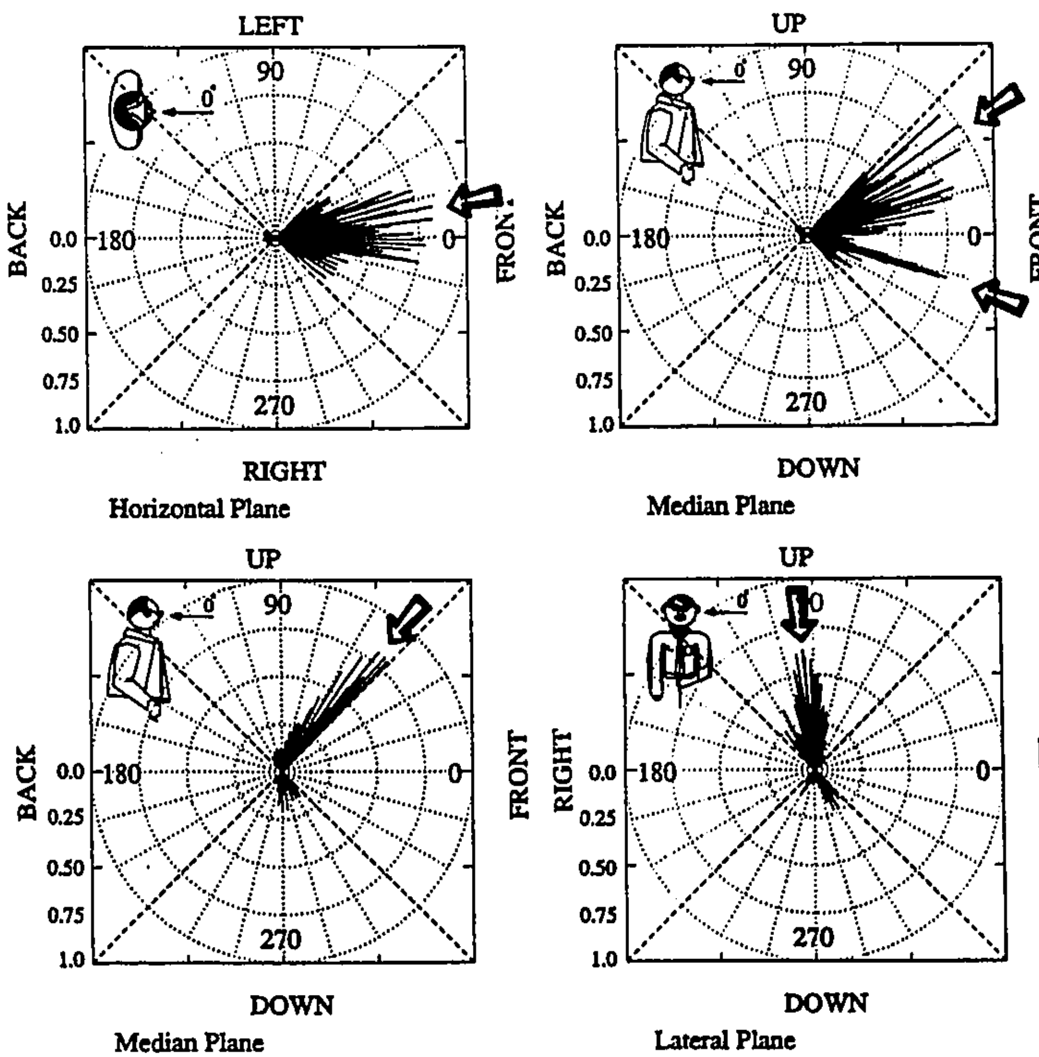


Fig. 6.13 Directivity patterns at listener position **R2**, in the "**LOYOLA**" concert hall, at 500 Hz. (Linear scale, Time interval: 0-1 sec)

6.3.5 Electroacoustics Modifications Sensing

The "*ST. LOUIS DE FRANCE*" church is equipped with an electroacoustical sound-reinforcement system which is installed to enhance speech intelligibility for the congregation. The sound system is designed to enhance speech intelligibility for both normal-hearing and hearing-impaired persons within the church's long reverberation time; typically this environment is in direct conflict with satisfactory speech perception. The installation is a distributed sound system with a total of 6 linear-array loudspeakers installed on side aisle structural columns with a uniform spacing of 10.5 m apart and directed towards the seating area in the main central nave, (see Fig 6.7(b)). This type of loudspeaker has high directivity to control the affected seating area without creating overlap or interference zones. Each loudspeaker is a linear-array of 8 single loudspeakers in a column. The linear-array loudspeakers are installed with the long dimension (L_s) vertical. The beam width of the radiated sound radiated from this type depends on the size of the line array in comparison with the wavelength. In the case at hand, the length of the line array (L_s) is 1.5 metres which means that it is effectively directive only from the frequency range 250 Hz and up, however its directivity begins to fall at distances greater than approximately ($L_s^2/2\lambda$) [3]. Considering that the speech spectrum is within the 300-3000 Hz, then the maximum effective distance covered by each loudspeaker ranges from ≈1.0 m at 300 Hz, ≈1.6 m at 500 Hz to ≈9.8 m at 3 kHz; thus the spacing between the loudspeakers locations may be considered adequate, although it is preferable to have it less than the reverberation distance which in this case is ≈7.5 m.

In practice, one to three omnidirectional microphones are used to pick up the speaker's voice on the altar, fed to an amplifier which in turn feeds the sound to the distributed loudspeaker columns without time delay.

During the measurement, when the system was "*ON* ", the control settings of the sound system were kept as normally used, only the gain had been slightly reduced to avoid overload from the test sound source.

To assess the capability of the measurement method to sense the effectiveness of the sound system, three positions were investigated with the sound system "*ON* " and "*OFF* " as judged by some contemporary sound quality indicators such as (EDT), (RT_{20}), (C_{80}), and (SNR_{95}). These measurement location were **L1**, **L2** and **L3**. The directional characteristics of the sound field at **L3** was measured employing the 3-D transient sound intensity method. Fig. 6.14 shows the measured values of room-acoustic indicators with the sound system "*ON* " compared to the "*OFF* " condition.

As can be seen from Fig. 6.14(a) to (d) minor or non-audible changes are found in the EDT and RT_{20} values in all three positions; this implies that the sound system when "*ON* " was ineffective. Other room-acoustic indicators such as (C_{80}) or (SNR_{95}) also remained unchanged when the sound system was "*ON* ". Speech

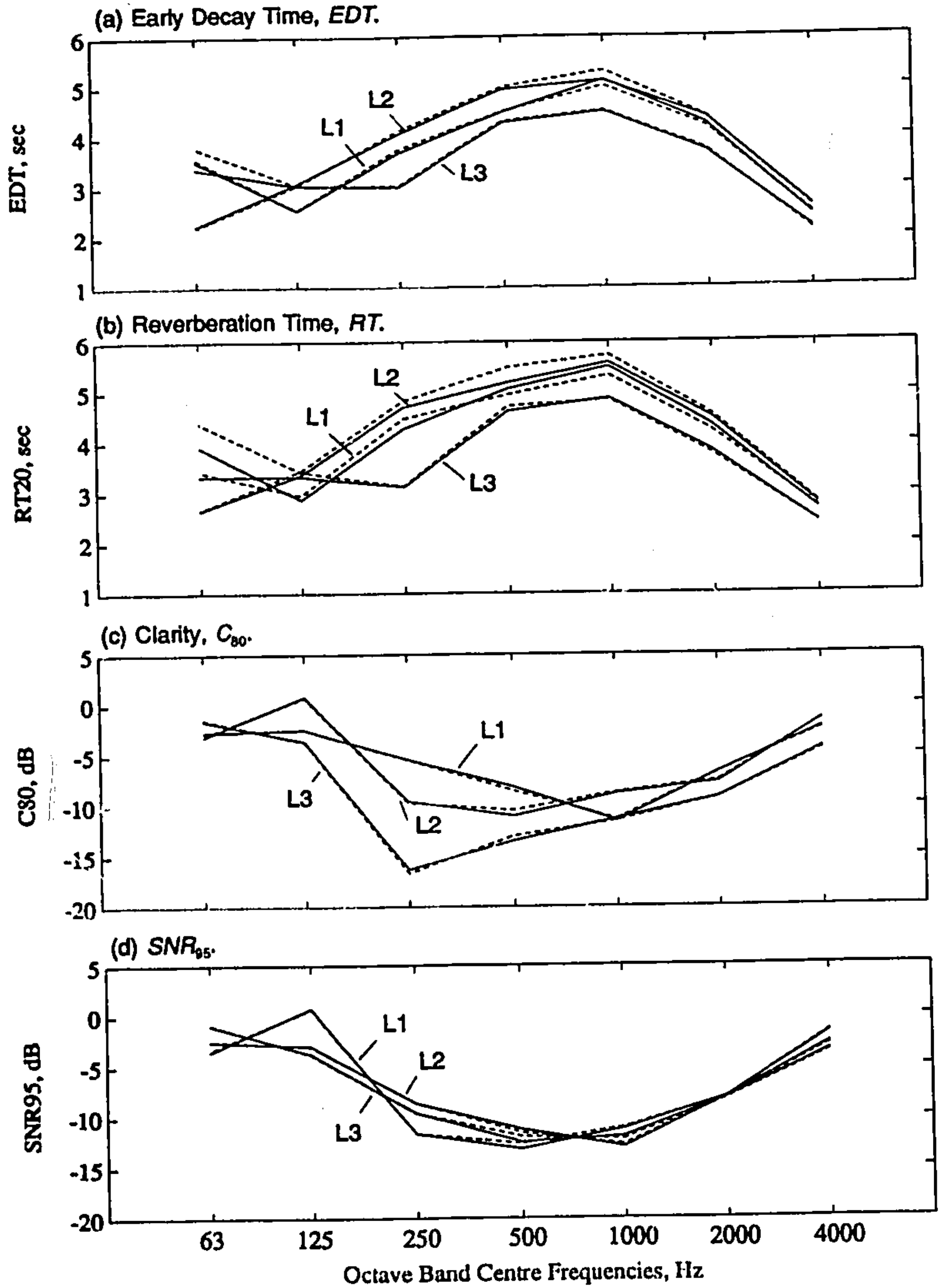


Fig. 6.14 Measured values of room-acoustic indicators with the sound system "*OFF*" (—) compared to the "*ON*" (---) condition in the "**ST. LOUIS DE FRANCE**" church, (a) Early Decay Time, *EDT*, (b) Reverberation Time, *RT*, (c) Clarity, C_{80}, and (d) SNR_{95}.

intelligibility depends not only on the sound loudness (i.e. the acoustical power radiated) but it is also dependent on the structure of the received sound impulse response, that is it depends upon both temporal and directional characteristics. Therefore a detailed investigation of the captured impulse response directional characteristics when the system was "*ON* " compared to "*OFF* " is beneficial for further assessment. Figs. 6.15(a) and (b) show the directional characteristics of the sound field received at position **L1** at the 1 kHz octave band frequency in the horizontal lateral and median planes with respect to the listener for the time period 0-1 sec). As can be seen, the frontal sound distribution is significantly different when the system is "*ON* " compared to "*OFF* " condition; sound rays can be observed reaching the listener from the locations of the linear loudspeakers near the seat location. Right and left reflections pattems are also changed. Examining the frontal sound directional distribution versus arrival time revealed that much of the reinforced sound energy arrives with more than a 95 msec delay after the direct sound and hence are detrimental for speech intelligibility. When the church is empty or occupancy is low the sound system is judged ineffective and this is due to the dominance of the church reverberation time particulary in the middle frequency range. The sound system's lack of signal delay feature is also found to be an influencing cause. Remedial treatments such as introducing a correct signal delay to the electrical signals applied to the distributed loudspeakers to provide useful sound energy integrated with the direct sound in the early reflection time period may cause the assisted resonance to be of some use.

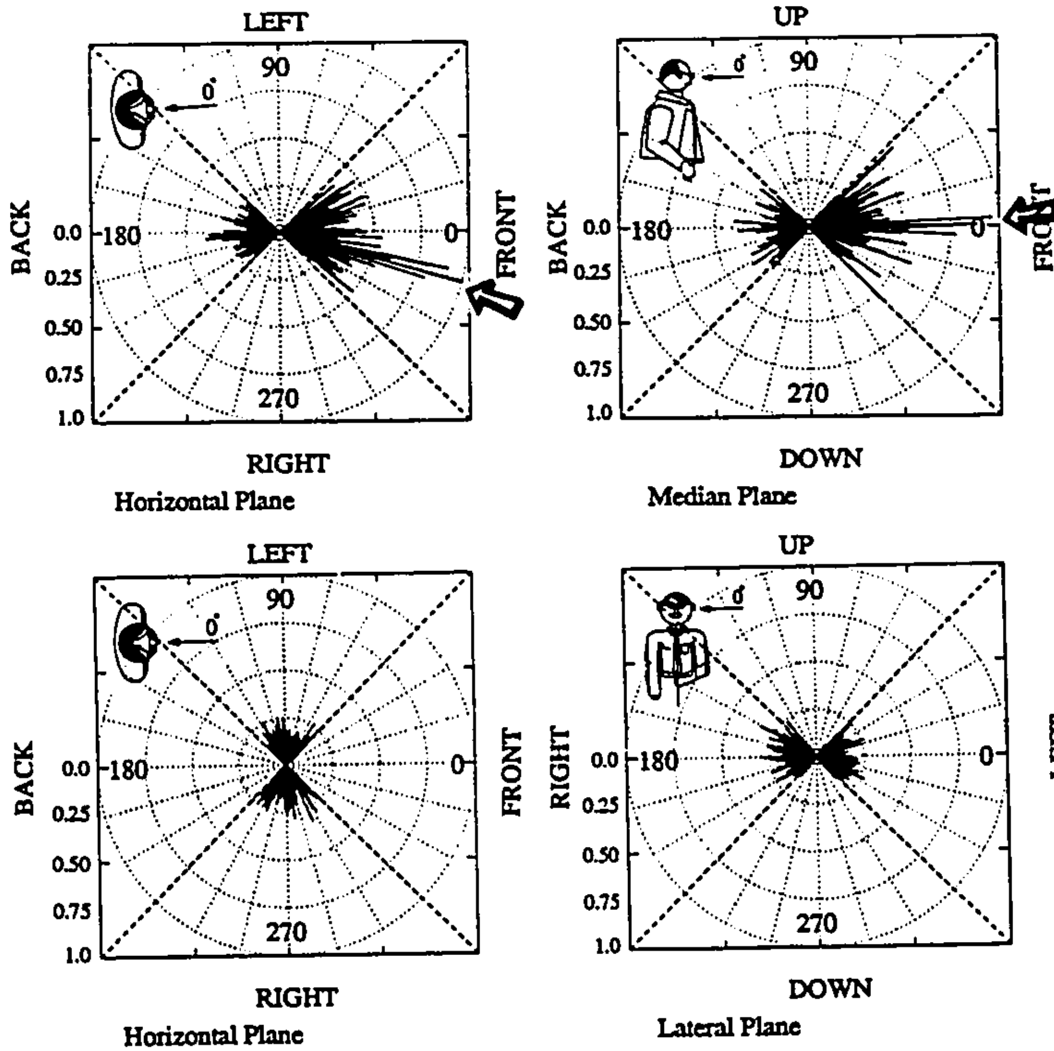


Fig. 6.15-a The directional characteristics of the sound field received at position **L1**, at 1000 Hz octave band frequency, with the sound system "*OFF*". (Linear scale, Time interval: 0-1 sec)

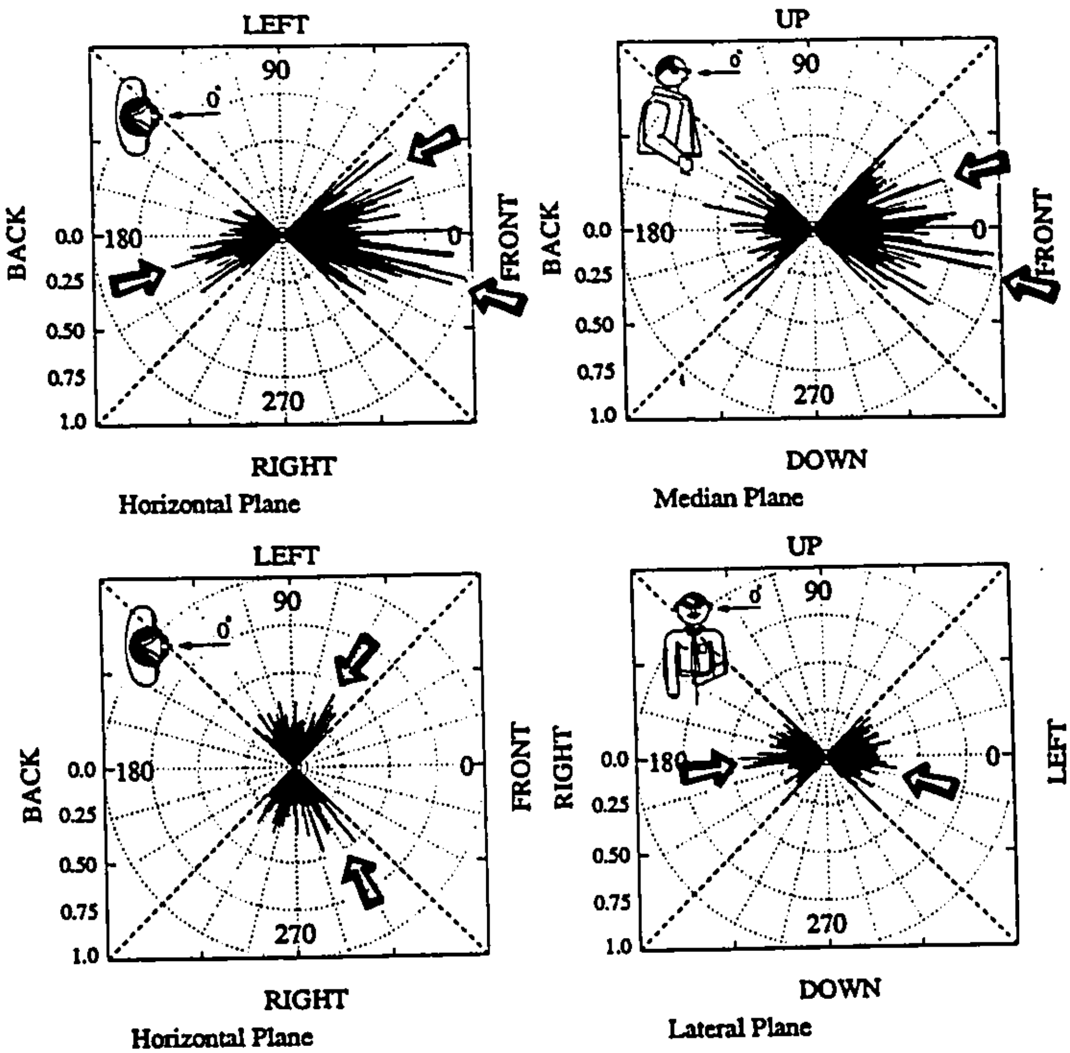


Fig. 6.15-b The directional characteristics of the sound field received at position **L1**, at 1000 Hz octave band frequency, with the sound system "*ON*". (Linear scale, Time interval: 0-1 sec)

A successive signal delays of 25, 58, and 90 msec for the loudspeakers nearest to the sound source (microphone) to the farthest loudspeakers respectively will achieve signal synchronization between the direct sound of the natural source to signals received from the column loudspeakers, and therefore reinforce the sound level in the first 95 msec from sound arrival at the listener locations. Adjustable digital electronic filters can also be recommended to smooth out the frequency response of the sound system and reduce its tendency for ringing. Further detailed analysis of direction-time energy can be made (see **Table 6.11**) but this will not be pursued here.

Overall it appears that a reduction of the long reverberation time by adding suitable absorbent materials to the wall surfaces or the addition of functional absorber would be the most useful course of action.

6.3.6 Detection of Interior Physical Changes

Two examples show the sensitivity of the measurement method and system to detect changes in the directional characteristics due to interior physical changes. The "***LOYOLA***" concert hall is designed as a variable hall with its adjustable rotating side wall panels, the acoustic spatial information at listener positions **R4** and **R5** were investigated when the side wall panels are "*OFF*" compared to the "*ON*" conditions. Figs. 6.16(a) and (b) depict the measured directivity patterns at **R4** in these two hall conditions. The first 300 msec in the 1 kHz octave band frequency is chosen for comparison. As can be seen in Fig. 6.16(b) discrete reflections are evident when the panels are "*ON*" in both front and lateral directions after the direct sound arrival. Left wall reflections are more pronounced than right in both conditions but with higher magnitude in the "*ON*" case. The changes in the directivity patterns are emphasized by the indicated arrows for ease of comparison. The changes in the received sound directivity can also be clearly seen in Figs. 6.17(a) and (b) which show the same spatial information but in a time segmented windows of 0-50, 50-100 and 100-200 msec and with the first 20 dB relative to the maximum received sound. Both received sound magnitude and directional distribution are quite modified in each time window. Similar changes at listener position **R5** are observed at 1 kHz. Comparison of time-segmented directivity patterns which are illustrated in Fig. 6.18(a) and (b) for both hall configurations showed modification of the sound spatial distribution. This location is near the hall back wall, thus reflections received from the back of the listener

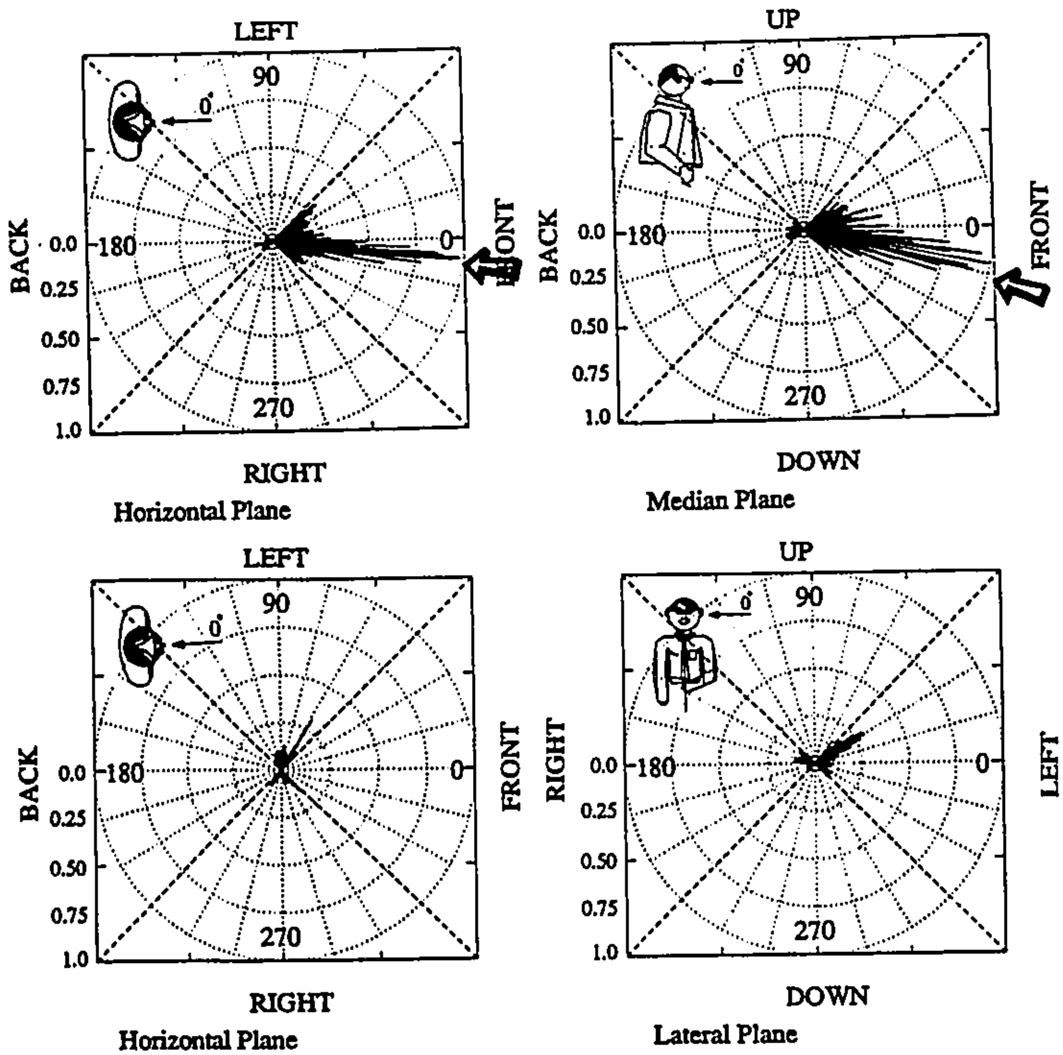


Fig. 6.16-a Directivity patterns of received sound energy at listener position **R4** in the "**LOYOLA**" concert hall, at 1000 Hz octave band frequency, with side wall reflective panels "*OFF*". (Linear Scale, Time interval: 0-1 sec)

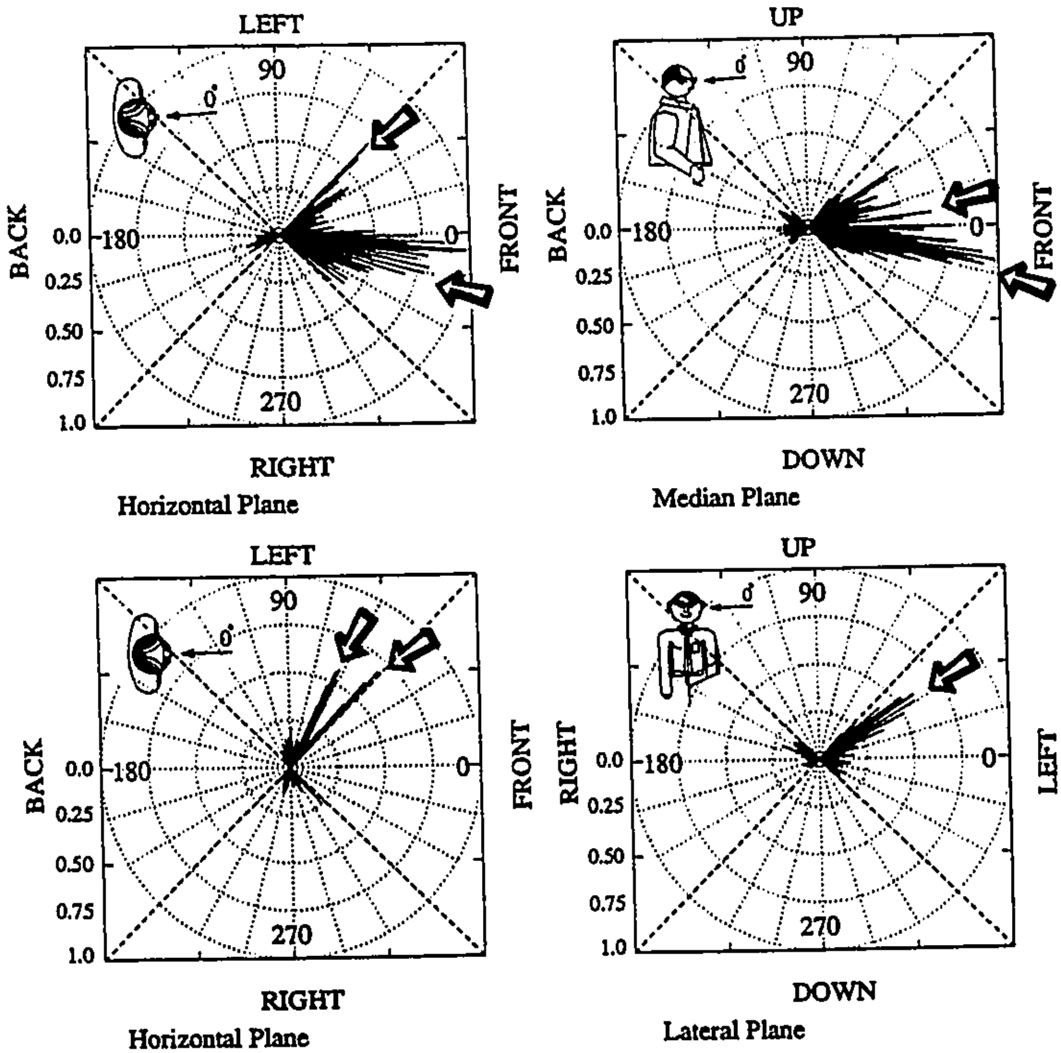


Fig. 6.16-b Directivity patterns of received sound energy at listener position **R4** in the **"LOYOLA"** concert hall, at 1000 Hz octave band frequency, with side wall reflective panels "*ON*". (Linear Scale, Time interval: 0-1 sec)

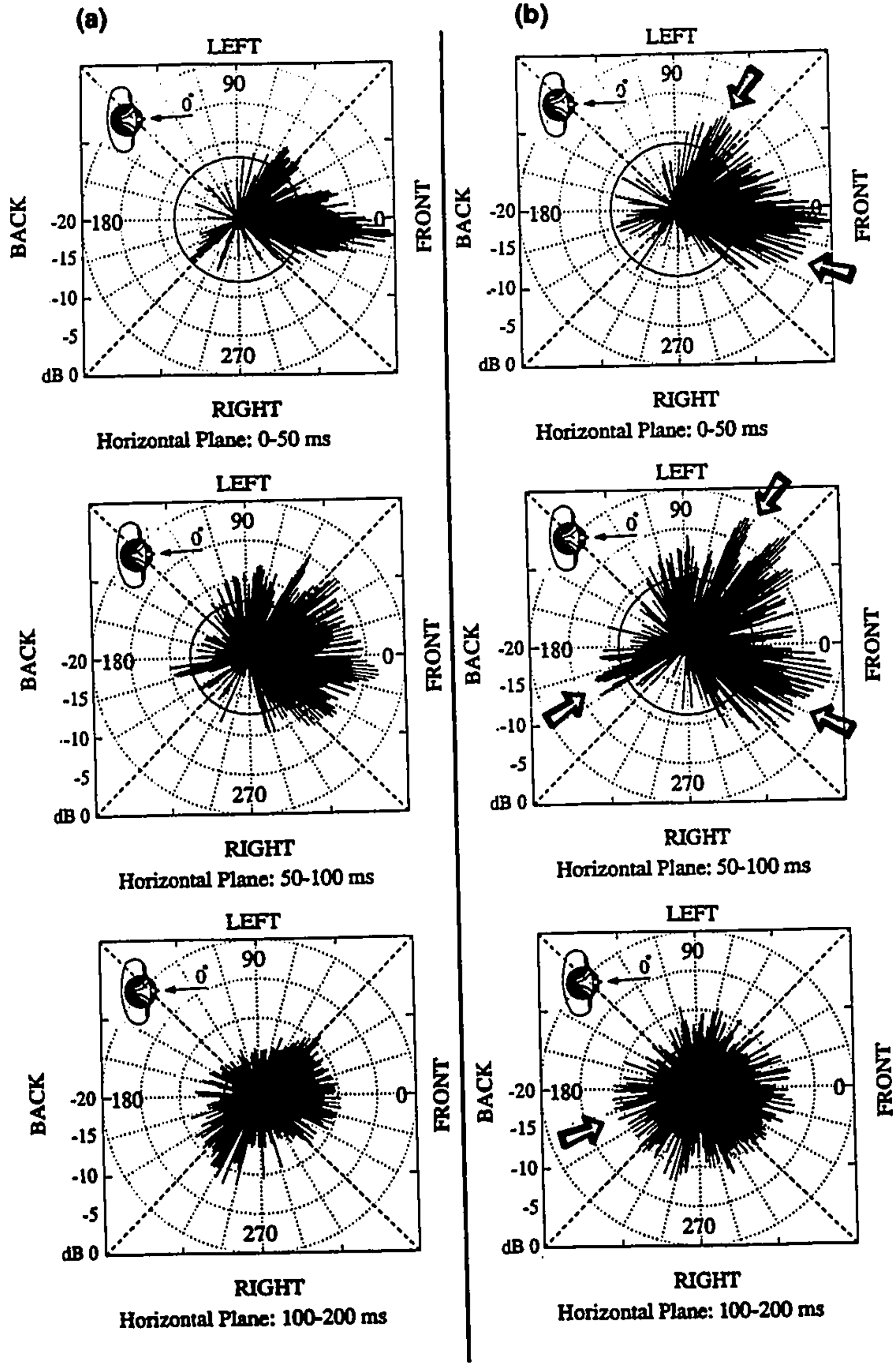


Fig. 6.17 Directivity patterns of received sound energy (in successive time windows) at listener position **R4** in the "**LOYOLA**" concert hall, side wall reflective panels (a) "*OFF*" and (b) "*ON*". (dB scale, Time interval: 0-300 msec)

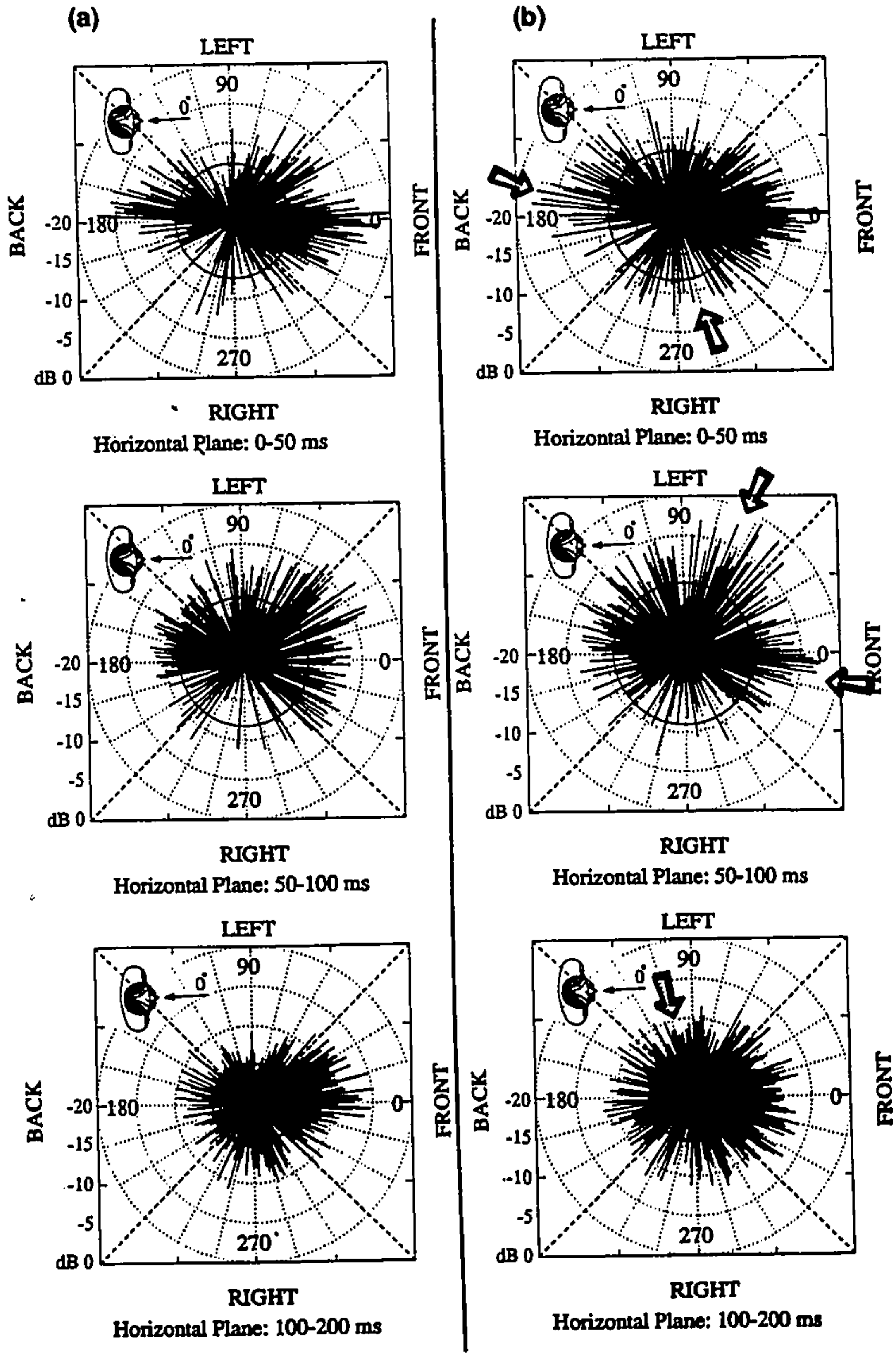


Fig. 6.18 Directivity patterns of received sound energy (in successive time window) at listener position **R5** in the "**LOYOLA**" concert hall, side wall reflective panels (a) "*OFF*" and (b) "*ON*". (dB Scale, Time interval: 0-300 msec)

can be readily visualized compared to those received at location **R4**. Other octave band frequencies were also examined and showed similar changes in the sound energy distribution.

6.3.7 The Effect of Surrounding Interior Features: Cause and Effect Appreciation

This section describes four examples of detecting the influence of surrounding features at a listener location.

Example 1 : Fig. 6.19 illustrates the directivity patterns at 500 Hz for position **R11** located near the back wall of the "***LOYOLA***" concert hall ; where the back wall upper surface is covered with downward tilted reflective panels (see Fig. 6.4) designed to avoid axial reflections and to minimize echoes. Their effect on reflections from above and back are clearly seen. Because this measurement position is so near to the right wall, many of the lateral reflections are received from the left hand side and with a wider solid angle. The discrete reflections due to the curved ceiling panels are also dominant at this location compared to the reflections observed (Fig. 6.20) for example at position **R8** located in the back part of the recessed lounge which has a low flat ceiling. The effect of the back wall of the lounge is also evident in Fig. 6.20.

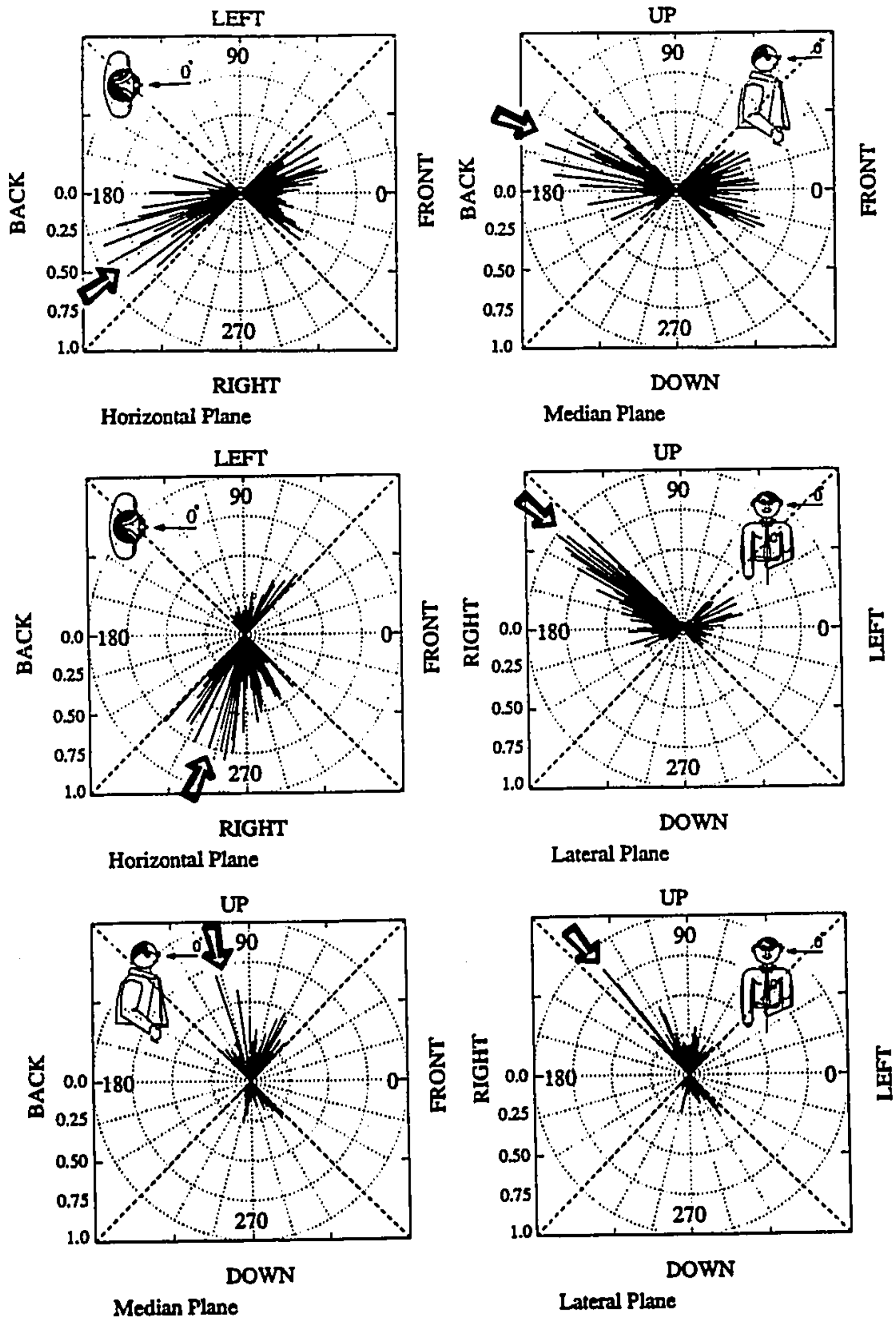


Fig. 6.19 Directivity patterns of received sound energy at 500 Hz, at listener position **R11** in the "**LOYOLA**" concert hall, side wall reflective panels "*OFF*". (Linear scale, Time interval: 0-1 sec)

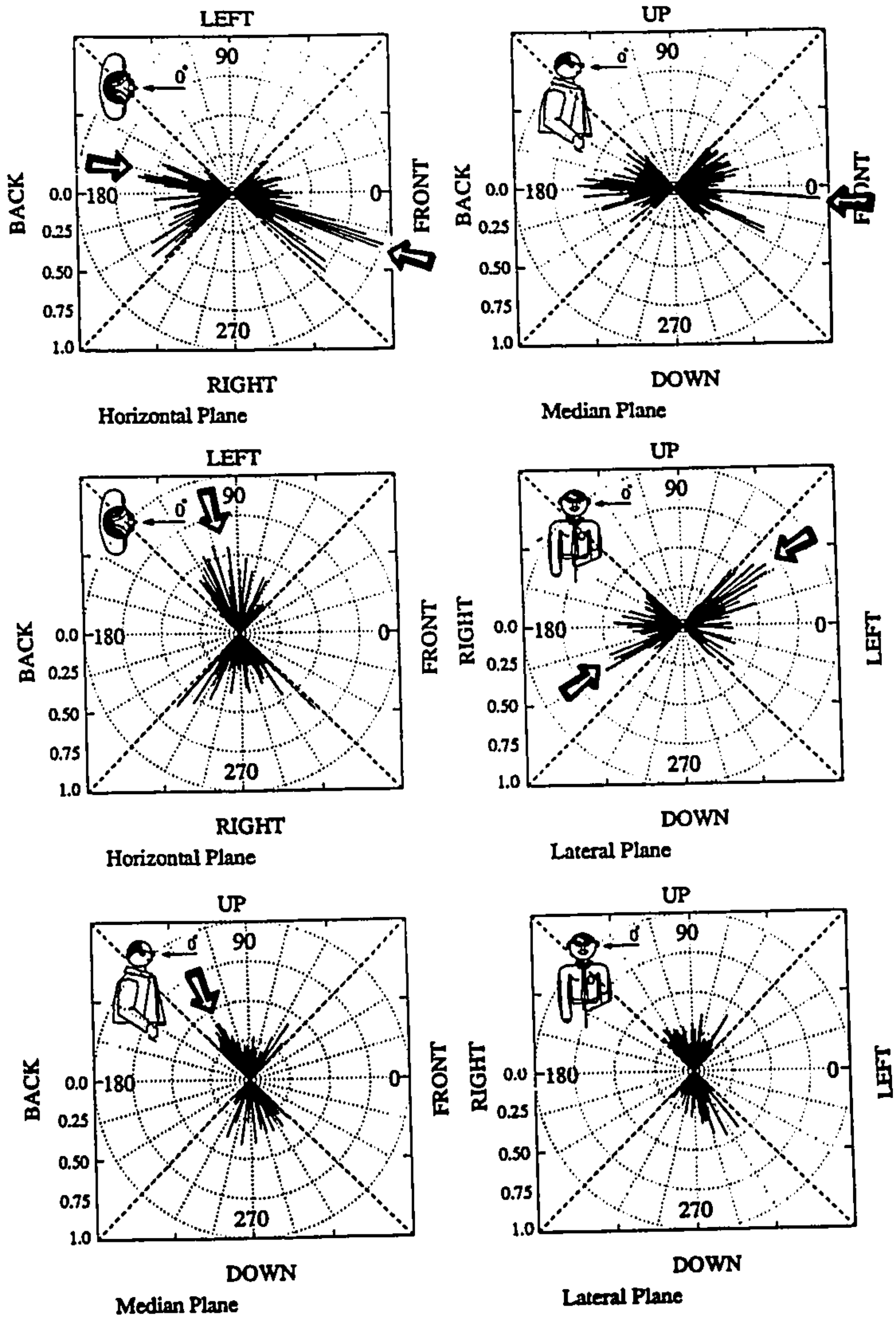


Fig. 6.20 Directivity patterns of received sound energy at 500 Hz, at listener position R8 in the "LOYOLA" concert hall, side wall reflective panels "*OFF*". (Linear scale, Time interval: 0-1 sec)

Example 2 : Examining listener locations **R1** to **R6** in the "LOYOLA" concert hall particularly at high frequencies and calculated (*F/B*) energy ratios revealed less sound energy from the back of the listener, except for the 2 and 4 kHz octave bands at location **R7**; from Fig. 6.21 one can see relatively strong reflections from the back and when the seat area was examined the vertical wooden facia of a recording control panel was found just to the back of the measurement point; the dimensions of the panel (height ≈ 30 cm above the seat back level) in relation to the wavelength of 2 and 4 kHz (17 and 8.5 cm respectively) make it an effective reflector in this frequency range.

Example 3 : Fig. 6.22 shows the directivity patterns at 125 Hz at the listener location **R7**, in the "***LOYOLA***" concert hall both in the median and lateral planes, where the effect of the ceiling configuration is evident by the sound reflections received from above with higher magnitude at around +30° above the horizontal plane; the symmetry of these reflections in relation to the seat position in the hall can be observed in the lateral plane. Relatively insignificant sound energy is received from the floor direction as one might expect.

Example 4 : Fig. 6.23 shows the directivity patterns at 125 Hz at listener location **P2** in the "***LOYOLA***" chapel both in the horizontal and median planes on a linear and dB scale (first 20 dB relative to maximum sound); the dominance of frontal sound energy is evident. It should be also noted that, in enclosures, low frequency sound wave fronts tend to curve (as part of spherical-shaped waves) and this

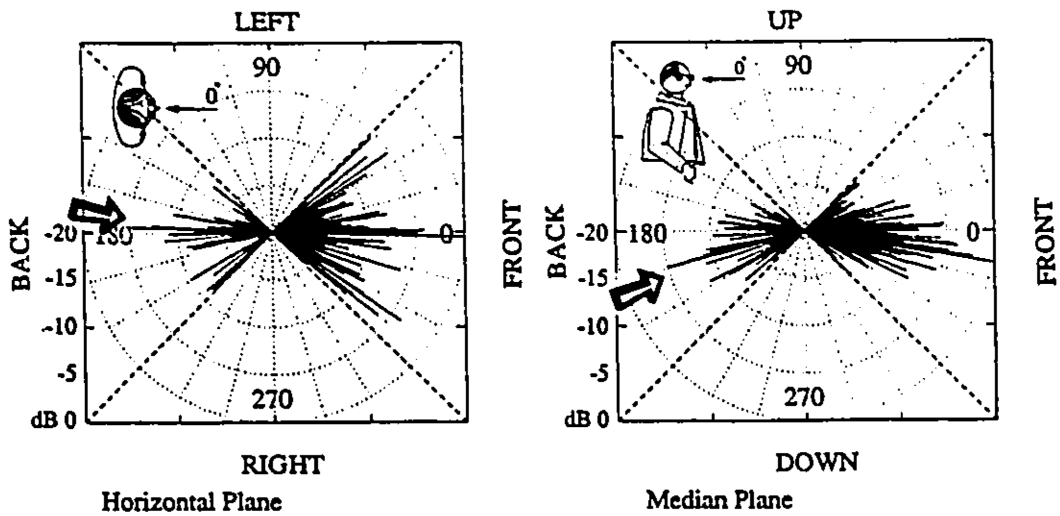


Fig. 6.21 Directivity patterns at 4 kHz of received front and back sound energy at listener position **R7** in the "**LOYOLA**" concert hall. (dB scale, Time interval: 0-300 msec)

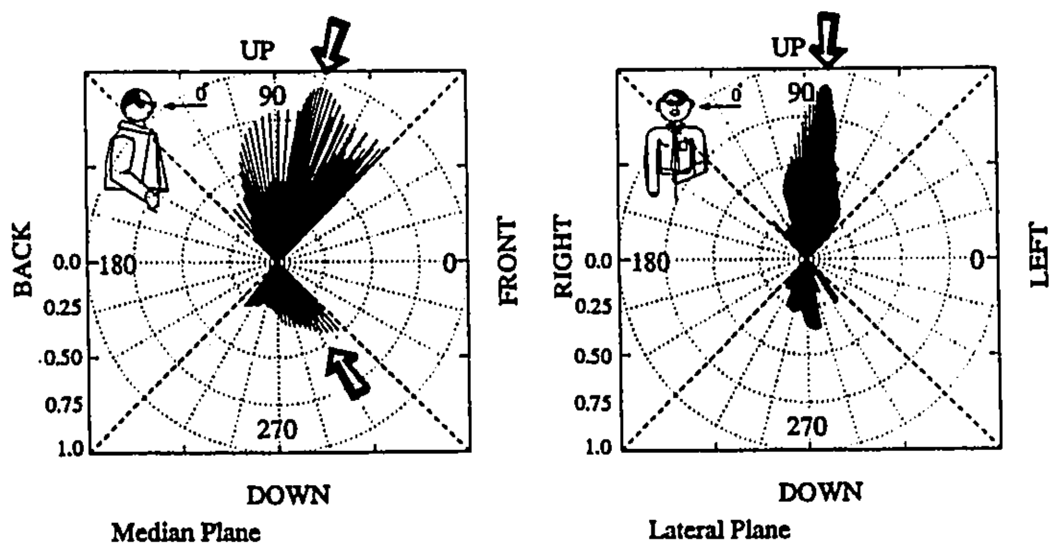


Fig. 6.22 Directivity patterns at 125 Hz of sound energy received from the ceiling and the floor at listener position **R7** in the "**LOYOLA**" concert hall. (Linear scale, Time interval: 0-1 sec)

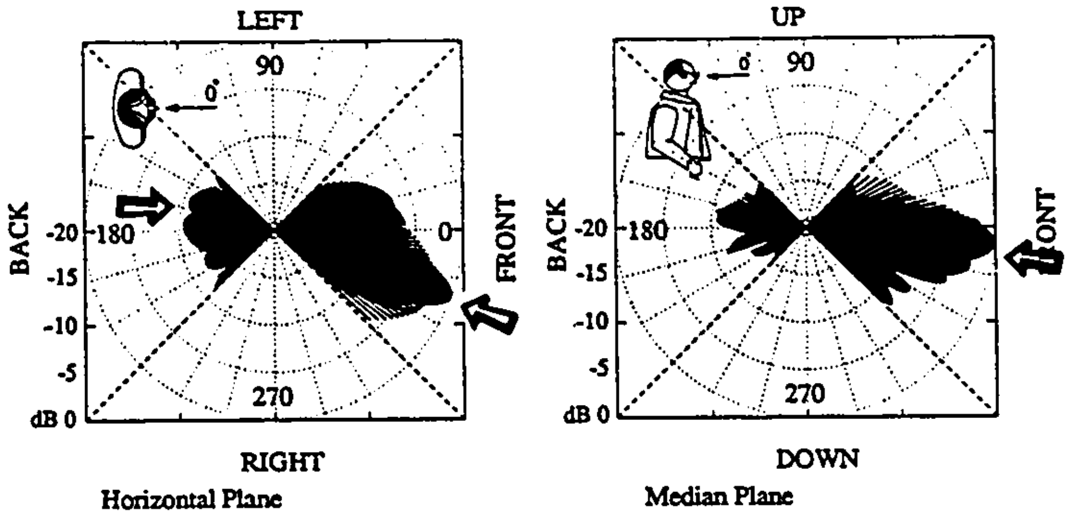


Fig. 6.23 Directivity patterns at 125 Hz of sound energy received from front and back at listener position **P2** in the "**LOYOLA**" chapel. (dB scale, Time interval: 0 - 500 msec)

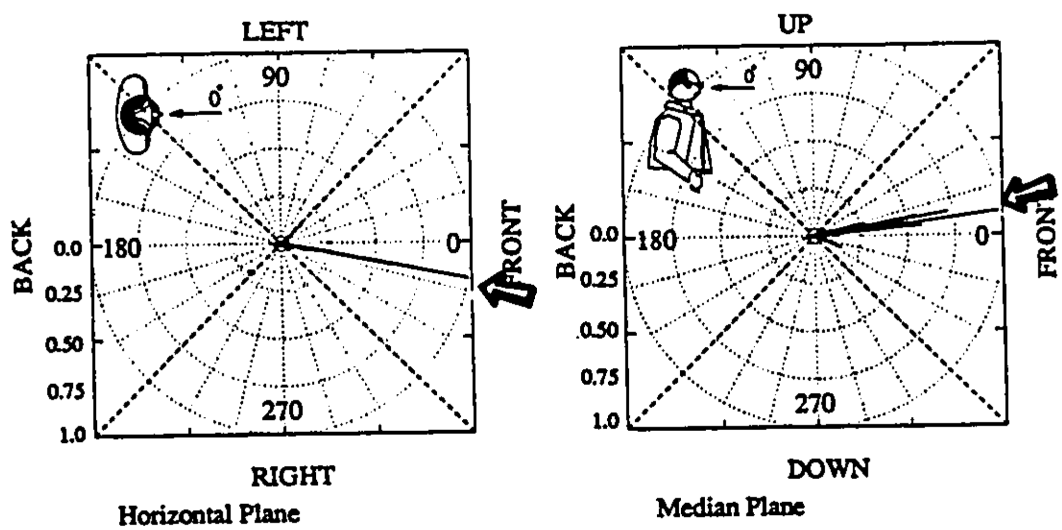
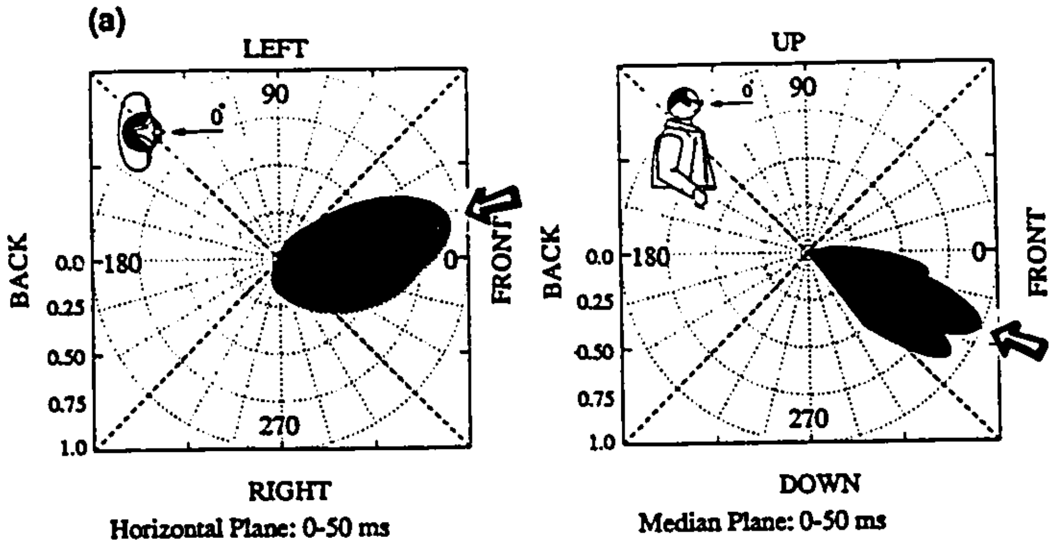


Fig. 6.24 Directivity patterns at 2 kHz of sound energy received from front and back at listener position **R1** in the "**LOYOLA**" concert hall. (Linear scale, Time interval: 0-1 sec)

appears in the successive time dependent intensity vectors forming a smooth envelope in both the horizontal and median planes. In the high frequency range (small wavelength compared to the space dimensions), they appear in the directivity patterns as either a distinct vector or a bundle of discrete vectors as shown in Fig. 6.24 of a 2 kHz measurement at location **P1** in the "LOYOLA" chapel. The spherical waveform shape phenomena is even more pronounced at 63 Hz; Fig. 6.25(a) illustrates the first 50 msec of the frontal intensity vectors at 63 Hz at location **R4** in the ***"LOYOLA"*** concert hall in the horizontal and median planes. Fig. 6.25(b) shows received vectors from all directions in the first 50 msec, back reflections are not evident but the overall envelope of received sound vectors confirms the interpretation of low frequency behaviour. Fig. 6.26 depicts the 63 Hz octave band frequency at listener position **B1** in the ***"LOYOLA"*** chapel and the previous findings with respect to the shape of the wave front are supported but because of its relative location, back wall reflections are evident.



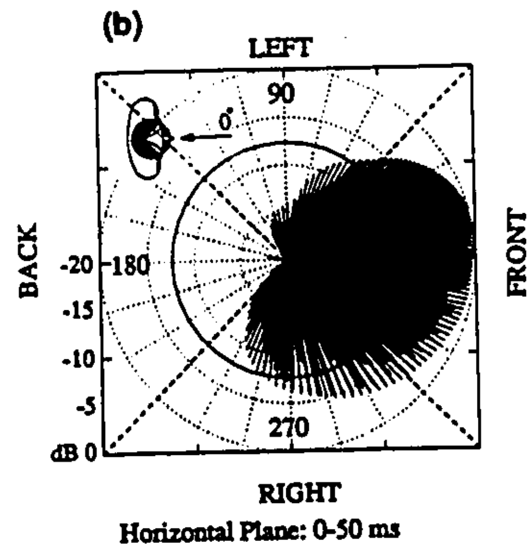


Fig. 6.25 The first 50 msec of sound intensity vectors received at listener position **R4** in the "**LOYOLA**" concert hall shown in both (a) The horizontal and median planes (Linear scale), and (b) Received vectors from all directions, (dB scale).

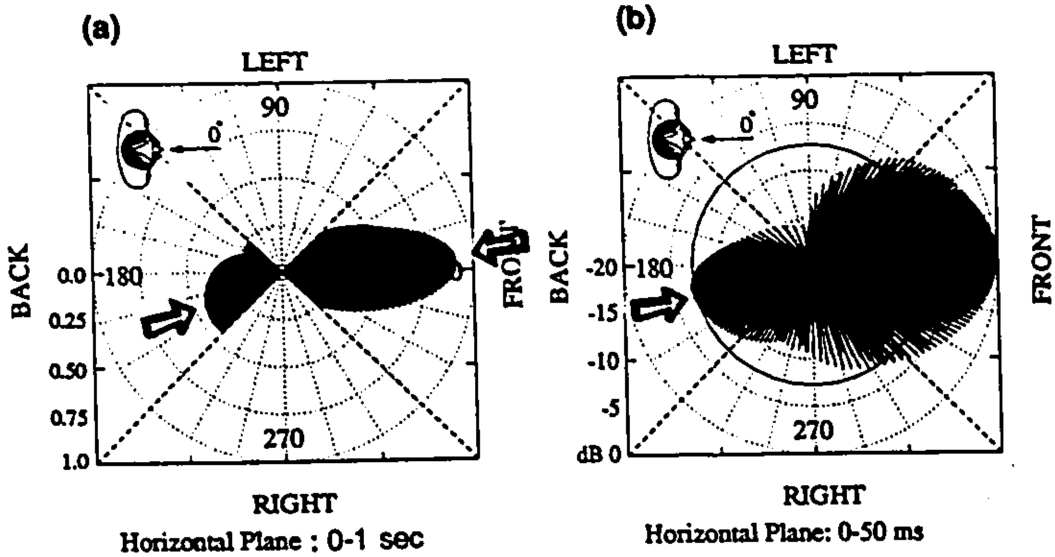


Fig. 6.26 Directivity patterns at 63 Hz in the "LOYOLA" chapel at position B1 in the balcony in the horizontal plane, (a) Front and Back sound energy, and (b) Sound energy of the first 50 msec.

6.4 SUMMARY

It is shown that by combining the sound field directional characteristics and room-acoustic objective indicators, new avenues for detailed sound quality evaluation and diagnostic assessment at different levels of detail are now possible. While such spatial information might seem excessive, the level of assessment can be chosen based on the case or the acoustical problem at hand. The measurement method is shown to be sensitive to variations or changes in the acoustic field caused either by physical changes and/or fixed surrounding features.

CHAPTER 7

CONCLUSIONS AND RECOMMENDATION FOR FUTURE STUDIES

7.1 CONTRIBUTIONS AND FINDINGS

During the course of this research the following contributions and findings are claimed:

** A three-dimensional transient sound intensity measurement method has been introduced which enables one to obtain the directional characteristics of sound fields in enclosures. The method has been validated and its accuracy found adequate.

** A simple, practical, and inexpensive measurement system has been developed and validated in order to obtain spatial information of sound fields in enclosures and contemporary room-acoustic indicators. The system utilizes sound intensity measurement from one pair of microphones in three successive orientations to calculate transient intensity vectors although it can be extended to three pairs. The sound field can then be visualized on an energy directional basis versus arrival time and hence analyzed in greater detail than hitherto possible. The system development takes into account the following aspects:

- Easy to implement with the minimum hardware and cost (Estimated costs ≈ $ 25,000 or less, compared to one only existing system "*FOURMIC* " selling for $ 140,000), and not yet fully described in research literature.
- Easy to calibrate, operate and update.
- Provide utilities for signal analysis and an easy to interpret information presentation which yields in-situ appreciation between cause, effect and acoustical quality at a listener position.
- Provide flexibility for conducting measurements over a wide range of frequencies (63 Hz to 8 kHz) in environments of both short and long reverberation times (up to 5-7 seconds).

The system, being the first of its kind, enables the study of acoustic attributes hitherto not possible for example, to examine diffuse sound in large rooms designed for music purposes, in order to enhance acoustical quality. Quality is said to be measured in existing rooms by known room-acoustic indicators but most of them have been developed without regard to direction sensing. Measurements in existing halls show that the sound field is far from diffuse. Measuring the directional characteristics of sound in existing halls provides the prospect of quantifying such temporal and spatial distribution. In order to utilize the now available directional information, existing directional diffusion indices and techniques for room-acoustic evaluations have been reviewed and suggestion made for their reconsideration.

** Prospective, simple quantifiers for sound diffuseness are developed and proposed for their systematic measurement and application in room acoustic evaluations.

Three case studies as example applications have been carried out to validate and investigate the use of both the method and the measurement system in real enclosures. Based on these measurements, the following findings are drawn:

- In addition to assessing acoustic quality using pressure-based room-acoustic indicators, visualizing the directional characteristics of sound fields at the listener position is also beneficial employing 3-D transient sound intensity impulse responses.

- Detailed changes in sound energy directional distribution due to electroacoustical modification can be detected and assessed for quality contribution whilst also providing the opportunity to consider surface/source cause and effect relationship.

- The effect of the enclosure details and features upon listener locations, such as reflective high or low ceilings, recessed lounges, acoustical panels and balconies upon listener locations can readily be visualized for both diagnostic purposes and the investigation of sound diffusion. Diagnostic assessment of acoustical defects is also possible, for example, the contributions of particular boundary surfaces to early sound reflections can be identified, isolated and "what if" designs evaluated.

- In order to characterize the acoustical characteristics of a room in terms of the absorption properties of boundaries, one should employ the reverberation time observed from the decay of sound energy expressed by acoustic energy density or sound intensity level (RT_I). RT_I is also found to be similar to early decay time (EDT_p) and thus is subjectively more relevant than the reverberation time based upon pressure (RT_p).

- Spatial information is shown to be essential and complementary to the single-number indicators for a full and comprehensive acoustical evaluations.

7.2 SUGGESTIONS FOR FURTHER WORK

The full potential of the measurement method is yet to be realized and the following research topics are recommended for future studies, for example:

- Determination of in-situ surface impedance employing sound intensity following research topics are recommended for future studies, for example:

- Determination of in-situ surface impedance employing sound intensity measurements to improve surface property values employed in computer simulation models.

- Sound directional characteristics received by the listener are affected by the head. The head effects might be taken into account by investigating a dummy head result from given intensity vectors input and then deducing the head transfer function. In this way, further dummy head results can be deduced directly from intensity vector assignments.

- The present study has developed and proposed a number of descriptors for sound diffusion quantification, but further research work is necessary to establish correlations between room-acoustic indicators relating to directionality and diffusion and to establish which of the proposed descriptors might better serve as quality indicators.

- The developed method and system allow the measurement of newly introduced objective direction-based indicators of known subjective relevance such as Front/Back energy ratio (*F/B*), but hitherto impossible to be measured in the field. The measurement of this indicator in a number of enclosures will allow a full assessment of its subjective relevance and may

- The measurement method and tool developed here expose a new dimensional perspective to workers developing objective indicators of subjective response.

- The study reveals the need for further subjective studies related to current room-acoustic indicators such as lateral energy fraction (*LEF*); a figure of eight microphone commonly used to measure it would indicate the resulting energy along the lateral axis whether the arriving energy is balanced from the left and right or received from only one side. This lack of sensitivity must be questioned and the subjective relevance of differing lateral energy distribution needs to be resolved. Also the temporal and the directional characteristics of the late reverberant reflections required to achieve a certain degree of envelopment needs further objective and subjective investigations.

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[156] R. W. Guy, **"Intensity Assessment Employing Probe Reversal,"** App. Acoust. Vol. **40**(1), pp. 57-68 (1993).

APPENDIX A-I: Interactive Control Panel for Setting the Analog Input (Acquisition) Parameters: Functions' Description

PANEL TITLE : Multiple Channel DAQ (Scan To Disk)

FUNCTION	:	**FUNCTION DESCRIPTION**
BOARD	:	Initialize the data acquisition board (slot 7)
CHANS TO BE USED	:	Indicate the total number of channels to be used
NUM OF AI CHANS	:	Specify the total number of channels to be used for acquisition
MAX FREQUENCY	:	Specify the maximum octave band frequency desired for the analysis
SAMPLES/CHAN	:	Specify the number of data samples to be acquired per channel
SAMPLE RATE/CHAN	:	Specify desired sample rate (F_s,samples/sec) per channel
SCANNING RATE	:	Specify desired scanning rate (samples/sec)
ACTUAL RATE/CHAN	:	The actual sampling rate employed by A/D counter
CH1... To....CH8	:	Specify desired channels' number and order in the acquisition process
ACCEPT	:	Accept the selected channels' number and order
REST	:	Reset channels' selection to NONE
MUX28	:	Use the channels multiplexer B&K (Type 2811)
GAIN	:	Specify the acquisition gain (Volts, bipolar)
FILE NAME	:	Input a file name to save the acquired data
FILE MODE	:	Set data file write mode
Append	:	Append data to an existing file
OverWrite	:	Write to a new file or overwrite if file exists
DITHER	:	Enable or Disable dithering while acquiring data
[*NOTE :*		*The dither circuity, when enabled adds approximately 0.5 LSB rms of white Gaussian noise to the signal to be converted by the A/D, it has the effect of forcing the quantization noise to become a zero-mean random variable and increases the resolution of the board to more than 12 bits*].
TRIGGER	:	Set desired triggering type
INTERNAL	:	Use the acquisition board internal trigger

APPENDIX A-I: Continued

FUNCTION	FUNCTION DESCRIPTION
EXTERNAL	Use an external triggering signal
COUNTER 5	Use counter 5 on the board as a triggering signal
ON LIGHT	Use internal trigger but put light on when triggering
DELAY	Specify a time delay for acquiring data (multiple of 55 msec)
ACCEPT ALL	Read all input parameters and check if any is invalid or outside a defined range
SCAN 2 DISK	Start the scanning Process (save data directly to disk)
RESET SCXI	Reset the signal conditioning chassis and board
BAK NOISE	Switch to "Background Noise" measurement panel
IR...<OBJ>	Calculate the room impulse response (*IR*) (if sequence length is > 8191 samples)
IR..<1 - 8 K>	Calculate the room impulse response (*IR*) (if sequence length is < 8191 samples)
PLOT AI	Show an interactive graph of the channels' acquired data (in sequence)
OVERPLOT	Show an interactive graph of the channels' acquired data (over-imposed)
CLEAR AI	Clear the analog input (acquisition)
NEXT 2 AO	Switch to the analog output interactive panel
FILE COPY	Copy utility
FILE DEL.	Delete utility
HARDCOPY	Print the current interactive panel
WRITE MAT	Write the acquired data in a "MATLAB" Software format
QUIT PROGRAM	Exit the program
T. BASE	Indicate the acquisition time base
T. INTV.	Indicate the acquisition number of intervals
dt	Indicate the acquired signal time interval (msec)
SEQ.(s)	Indicate the total duration of acquired data sequence
ACQ. STATUS	Indicate the acquisition status (0 if successful)
ACQ. SAMPLES	Indicate the total number of acquired data samples
ERROR FUN.	Message of ongoing action or function name
ERROR	Indicate error number (if encountered)
HARDCOPY	Print a hard copy of the current interactive control panel

APPENDIX A-II: Interactive Control Panel for Setting the Stimulus Signals Analog Output Parameters: Functions' Description

PANEL TITLE : ANALOG OUTPUT (FROM BUFFER OR DISK)

<u>FUNCTION</u>	: <u>FUNCTION DESCRIPTION</u>
AO CHAN	: Select Analog channel number
MLS FILES	: Available m-sequences (255- 65535 samples)
SINE WAVE	: Generate a defined sine wave
PULSE	: Generate a defined pulse
IMPULSE	: Generate a defined impulse
CHIRP	: Generate a chirp signal
WHITE NOISE	: Generate white noise
AVAILABLE "STIMULUS"	: Other available stimulus signals saved in files
SCALE NOISE	: Scale the generated noise to a given voltage
SCALE 2 BUF	: Scale the generated signal to a given voltage
DISK -- AO	: Output signal from file on disk to selected analog output channel
BACK 2 AI	: Switch to Analog input interactive panel
ACQ. FILE	: Show the acquisition file name
WFLOAD	: Load the given waveform into the output buffer
WFS&SCAN	: Start waveform output and scan input channels simultaneously
WFSTART	: Start waveform output without acquiring data
WFPAUSE	: Pause ongoing waveform output
WFRESUME	: Resume waveform output
WF CLEAR	: Clear the analog output process
WFDBDAQ	: Start waveform output and double buffer data acquisition simultaneously
MLS65K DB. WFM	: Output an m-sequence of 65535 samples length
T. BASE	: Indicate Waveform Time base
UP.INTV	: Indicate Number of waveform samples interval
MLS BANDWIDTH	: Indicate the m-sequence frequency bandwidth (Hz)
FQ-FL	: Indicate lower bandwidth limit (Hz)
FQ-FU	: Indicate upper bandwidth limit (Hz)
WF. DURATION	: Indicate total Waveform duration (sec)

APPENDIX A-II: Continued

FUNCTION	: FUNCTION DESCRIPTION
PLOT AO	: Show an interactive graph of the data loaded for output
SAMPLES NUM	: Specify number of data samples to output
START POINT	: Specify the number of the first data sample to start output from
END POINT	: Specify the number of the last data sample to end output at
VOLTAGE X..	: Specify desired output voltage
UPDATE RATE	: Specify the waveform generation rate (= F_s or different)
ITERATIONS	: Specify number of waveform iterations
GEN. STATUS	: Indicate waveform generation status
ITERA.DONE	: Indicate waveform iterations, already done
ERROR FUN.	: Message of ongoing action or function name
ERROR	: Indicate error number (if encountered)
QUIT PROG.	: Exit program
HARDCOPY	: Print a hard copy of the current interactive control panel

APPENDIX A-III: **Interactive Control Panel for the Calibration of Data Acquisition Process via the Microphones: Functions' Description**

PANEL TITLE : MULTIPLE- CHANNEL DAQ (CALIBRATION)

FUNCTION	: **FUNCTION DESCRIPTION**
AI CHAN NUM	: Specify the channel number to calibrate
DITHER	: Enable or disable dithering process during data acquisition
START CALIB	: Activate the panel interactive buttons
SAVE CALIB	: Save the correction factors in a file
EXIST CALIB	: Exit the calibration interactive panel
CH1..TO...CH8	: Show Calibration Correction Factor
GAIN	: Specify the analog input gain for the acquisition process
PRESSURE (DB)	: Indicate interactively the pressure of the applied signal
SET-REF.	: Set desired Calibration reference (dB)
250 Hz	: Filter the acquired signal at 250 Hz octave band
CHECK STATUS	: Start acquisition and calibration process
CALIBRATE	: Calculate amplitude correction factors
LOOP (*)/10	: Specify the number of data blocks (= 8192 samples) to acquire for the calibration process
ERROR FUN.	: Message of ongoing action or function name
ERROR	: Shows error number (if encountered)
QUIT PROG.	: Exit program
HARDCOPY	: Print a hard copy of the current interactive control panel

APPENDIX A-IV : "LOYOLA" Concert hall: Measurement Procedure and Sound Quality Assessment

Eleven measurement locations were selected* (R1-R11, see Fig 6.1(a)). These locations subjectively represent different sound quality impressions as expressed by a typical audience. One source location on the stage floor (referred to in Fig. 6.1(a) by the letter (S) was chosen for the hall excitation; it was located on the centre line of the hall 1.8 m from the leading edge of the stage floor.

All measurements were undertaken with the ventilation system "*OFF*". Under these conditions background noise levels were measured in the centre of the main audience floor. The levels were then compared with *NC* (i.e. Noise Criterion) rating curves. The background noise level was approximately *NC-15* but rising at high frequencies to *NC-20*.

Room-acoustic indicators were calculated in octave frequency bands from 63 Hz to 4 kHz, however discussions and results related to sound quality assessment will be restricted to frequency bands from 125 Hz to 4 kHz and for only eight receiver positions (i.e. **R1** to **R7**, and **R9**) for the purpose of results comparison to averages of measurements conducted by another worker employing *RAMSoft-II*[138] in these locations. An *m-sequence* of length 32767 samples has been used as a sound source to excite the concert space. The excitation time was set to 2.56 seconds.

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* With the assistance of Dr. Mark Corwin, the Musical Dept., Concordia University.

Fig. A-IV.1 shows a comparison of spatially averaged room-acoustic indicators in both hall configurations (i.e. adjustable panels "*ON*" and "*OFF*") indicated by solid lines. Average results of *RAMSoft-II* measurements in both cases are also shown as dashed lines for comparison. Differences at 4 kHz may be due to the different sound sources directivity at high frequencies and/or to differences in the hall air absorption, since measurements were conducted in different times.

Fig. A-IV.1(a) presents a comparison of spatially averaged *EDT* values in both hall configurations . The *EDT* value at mid-frequencies is found to be 1.20 seconds which is below the recommended optimal range (1.6-1.9 sec or 2.1 sec)[2,5,16,28], however it may be considered satisfactory when the hall volume is taken into account. *EDT* mean difference values in both hall configurations; panels "*OFF*" and "*ON*" indicate slight influence for rotating the side wall reflective strip panels. The values exhibit mean mid-frequency difference in the order of 80 msec. The standard deviation of *EDT* indicated a reasonable spatial variation but higher at 125 Hz octave band frequency (≈ 200 msec) in both cases compared to other frequency bands.

Fig. A-IV.1(b) compares the hall RT_{20} spatial average values and indicates that the measured mid-frequencies RT_{20} values are below the 1.9 seconds optimum. The changes in RT_{20} mid-frequency values are not significant (≈ 70 msec) when the adjustable panels are "*ON*". Mean difference is in the order of 85 msec. The spatial variation of RT_{20} values was found to be small with higher variation at 125

Hz octave band (≈ 100 msec).

Fig. A-IV.1(c) shows the hall spatial average values of Clarity, C_{80} with the side walls panels "*ON*" and "*OFF*". Mid-frequency value (≈ 1.5 dB) is in the optimum range for orchestral and ensemble music. It can be also seen that minor changes in the order of 0.6 dB to C_{80} in the mid-frequency range (500-1000 Hz) are present when side wall panels are "*ON*". Seat-to-seat variation of C_{80} is relatively high and indicates that measured Clarity is not evenly distributed throughout the hall with a mean variation of around 1.5 dB and rises to 2.5 dB at low frequencies.

Fig. A-IV.1(d) compares the hall spatial average values of the Centre Time, T_S when the panels are "*ON*" and "*OFF*". T_S also indicates clarity and blend of music but with the elimination of doubt about the upper limit of integration that is 50 msec as used in the Definition, D_{50} and Running liveness, *R* or 80 msec used in Clarity, C_{80}. Insignificant differences of the order 10 msec are observed between the two hall conditions. The mid-frequency value i.e. 100 msec is within the optimal 144 msec value range.

Fig. A-IV.1(e) compares measured spatial mean values of the Relative Strength, *G* in the hall and shows that *G* values are around 0.75 dB greater when the side wall adjustable panels are "*ON*" than in the "*OFF*" condition. The mid-frequency average is around 7.7 dB and may be considered acceptable when the volume of the hall is taken into account. The mid-frequencies high average values mean that the sound can be quite loud in the hall and that music can be produced with large dynamic range. Values in both cases agree well with measurements results of *RAMSoft-II* in the frequency range from 250 Hz to 2 kHz but with 1.5 dB less at 125 and 1.8 dB at 4 kHz; this can be attributed to errors associated with determining the sound source power at low and high frequencies.

Fig. A-IV.1(f) presents the average of mid-frequency (500-1000 Hz) Relative Strength, *G* versus source-receiver distance based on source (S) location; measurements are plotted in relation to source-receiver distances from front seats to the rear of the hall. *G* values decrease as expected from front to the rear of the main audience floor. The straight lines fitted to the data in both cases show a decrease of 3.0 dB for 10 m distance which is slightly higher than decreases found in other halls, however this can be attributed to the ceiling diffusing properties and the gradual decrease of its height in relation to seating floor. Measured values agree well with predicted values from Barron's theory[14].

Fig. A-IV.1(g) shows the measured values of LEF_5 in both hall configurations and indicates a mid-frequency increase in the order of .025 when the side panels are "*ON*". Higher increases of around 0.6 in the high frequency range (2-4 kHz) are also observed.

RASTI ratios evaluation indicates an average "*Fair*" rating for speech intelligibility but with values very close to the lower limit of "*Good*" rating (i.e. 0.6) particularly at front locations. The percentage of Articulation Loss of Consonants, AL_{cons} is also considered and found to be between 7% and 9% and therefore support the overall speech intelligibility prediction. The mid-frequency spatial average value of SNR_{95} is found to be around ≈1.0 dB, and assuming that the useful speech level should be 65 dB then from the curves depicted in Fig. A-IV.2 [5], the percentage of intelligible syllables can be seen to be more than 90% which also supports the *RASTI* prediction of a reasonably "fair-to-good" speech intelligibility.

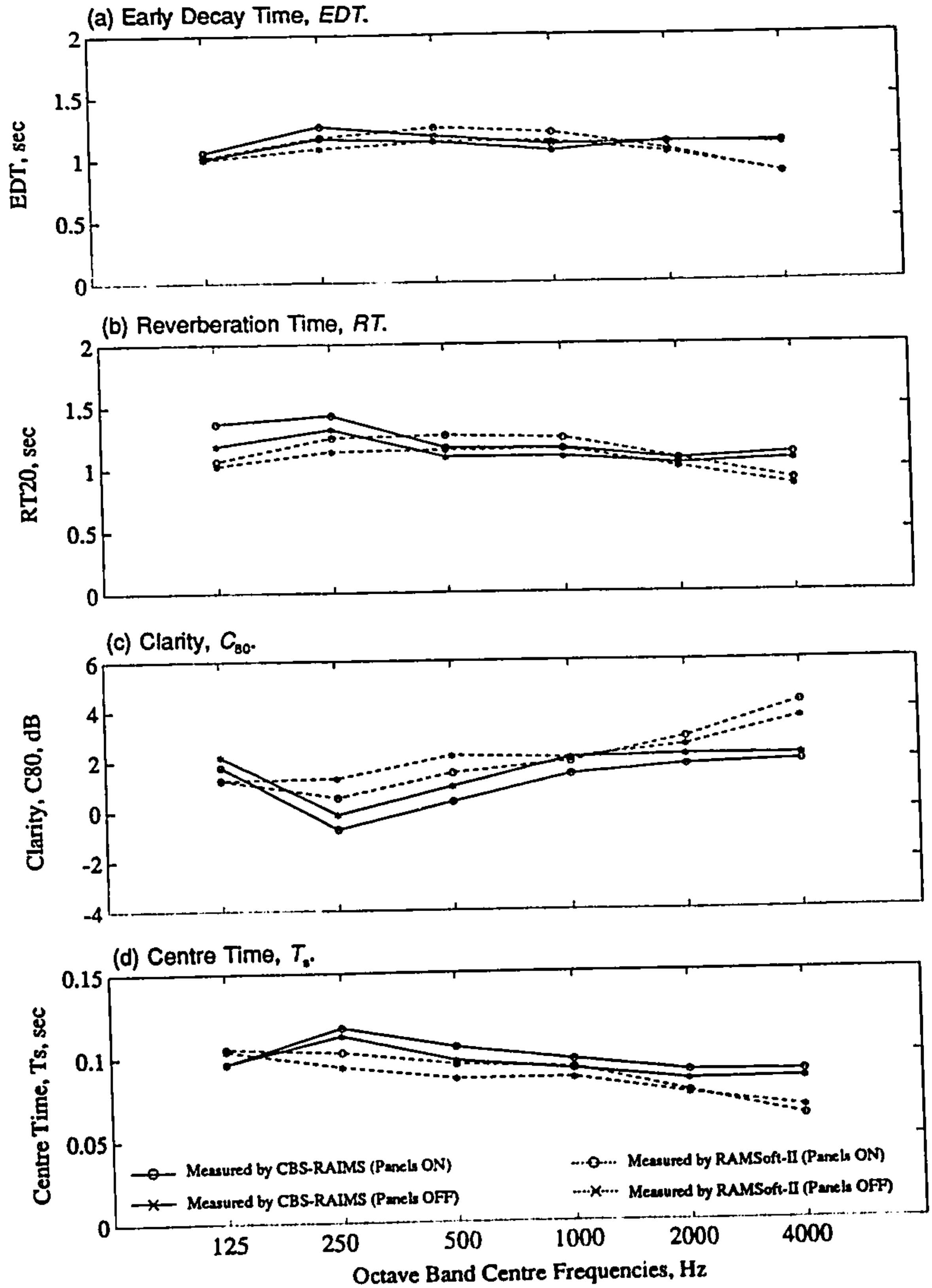
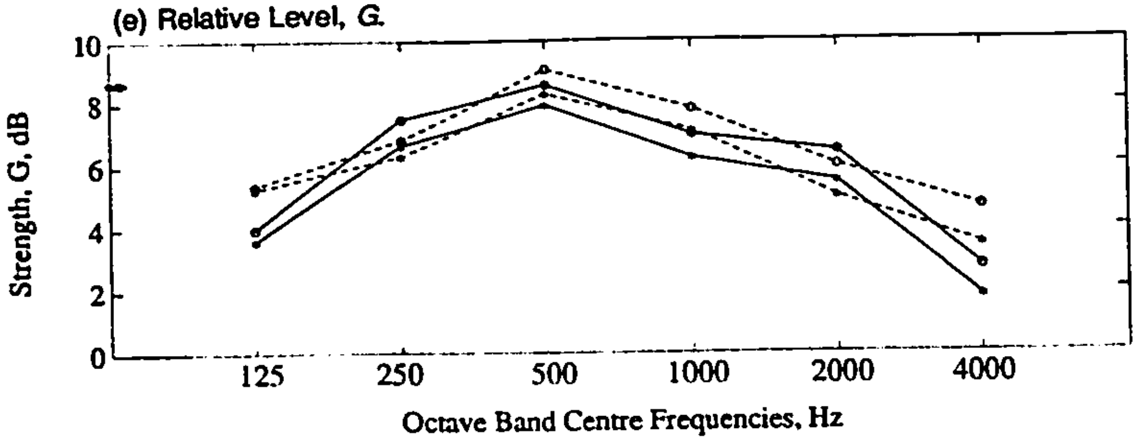
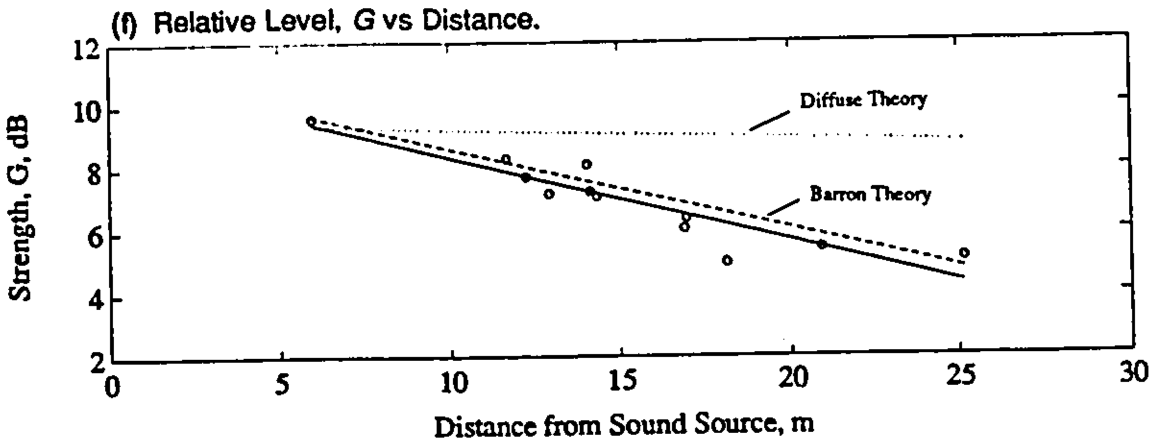


Fig A-IV.1 Comparison of spatially averaged room-acoustic indicators evaluated for the "**LOYOLA**" concert hall, adjustable side wall panels "*OFF*" and "*ON*"

(Note: Differences of results at 4 kHz are most likely due to differences in air absorption since measurements were conducted at different times, and/or due to the different directional characteristics of the employed sound sources)





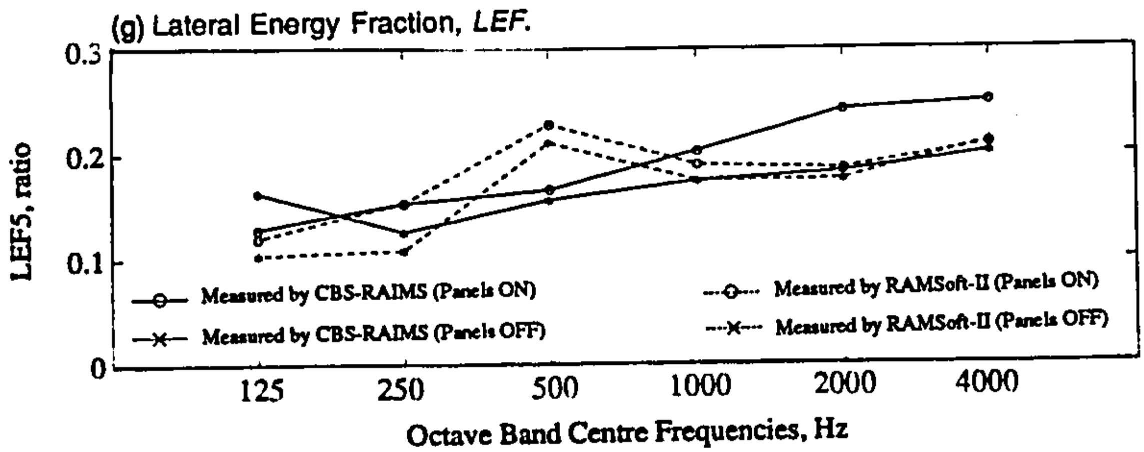


Fig. A-IV.1 Continued.

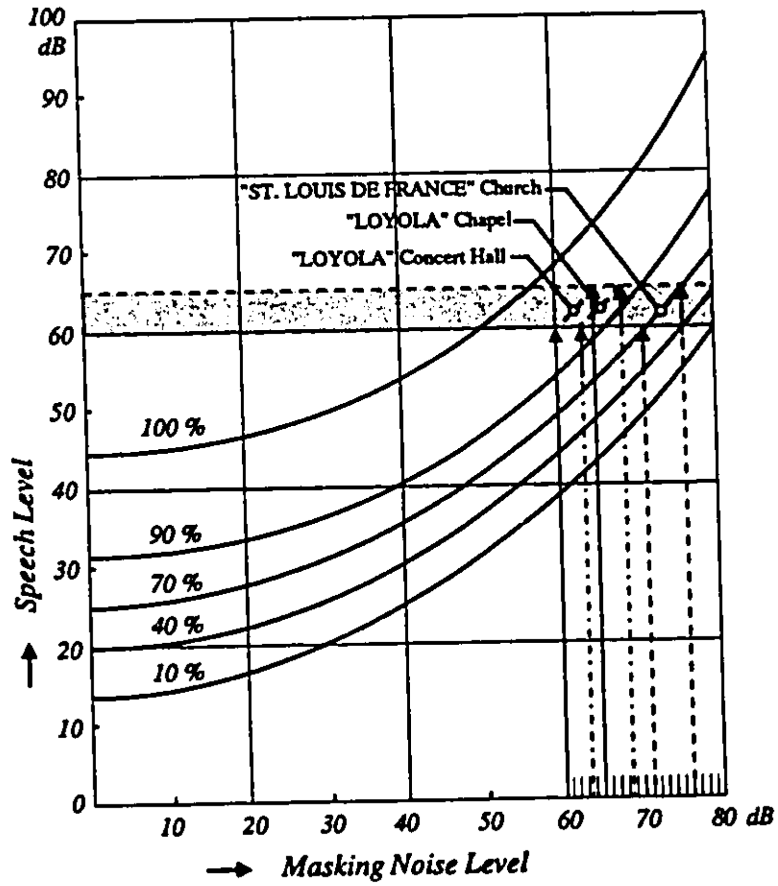


Fig A-IV.2 Speech intelligibility (rate of intelligible syllables) as a function of Speech level and noise level [reference[5]].

APPENDIX A-V : "LOYOLA" Chapel: Measurement Procedure and Sound Quality Assessment

Since the chapel enclosure design is symmetric, eight measurements locations have been selected on one side of the space with one location in the balcony. These seat locations cover different congregation seating zones and are shown in Fig 6.4(a); each is indicated by numerated letters (P) and (B) according to the measurement order and will be subsequently used to identify measurement results in the following text and figures. An *m-sequence* of length 32767 samples has been used as a sound source to excite the chapel space. The excitation time was set to 2.56 seconds in order to cover the reverberation time of the chapel. The loudspeaker location; referred to on the chapel plan with the letter (S) was located on the centre line of the chapel sanctuary 1.10 meters from the leading edge of the sanctuary floor and 3.50 meters from its back wall.

The chapel background noise levels were measured in the centre of the main congregation seating area. The measured levels were then compared with Noise Criteria, *NC* rating curves and found approximately to be *NC-25* but rising to *NC-30* at frequencies above 1000 Hz. The preferred range of Noise Criteria is *NC-20* or less, therefore the measured background noise is found to be slightly above that normally recommended, but not critically so. To evaluate the sound quality all subsequent measurements results were undertaken when the chapel was empty.

Fig. A-V.1(a) presents a spatial average of measured *EDT* values for the eight selected locations and indicated on all subsequent graphs with a continuous line. The standard deviation is also shown as a dashed line; this indicates the overall spatial variations about the chapel spatial average. The *EDT* average value at mid-frequencies is found to be 2.10 seconds which is less than the recommended optimum (i.e. [3.3 x 0.9] = 2.9 sec) in this case when the chapel occupancy condition and volume are taken into account). In general the *EDT* spatial variation is considered reasonable compared with the overall average but higher (i.e. ± 0.48 to ± 0.25 sec) at low octave frequencies (63-250 Hz) compared to other frequency bands.

Fig. A-V.1(b) shows the chapel measured mid-frequency RT_{20} values the spatial average value is 2.20 seconds; this is less than the upper limit of the suggested optimal 3.3 s for the chapel empty based on its volume[3,28]. However the RT_{20} value is within the recommended range for chorus or organ music (i.e. from 2.0 to 2.6 seconds). The spatial variation of RT_{20} values is seen to be similar to the *EDT* variation with higher variation at 63-250 Hz octave bands but less than the *EDT* spatial variation found in the same frequency range. The most reverberant measurement locations are found to be **P7** and **B1** as might be expected, less reverberant locations are found to be the nearest to the sanctuary floor area such as **P1** and **P2**. It should be also noted from the graph that the low RT_{20} at 2 and 4 kHz is due to high frequency attenuation caused by the chapel air volume. At low frequencies high RT_{20} values can be attributed to the walls' wooden panels reflective characteristics. However, considering the dependence of the *RT* on frequency in a church it is desirable with regard to the lowest register of the organ to have *RT* increased toward the low frequencies and this is clearly the case at hand where the chapel low frequencies *RT's* are more pronounced than in the middle or high frequency bands.

Fig. A-V.1(c) presents values of the indicator Definition, D_{50}; it shows good speech and music definition particulary at frequencies most important to speech intelligibility (500 Hz to 2 kHz). The average value (500 Hz, 1 and 2 kHz) is 0.25 which is less than the 0.34 optimal value[3]. However, speech intelligibility is expected to be higher than measured values due to the fact that during sermons and prayers, the speaker bench location is set near to the congregation on an elevated floor to facilitate the direct sound reaching all listener positions. Considerable spatial variation of D_{50} between locations at low frequencies, particulary at 63 and 125 Hz octave frequency bands can be seen but these frequencies are below human speech spectrum (i.e. from 300 Hz to 3 kHz) and therefore will not critically affect speech intelligibility.

Fig. A-V.1(d) shows measured values of Clarity, C_{80}. The mid-frequencies value is -2.9 dB and is still within a tolerable range for ensemble or symphonic music (i.e. -3 dB)[5]. The spatial variation is in the order of ± 2 dB particulary at frequency bands 125, 250, and 500 Hz. This means that measured Clarity is not evenly distributed throughout the congregation area with a maximum variation of ±2.6 dB.

Considering the C_{80} predicted values calculated by employing both the ideal diffuse exponential decay theory and a revised theory proposed by Barron and Lee[14]. Spatial average of C_{80} mid-frequencies value is 1.5 dB below predicted when the diffuse theory is considered and 1.0 dB less than value calculated by Barron theory.

Fig. A-V.1(e) depicts measured values of the Centre Time, T_S. Mid-frequency average value, i.e. 185 msec is higher than optimal (140-144 ms) and as can be seen the T_S value increases beyond the optimal at 250-1000 Hz octave bands by approximately 30%. Spatial variation of T_S is found to be within a range of 6 to 40 msec.

Fig. A-V.1(f) shows measured values of the Relative Strength, G. Theoretical values have also been calculated assuming an ideal diffuse sound field and employing Barron's theory. The mid-frequency mean value is below mean ideal value by about 1.0 to 2.0 dB. The mid-frequency average value is close to the predictions of Barron's theory and 2.5 dB below the value of the ideal diffuse sound field. The 8.0 dB mid-frequency average value implies that music can be reasonably loud in the congregation floor and therefore can be produced with large dynamic range.

Figs. A-V.1(g) shows the average G of the mid-frequency bands (500-1000 Hz) versus source-receiver distance based on source (S) location and plotted in relation to source-receiver distances from front seats to the rear of the chapel hall. *G* values decrease as expected from front to the rear of the main congregation floor. The straight lines fitted to the data show a decrease of 0.8 dB/10 m distance which is less than decreases (slopes) found in other halls but closely agrees with predicted values from Barron revised theory which is reported to be more accurate.

The spatial average of calculated *RASTI* ratios results in a "*Fair*" rating for speech intelligibility at all measurement positions with almost a constant value of 0.45. The percentage of Articulation Loss of Consonants, AL_{cons}, was found around 15% and confirms the speech intelligibility prediction. However, speech intelligibility rating is expected to increase to the "*Good*" rating due to the fact that during sermons and prayers, the speaker bench location is set near to the congregation on an elevated floor to facilitate the direct sound reaching all listener positions; (see Fig. 6.4(a)) for the chapel seating arrangement.

Also with the use of the measured mid-frequency indicator SNR_{95}, the spatial mid-frequency average value of SNR_{95} is found to be -3.3 dB; assuming that the useful speech level should be 65 dB then the masking "noise" level is 68.3 dB, from the curves depicted in Fig. A-IV.2 [5], the percentage of intelligible syllables can be seen to exceed 90%, this also supports the *RASTI* prediction of a reasonably good speech intelligibility.

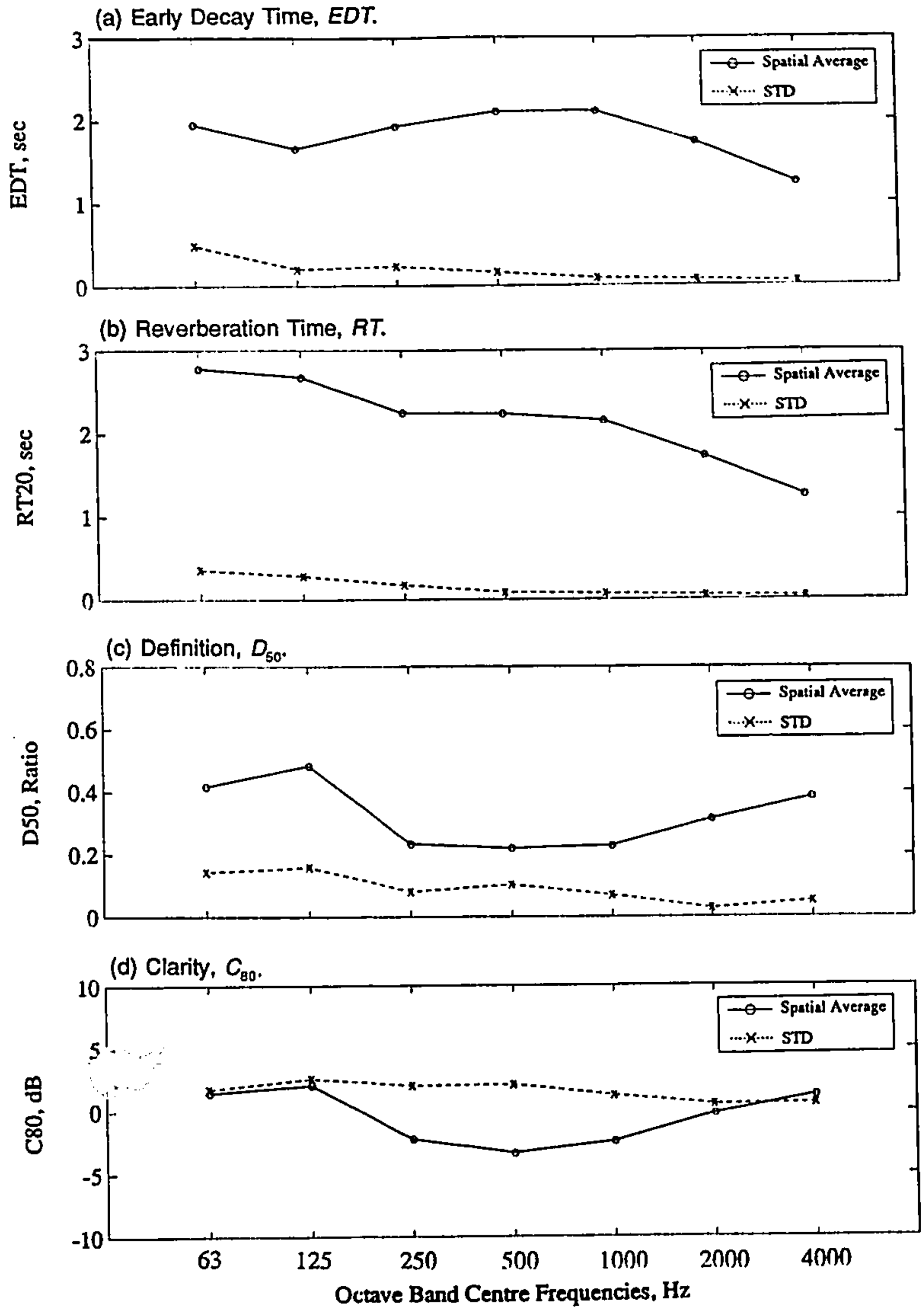
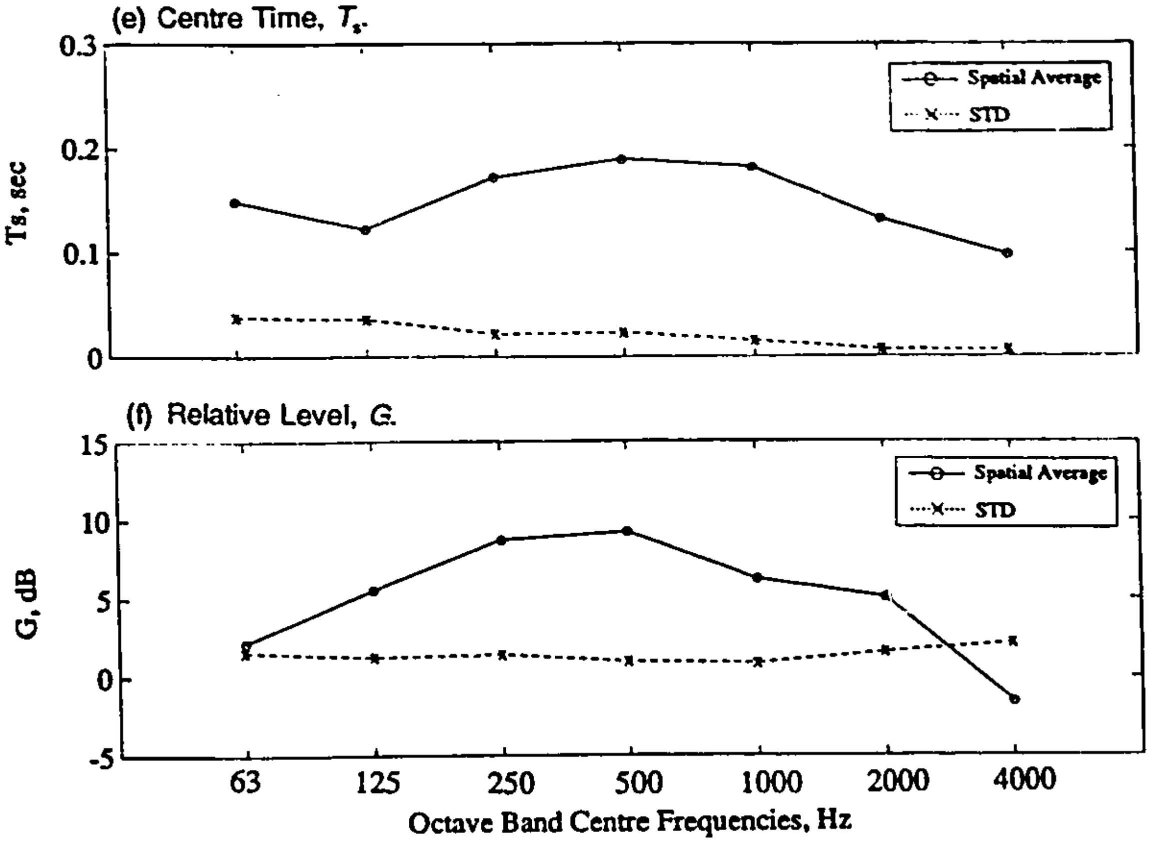


Fig A-V.1 Space averaged room-acoustic indicators evaluated for the "**LOYOLA**" chapel.



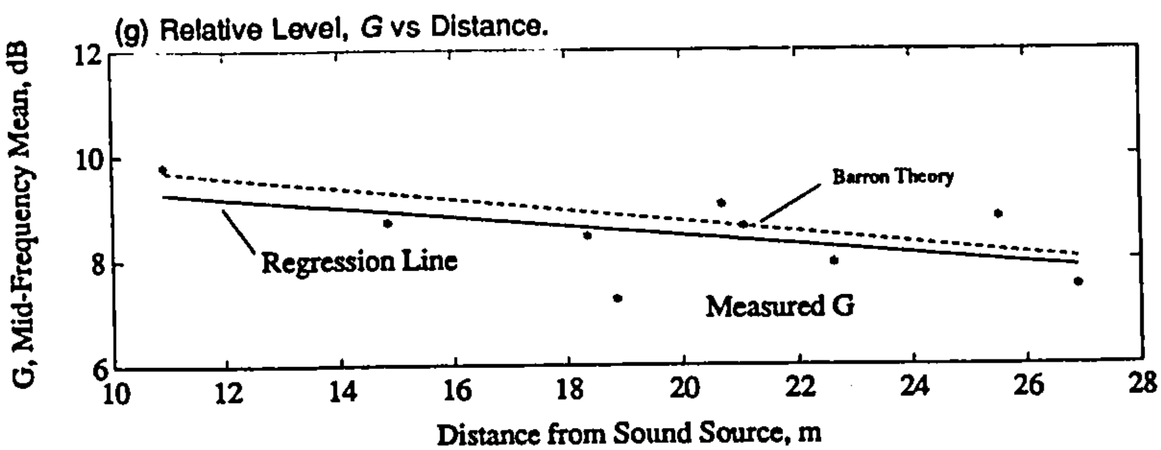


Fig A-VI.1 Continued.

APPENDIX A-VI : "ST. LOUIS DE FRANCE" Church: Measurement Procedure and Sound Quality Assessment

Eight measurement locations have been selected on one side of the church symmetric space. These seat locations cover different congregation seating zones and are shown in Fig. 6.7(a); each is indicated by a numerated letter (L) according to the measurement order and will be subsequently used to identify measurement results. A *m-sequence* of a length 65535 samples has been used as a sound source to excite the church space. The excitation time was set to be 5.11 seconds in order to cover the long reverberation time of the church The loudspeaker location; referred to on the church plan with the letter (S) was located on the centre line of the church central nave 3.5 meters from the leading edge of the sanctuary floor and 5.0 meters from its back wall.

The church background noise levels were measured in the centre of the main congregation seating. The measured levels compared with *NC* rating curves showed that the background noise level are approximately *NC-25* but rising to *NC-30* at frequencies of 1 and 2 kHz, these are extremely important frequencies for speech perception. For a large church and excellent listening conditions (i.e sufficiently quiet background) , the preferred range of *NC* is less than *NC-20*, therefore the measured background noise level is found higher than normally recommended but not critically so.

To evaluate the sound quality in the church all subsequent measurements were undertaken when the church was empty and with the electroacoustical sound-reinforcement system "*OFF*". Fig. A-VI.1(a) presents the spatial average of the measured *EDT* values for the eight selected locations in addition to the standard deviation. The *EDT* spatial average value at mid-frequencies is found to be 4.5 seconds which is higher than the recommended optimum (i.e. 3.9 sec in this case when the church occupancy conditions and large volume are taken into account[3]). In general the *EDT* spatial variation is considered reasonable compared with the overall average but higher at low octave frequencies (63-250 Hz) compared to other frequency bands.

Fig. A-VI.1(b) depicts the church spatial average of RT_{20} values and indicates that the measured mid-frequencies RT_{20} spatial average value is 4.9 seconds; this is longer than the upper limit of the optimal 4.3 sec for the church empty. The spatial variation of RT_{20} values is seen to be similar to the *EDT* variation with higher variation at 63-250 Hz octave bands. The most reverberant measurement locations are found to be **L1** and **L7** as might be expected, less reverberant locations are found to be the nearest to the sanctuary floor area such as **L5**, **L6**, and **L8** locations. It should be also noted from the graph that the low *RT* at 4 kHz is due to high frequency attenuation caused by the church air volume. At low frequencies low *RT* value can be attributed to the walls' cork panels absorbent characteristics.

Fig. A-VI.1(c) shows the measured average values of Clarity, C_{80}. The mid-frequencies value is (-10.5 dB) and is well outside the optimum range for ensemble or symphonic music (i.e. ± 3 dB). The spatial variation is to some extent high, particulary at frequencies 250, 500 and 1000 Hz. This means that measured Clarity is not evenly distributed throughout the congregation area with a maximum variation of ± 3.0 dB. The theoretical C_{80} values were also calculated by employing both the ideal diffuse exponential decay theory and a revised theory proposed by Barron and Lee[14]. Clarity average values were found below predicted at most frequency bands such as 250, 500 Hz, and 1 kHz.

Fig. A-VI.1(d) shows the spatial average values of the Centre Time, T_S. The mid-frequency spatially averaged values are higher than optimal (140-144 ms) and as can be seen the T_S value increases dramatically beyond the optimal at 250-2000 Hz octave bands. Spatial variation of T_S is found to be within a range of 25-60 msec.

Fig. A-VI.1(e) presents the measured values of the Relative Strength , G. Theoretical values have also been calculated assuming ideal diffuse sound field and employing Barron's revised theory. The mid-frequencies mean value is below mean ideal values by about 1 to 2 dB. Mid-frequency average values are higher than the suggested optimal, however they are close to the predictions of Barron's theory and to values of the ideal diffuse sound field particularly, at 500 Hz and 2 kHz.

Fig. A-VI.1(f) shows the measured value of the direct sound level at the measurement locations. Direct sound level is important to both sound source localization and speech intelligibility. Significant attenuation of the direct sound at 250 Hz (176-356 Hz) can be seen; this can be readily attributed to the attenuation due to sound propagation at grazing incidence over the congregation horizontal seating. This is referred to as the Seat dip phenomena and is very detrimental for the congregation especially with respect to music because the sound that they hear is a combination of the direct and reflected waves i.e. in this case the sum of the two are nearly equal and out of phase sounds that might completely cancel the direct wave, particulary when the seats are not occupied.

The average of mid-frequency (500-1000 Hz) relative level, *G* is investigated in relation to source-receiver distances from front seats to the rear of the church hall. Strength values decrease as expected from front to the rear of the main congregation floor. The straight lines fitted to the data show a decrease of 1.5 dB/10 m distance which is less than decreases (slopes) found in other halls but closely agrees with predicted values from diffuse sound theory.

Fig. A-VI.2 the calculated *RASTI* ratios versus source-receiver distance. The spatial average of both indicators result in a "*Poor*" rating for speech intelligibility at all measurement positions but with higher values close to the "*Fair-Poor*" limit at locations **L5**, **L6** and **L8**. The figure also depicts the percentage of Articulation Loss of Consonants, AL_{cons} and confirms the speech intelligibility deficiency at locations L1, L2, and L7 more so than other measurement positions. This can be attributed to excess reverberance at these particular locations when the church is empty.

The mid-frequency spatial average value of SNR_{95} is found to be -11 dB, and assuming that the useful speech level should be between 60 to 65 dB then the equivalent masking "noise" level is 71 to 76 dB, from the shown curves shown in Fig. A-IV.2 [5] the percentage of intelligible syllables can be seen to be about 70% which also supports the *RASTI* prediction of poor speech intelligibility.

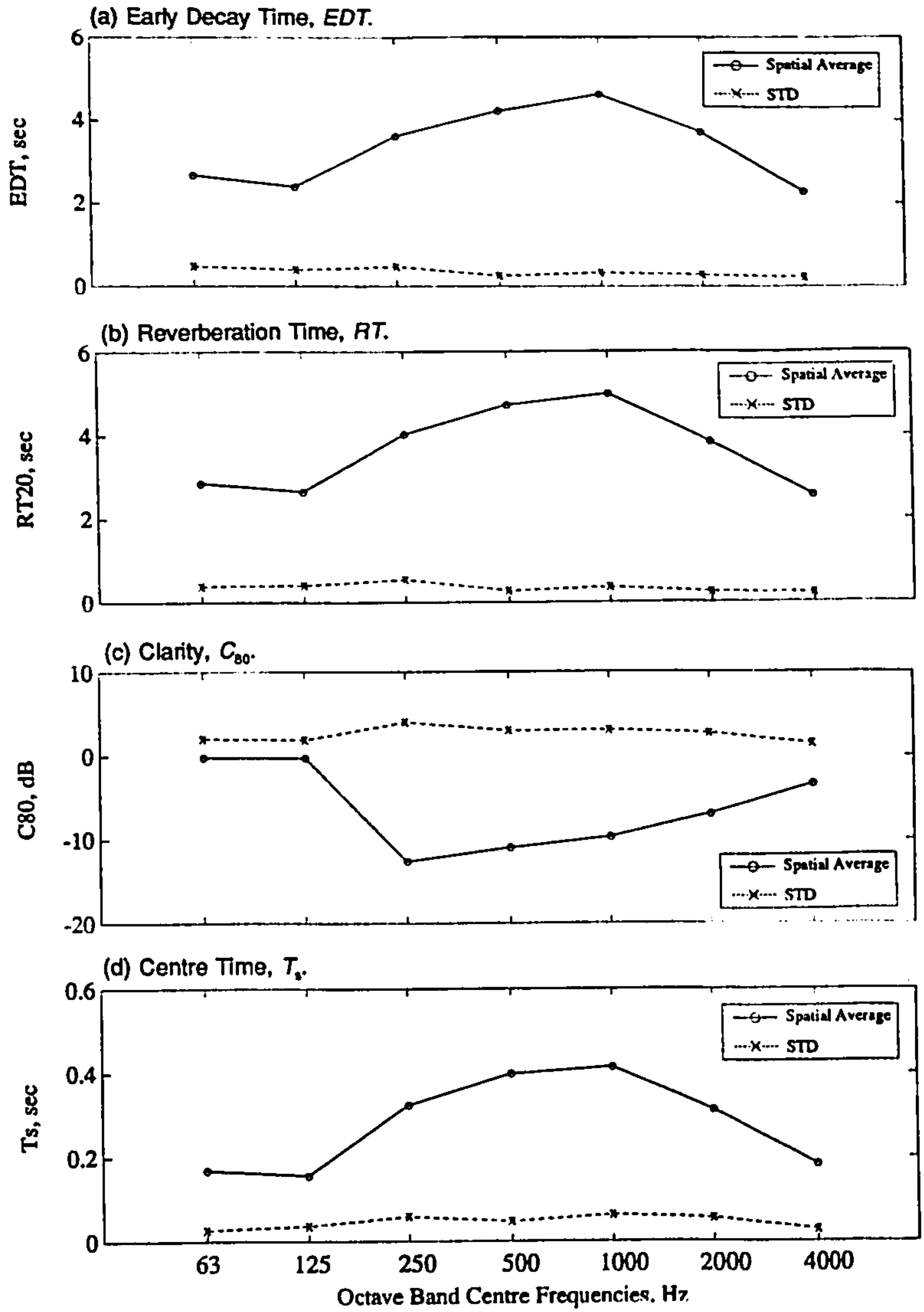


Fig A-VI.1 Space averaged room-acoustic indicators evaluated for the "**ST. LOUIS DE FRANCE**" church.

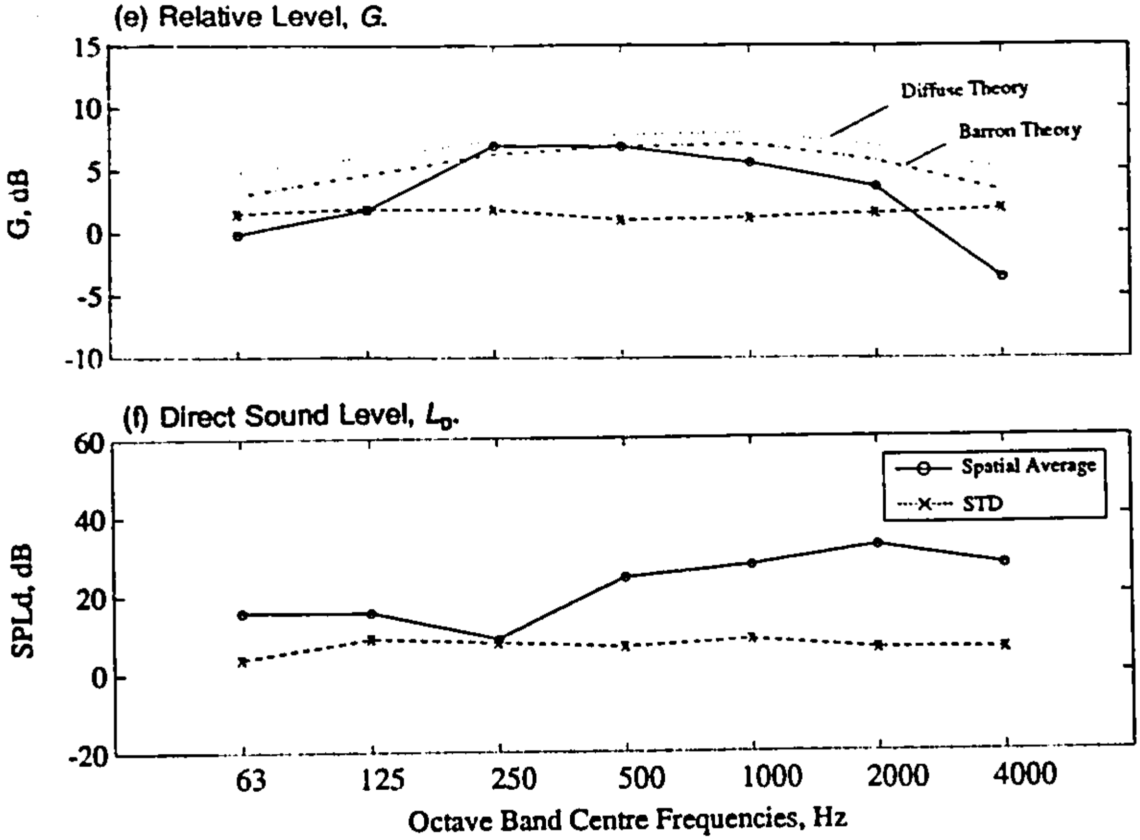


Fig. A-VI.1 Continued.

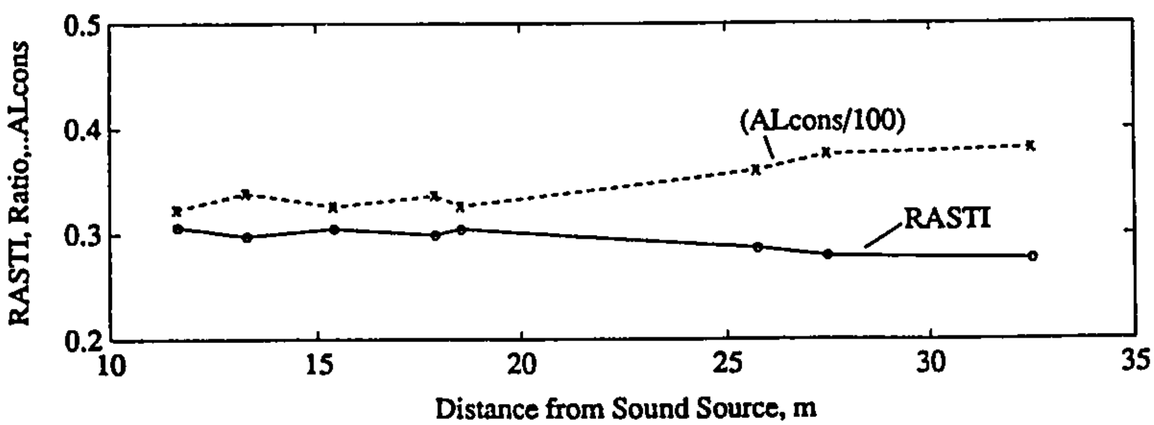


Fig A-VI.2 *RASTI* and AL_{cons} values versus source-receiver distance evaluated for the "ST. LOUIS DE FRANCE" church.