DESIGN AND OPTIMISATION OF VOICE ALARM SYSTEMS FOR UNDERGROUND STATIONS

By

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Declaration:

I declare that this thesis is the original work of the author, Mr Luis Gomez Agustina, except where otherwise specified, or where acknowledgement is made by reference. It was carried out at the Faculty of Engineering Science and the Built Environment *(FESBE)*, London South Bank University, under the main supervision of Dr Stephen Dance. The work has not been submitted for another degree or award of any other academic or professional institution.

Thesis author's signature :

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Glossary of terms and abbreviations

Term	*Description*
%Alcons	Percentage articulation loss of consonants. Parameter used for the objective assessment of speech intelligibility.
3D	Three dimensional representation of an object or space
A weighing	Filter network that modifies the response of a sound level meter with frequency in approximately the same way as the human ear, in order to assess subjective loudness
Absorption coefficient	Frequency dependent ratio of sound energy absorbed by an object/surface to the total sound energy randomly incident on its surface, with 0 indicating a totally reflective material and 1. Usually indicated by the greek letter α (alpha)
Acoustic absorption	Measured in Sabins, defined the loss of sound energy when impinging a surface material, or due to its transmission outside the space considered. It is the product of the material's absorption coefficient and its surface area.
AI	Articulation index. Method for the calculation of speech intelligibility or privacy defined in ANSI 3.5 (1969) that accounts for the influence of background noise on the relevant speech signal at the receiver position (but not distortions in the time domain, e.g reverberation, echoes)
Aim	Direction indicated by the centre axis of the loudspeaker cone
Air sound absorption	The act of sound waves energy being reduced by the transmission medium (e.g air) or on impinging surfaces or objects.
Aka	Also known as
ANSI	American national standards institute
Auralisation	The technique of creation and reproduction of sound on the basis of computer data. With this tool is it possible to predict the character of sound signals which are generated at the source and modified by reinforcement, propagation and transmission in systems such as rooms and buildings.
Background Noise	Noise present in the space under consideration in the absence of the sound signal of interest (sound signal)
BGNL	Background noise level (dB) sound pressure level of the background noise
BS	British Standard institution
C	Clarity index (dB) it describes the clarity of the signal on propagation, at a receiver position. This energy ratiois defined as the ratio of early to late arriving sound. The index C50 relates to the time threshold for defining late arriving sound (in milliseconds). A 50ms limit is commonly used for speech intelligibility applications.
C50	Ratio (dB) of early (useful) arriving sound energy (within 50 ms of direct sound), to late arriving sound energy (after 50 ms of direct sound)
CAD	Computer aided design. Refers to software tools for 3D model generation.

CIS	Common intelligibility scale. Included in BS EN 60849, CIS relates a number of speech intelligibility measures on a single scale, also including common subjective assessment methods.
Convolution	Convolution is a mathematical operation on two functions producing a third function as a modified version of one of the original functions. The convolution of two waveforms in the time domain becomes the multiplication of their spectra in the frequency domain and viceversa.
Coverage angle	The coverage angle of a loudspeaker is the angle within which the SPL is no more than 6 dB below the normalized on-axis level for a given bandwidth or frequency centre
CVC	Consonant-vowel-consonant. Defines a sequence of words as arranged by their starting letter. CVC word lists are used in subjective speech subjective intelligibility assessments
D	Definition (%). D is an energy ratio (early energy / total energy) relating to the distinctness of sound in a room. The measure defines a condition as a percentage. The index D50 relates to the time threshold for defining early or useful arriving sound (in milliseconds). A 50ms limit is commonly used for speech intelligibility applications.
D50	Ratio of early arriving sound energy (useful) (within 50 ms of direct sound) to total arriving sound energy (after 50 ms of direct sound)
Decay curves	Decay of sound pressure level as a function of time at one point in the room after the source of sound has ceased. Used to determine reverberation time parameters
Deterministic	The concept by which every state, condition or event is the inevitable consequence of antecedent states. In a deterministic system no randomness is involved in the development of future states of the system.
DI	Directivity Index (dB), DI = 10log 10*Q. This parameter quantifies the directionality of a sound source. See “Q” below
Diffuse field	A sound field in which the sound pressure level is the same everywhere and the flow of energy is equally probable in all directions.
Dispersion (loudspeaker)	See coverage angle
Distributed sound system	Arrangement by which loudspeakers (usually equal) are fed with the same signal and are evenly spaced throughout the space, each covering an specific area
DRR	Direct to-reverberant energy ratio (dB). SPL difference between direct sound and reverberant sound components at a receiver
Echo	A delayed return of sound that is perceived by the ear as a discrete sound image arriving at the listener some time after the direct sound.
Echogram	A record of the very early reverberatory decay of sound in a room. An echogram is an approximation to the square of a RIR used to display early reflection arrivals
EDT	Early Decay Time (seconds). Reverberation time derived from the first 10 dB of level of sound energy decay, extrapolated to a 60dB decay.
ETC	Plot which shows how sound energy amplitude input into a room (in dB) decays over time.
Free field	An environment in which there are no reflective surfaces within the frequency region of interest and the sound is isotropic and homogenous
Frequency response	In a loudspeaker, it refers to the frequency range over which the loudspeaker responds, usually with a tolerance range for level variation.
G	Strength (dB). G, Measure relating to the overall energy transferred from the sound source to the receiver after subtracting the influence of the direct field.
Gain	The increase in level of a signal produced by an amplifier.
IP	Loudspeaker Ingress Protection rating against dust and water,
IR	Impulse response. Plot as a function of time of the sound pressure level in a space as a result of excitation of the room by a Dirac delta function
ISM	Image source acoustic modelling technique
ISO	International organization for Standarisation
JND	Just noticeable difference. Determines the human perception threshold for changes in a particular condition or parameter on a subjective basis. The JND is also sometimes known as the ‘limen’.

LAeq	A weigthed calculation based on Leq per octave of the frequency value of the notional steady continuous sound pressure level that would, over a given period of time, deliver the same sound energy as the actual fluctuating sound over the same period (dBA)
Limen	See JND
Loudness	The subjective judgement of intensity of a sound by humans. Loudness depends upon the sound pressure and frequency of the stimulus.
Loudspeaker sensitivity	Related to the efficiency of the unit, determines the SPL able to delivered at a certain distance for a certain input electrical power. Generally given as a SPL measured at 1 meter on axis with a 2.83 volt input (1 watt @ 8 ohms).
LTI	A Linear Time Invariant system in the electroacoustic domain is characterised by a linear response and its invariability in time. In practical terms those conditions involves the absence of non-linear distortions (e.g. loudspeaker total harmonic and intermodulation distortions) and movement of air or objects and/or changes in environmental conditions during a measurement.
LU	London Underground (also London Underground Limited, LUL)
Lw	The sound power level, in dB, at which a source produces sound energy per unit of time, usually given in octave bands. A power expressed in dB above the standard reference level of 1 picowatt. Sound Power Level = 10 log10 (W/W0), where W is the emitted power and W0 is the reference power (10-12 W).
m(F)	Function describing the reduction in modulation index between the input and the output of a communication system or transmission path
MFP	Mean Free Path, For sound waves in an enclosure, it is the average distance travelled between successive reflections.
MLS	Maximum length sequence. Type of test signal used in acoustic measurements, described as a deterministic periodic pseudo-random binary sequence with similar properties as a true random white noise signal. Correlation function is a Dirac impulse.
MRT	Modified rhyme test, subjective method for the assessment of speech intelligibility.
MTF	Same as m(F)
MTI	Modulation transfer index. Calculated as the average TI per octave band k. A direct averaging of MTI values will result in a basic STI, a number of corrections are normally applied for a more realistic STI.
Noiseless	Acoustic condition when the SNR is very high so the noise contribution is not existent or negligible
PA	Public Address system. Normally not linked to the fire/emergency alarm system so not for life critical purposes.
PB word lists	Phonetically balanced word list which comprise purposely designed word lists forming the basis for a subjective methodology for the assessment of speech intelligibility.
Point source	Sound source sufficiently small compared to the wavelength so that the radiation pattern can be considered omni-directional and the propagation inverse square law applies
Power tapping	Attenuation system (impedance transformer) built into PA or VA loudspeaker which limits the maximum output acoustic power
Q	Ratio of the sound intensity level in the direction of interest, to the sound intensity level averaged over all angles.
R	Pearson's correlation factor (also represented as *r*)
RASTI	Rapid acoustics STI, Index obtained by a condensed version of the STI method (using only 500 Hz and 2 kHz octave frequency bands), Developed for direct natural communication between persons without using amplification. Now obsolete.
RIR	Room impulse response. Plot as a function of time of the sound pressure level in a room as a result of excitation of the room by an unit pulse (Dirac delta function).
RT	Reverberation time. RT (seconds) is defined as the time taken for a sound to decay by 60dB (RT60) after the excitation source has ceased. A number appended indicates the decay range (in dB) period considered. Examples are RT20, RT30, RT60.

RT20	The reverberation time in seconds calculated from the 20 dB decay range from -5 dB to -25 dB below the initial level (0dB)
RT30	The reverberation time in seconds calculated from the 30 dB decay range from -5 dB to -35 dB below the initial level (0dB)
Scattering coefficient	Defined as the ratio between the acoustic energy reflected in non-specular directions and the totally reflected acoustic energy (range between 0 and 1).
SII	Speech Intelligibility Index. Objective method for prediction of intelligibility (or privacy) based on the Articulation Index and defined in ANSI 3.5 (1997) representing the proportion of speech cues available to the listener. Accounts for the influence of background noise on the relevant speech signal at the receiver position but not distortions in the time domain.
SIL	Difference between A-weighted (dB) speech level and the arithmetic average of sound pressure levels of ambient noise in four octave bands with central frequencies of 500Hz, 1kHz, 2kHz and 4 kHz
SNR	Signal to noise ratio (dB). It is a fundamental parameter in acoustic measurements and relevant to speech intelligibility predictions
Sound diffusion	The act of sound waves spreading out over a wide area (non specularly), reflecting off a convex or other uneven surface.
Sound reflection	The return of a sound wave from a surface.
Specular	Geometrical reflection of a sound wave in which the angle of incidence is equal to the angle of reflection
Speech Intelligibility	The degree to which speech can be understood. With specific reference to speech communication system specification and testing, intelligibility denotes the extent to which trained listeners can identify words or phrases that are spoken by trained talkers and transmitted to the listeners via the communication system
SPL	Sound Pressure Level (dB); The sound pressure level of a sound in decibels, is equal to 20 times the logarithm to base 10 of the ratio of the RMS sound pressure to the reference sound pressure 20 mPa (2 ×10-5 Pa).
SPL(A)	A weighted Sound pressure level (dBA). A single figure value of the sound pressure level using the A-weighting network that modifies the response of a sound level meter with frequency in approximately the same way as the human ear, in order to assess subjective loudness.
std	Standard deviation, also denoted as greek letter σ is a statistical measure of dispersion of a set of values from the mean. Square root of the variance of a set of values
STI	Speech transmission index. Objective method for prediction and assessment of speech intelligibility. It is based on the determination of m(F) values at 98 data points. Unitless quantity representing the transmission quality of speech with respect to intelligibility. Accounts for noise interference and distortions in the time domain (reverberation) (see appendix B.2)
STIPA	Speech transmission index for public address systems. A subset of full STI specifically developed for practically predicting/assessing speech intelligibility for transmission channels involving a electro acoustic sound system. Normally determined by means a portable meter in conjunction with its purposely developed test signal (see appendix B.2)
Stochastic	Refers to systems whose behaviour is intrinsically non-deterministic, sporadic, and categorically not intermittent (i.e. random).
TI	Transmission index. Determined from the apparent S/N ratio, specific for octave band k and modulation frequency f. It forms the basis for the derivation of MTI and subsequent STI
VA	Voice Alarm system, normally connected to the fire/emergency alarm systems of a building or underground space normaly specified for life critical applications.
α	See absorption coefficient.
σ	Standard deviation. See std

Abstract

Voice Alarm systems (VA) are an essential part of subsurface underground station emergency and evacuation systems. Their main purpose is to assist in the management of emergency situations and evacuation procedures by providing key verbal instructions to the occupants. However these life-critical systems will be ineffective if the messages broadcast are unintelligible. Unfortunately, in most London underground subsurface areas the announcements broadcast by the VA system are not adequately intelligible and often do not reach a minimum specified performance target.

The performance of VA relating to its electro-acoustic characteristics is relatively complex and depends on multiple interrelated factors and operational constraints . Underground stations present complex geometrical and architectural features which severely challenge the achievement of satisfactory performance.

Despite the importance of VA system, there are few works in the literature providing practical and applicable design knowledge in the context of real world underground spaces. Moreover contractual performance requirements are not suitably laid out and this can lead to ineffective designs.

This research aims to provide practical design knowledge and understanding for the improvement of VA speech intelligibility performance in underground spaces. Research results were derived from measurements and designs undertaken for real scenarios. A specific knowledge base is provided on the acoustics of underground spaces, speech intelligibility and VA systems. A critical review of relevant research and performance specifications and standards is undertaken and a new performance design parameter is proposed. An empirical prediction model tool based on a large pool of measured survey data is developed for the prediction of the Speech Transmission Index of VA on platforms. A validating and comparative study is undertaken for two widely used commercial acoustic simulation programs to assess their suitability as design tools for VA systems on platforms, CATT-Acoustic and Odeon. The impact on VA performance of design variables are investigated using a computer simulation of a representative platform. A novel acoustic treatment design concept is proposed. The Yang quasi diffuse sound field theory for platforms is verified and derived knowledge expanded. Practical design recommendations are provided as well as suggestion for further work.

Chapter 1 Introduction

1.1 Foreword

The acoustic performance of voice alarms systems (VA) in underground environments have been traditionally neglected to the point where the messages broadcast are unintelligible and therefore the whole system can become inefficient or even counterproductive. After the 1987 Kings Cross underground station disaster and more recently after the July 2007 bombings on London Underground (LU)[1] , the importance of a highly intelligible VA system for a safe and efficient evacuation procedure has been realised (Fire Precautions Sub-Surface Railway Stations-regulations, 1989), (BS 5839-8, 2008).

The increasing demand for improved acoustic performance of VA systems in underground stations does not only seek to provide audible and intelligible vital instructions during an emergency, it also aims at assisting passenger flows and providing necessary travel/passenger information with a high degree of clarity and acoustic comfort thus conveying an increased sense of well being and quality in the service provided.

However broadcast speech communications in underground stations are severely hindered by the space's irregular and disproportionate geometry, surfaces' high sound reflectivity and significant levels of background noise. Compliance with other station constraints such as health and safety regulations, accessibility, maintenance, cost and aesthetics further restrict the design and testing options available. On the other hand stipulated minimum performance specifications for underground station VA systems (London Underground, 2004), (VA system specifications, 2005) are difficult and often unattainable for the conditions mentioned above. Hence, the acoustic design of VA systems in those spaces is a rather challenging task. There is a need to remind the vital importance of VA systems and

[1] London Underground (LU) is operated by Transport for London (TfL).

review performance specifications in light of practical constraints and relevant research knowledge to facilitate the attainment of balanced performance.

The design of VA systems in underground stations is only a small part of the complete station design which includes provision or maintenance for a large range of many other additional equipment and services frequently of conflicting interests. External factors such as site accessibility, maximum test duration, test conditions and health and safety normally affect the effectiveness and efficiency of the procedures involved in the design of the VA systems. Understanding of the design environment, knowledge of the relevant acoustic theory and research as well as modelling techniques can greatly assist to circumvent the effect of inevitable external factors and practical constraints .

There is a range of sophisticated acoustic simulation software and simple prediction models which are currently used or could be potentially employed for assisting in the design process, however their suitability and reliability for underground stations has not been examined.

Typically, in industrial applications and installations, the engineering companies appointed to design and install the VA systems require reliable processes which are simple, swift and cost effective. Lengthy and expensive acoustic design processes are carried out for each public area of each underground station. However, it is apparent that some common station areas are geometrically and architecturally very similar (e.g. platform areas). The comprehensive study of the acoustic characteristics, influencing variables and best design options of VA systems on underground station platforms can replace the need for systematic acoustic survey and design of similar platforms.

Most of the related research literature concerns with the study and theoretical prediction of sound propagation and reverberation time in long enclosures (i.e. traffic tunnels, long corridors). Few authors have examined sound field characteristics on different underground station platforms. However those works were undertaken with a single sound source, which is not representative of real underground platform multisource VA systems. The limited research on acoustics of underground platforms equipped with multisource VA systems concerns the development of mathematical prediction algorithms based on single source theory. They were also based on important simplification assumptions and provided limited analytical capabilities. The value of these works for industrial real world applications is also limited since the tools and knowledge created have not been extensively tested, further developed or are not available for general use. There is a need to provide research based knowledge, guidance and tools based on real world conditions and data.

1.2 Background to the research

LSBU (Acoustic group) was approached by a company operating in the transportation sector for acoustic expertise and assistance in the design and optimisation of renovated VA systems on LU stations. The initial request gave rise to a collaborative relationship between LSBU (Acoustics group) and the company which in turn generated part of the research work presented.

1.3 Aims

This research project aims to generate practical knowledge and guidance to improve the designs of VA systems of underground stations and help attaining the required performance specifications. This is to be accomplished by providing a review of theoretical and practical understanding of underground platform spaces, creating a dedicated simple prediction model, validating commercial acoustic simulation programs for underground platforms and investigating design key variables and theory in the design of those spaces.

It is intended that the practical knowledge and design advice provided by this research, will equip VA system designers, architects, manufacturers and other stakeholders in taking decisions affecting the acoustic performance of new and refurbished underground VA systems.

1.4 Objectives

The following objectives were established towards achieving the research aims defined above.

- Provide an insight into the practical aspects of electro acoustic design of VA systems under real conditions found in underground stations.
- Present an informative and critical review of the relevant literature concerning the areas of theoretical background, related research, standards and criteria.
- Determination and analysis of existing acoustic and speech intelligibility performance parameters baseline in London underground deep platforms[2]
- Development of an empirical predictive simple model for station platforms
- Validation and suitability evaluation of advanced acoustic simulation programs and simple models as predictive tools for design.
- Evaluation of the effects of VA system design variables on predicted platform speech intelligibility performance.
- Review and verification of the quasi diffuse sound field theory for underground platforms equipped with VA systems
- Provide practical guidance and recommendations to aid in the design and optimisation of new or refurbished VA systems in underground platforms.

1.5 Scope and focus

The scope of this research pertains only to the last section of an underground station VA system that is the electro-acoustic transmission channel (figure 3:2). This channel is defined from the loudspeakers of the system to the receiver. It is assumed that the rest of the VA system operates under ideal condition giving optimal performance.

[2] Deep platform (or deep level platforms) refers in this thesis to station platforms situated well below the street surface level and therefore are acoustically isolated from surface noise activity. They account for almost half of station platforms of the entire LU network.

The practical context of the research is the electro-acoustic design and performance compliance of renovated VA systems for a large number of existing London underground deep platform stations under numerous practical constraints .
The relevant design and performance specifications laid by London Underground and its contractor company as well as relevant standards are taken as target references.

The focus of the research is on predicted speech intelligibility as this is the most important performance parameter and attribute of a VA system.

1.6 Thesis outline

This thesis has been divided in nine chapters. Chapter 1 provides introductory and background information on the research work. Scope, aims and objectives are outlined together with information on intellectual property restrictions imposed on the thesis. A summary of acoustic theoretical background relevant to VA systems in enclosures is outlined in chapter 2. Chapter 3 presents a review of the specific aspects of underground station acoustics, characteristics and limitations in electro-acoustic design. The chapter explores and outlines the necessary knowledge on VA systems. A summary and a critical review of the relevant legislation, standards, performance specifications and criteria are also provided in this chapter including suggested modifications and a proposed novel design performance parameter. Chapter 2 and 3 are considered essential background reading for the prospective VA system designer and station design stakeholders since they can provide a grounding and a realistic understanding of the limitations, challenges, context and opportunities in the acoustic design and implementation of VA systems in underground stations.

A generic review of the related and relevant research work found in the literature is provided in chapter 4. However, additional research literature specific to particular chapter topics are also presented in the relevant chapters.
In chapter 5 a collection of measured acoustic and speech intelligibility predictor data from a large sample of London Underground deep underground platforms is compiled into a database for analysis. This analysis provides information on baseline conditions, variability dependence on several factors and correlation

studies. Ultimately the database is used to derive a novel empirical simple prediction model.

Chapter 6 examines the suitability of two industry standard commercial acoustic simulation programs as tools for the design of VA systems in various underground spaces. As part of the examination, acoustic prediction validations against measurements are carried out for both programs. Also in this chapter validation and inter-comparison of several simple models capable of predicting speech intelligibility of VA systems on station platforms is undertaken.

An evaluation of the most significant design variables affecting predicted speech intelligibility performance from platform VA systems is investigated in chapter 7. This chapter also suggests optimised design approaches which can achieve the speech intelligibility performance target. Practical design guidelines and recommendations are provided as a result of the evaluation and design investigations. A novel design of an acoustic treatment for platform ceilings is presented.

Chapter 8 presents a review and verification of the quasi diffuse sound field theory suggested by Yang (1997) for underground platforms equipped with VA systems. The validation procedure is expanded and further findings and conclusions related to the theory are provided.
Chapter 9 provides the conclusions of the study, a list of design recommendations and proposed further work.

1.7 Non disclosure agreement

Only the content material and information presented in chapter 5 of this thesis is partly based on data and research material generated by the author during the partnership described above. During the partnership the author was the research associate on behalf of LSBU.

LSBU (Acoustic group) and the company agreed that the contents of the thesis which directly concern work or data undertaken during the partnership, shall be written following intellectual property restrictions summarised in appendix A.

Chapter 2 Theoretical background

This chapter presents a summary of fundamental acoustic theory, concepts, and metrics relevant in the understanding of subsequent chapters. Further background knowledge specific to each chapter is provided on a chapter by chapter basis.

2.1 Room acoustics

2.1.1 Sound fields in an enclosure

The research work presented in this thesis falls within the acoustic branch of room acoustics. Room acoustics can be defined as the study of the behaviour of sound in enclosed spaces. In an enclosed space equipped with a sound source, two distinctive sound fields can be defined: The near field and far field. The far field comprises the free or direct field and the reverberant field (figure 2.1).

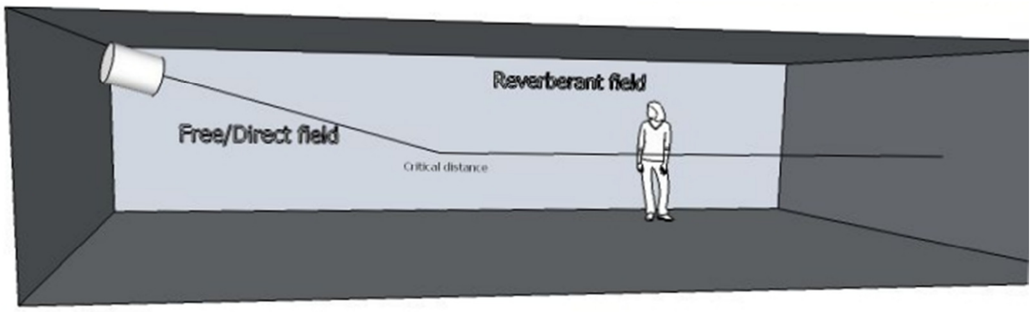

Figure 2:1 Representation of the direct and reverberant sound fields produced by a single loudspeaker in an empty large room.

The extent of these fields depends on the dimensions and shape of the room, the sound source characteristics, the reflective (or absorptive) qualities of the room's surfaces, and the frequency of the sound considered. The near field is the region very close to the sound source (approximately within 2λ) where the sound pressure and particle velocity are not in phase resulting in fluctuation of sound pressure level

(SPL) in a relatively unpredictable manner. For the frequency range concerned in this research (125Hz to 8kHz[3]) the extent of the near field becomes irrelevant for purposes of this study except for 125Hz octave band where the associated near field can contribute to measurement and prediction uncertainties.

In the far field region beyond the near field, the free-field is characterised by the SPL logarithmic decay variation as the receiver moves away from a point source[4]. The SPL drops off at a rate of 6 dB per doubling of distance from a point source, in accordance with the inverse square law of free field spherical propagation, which is simplified in the following equation:

$$SPL_2 = SPL_1 - \left[20 * log\left(\frac{d_2}{d_1}\right)\right]$$

Equation 2:1 Free field SPL propagation determination according to inverse square law

where SPL_1 in dB is the sound pressure level at the location closer to the sound source, SPL_2 in dB is the sound pressure level at the location farther from the sound source, d_1 is the distance from the source at which SPL_1 is measured, and d_2 is the distance from the source at which SPL_2 is measured.

This specific region in which equation 2:1 is valid is called the free-field. Free-field conditions typically exist in large open outdoor spaces or in rooms possessing highly absorptive surfaces where sound reflections cannot occur. The free field can also be referred as the direct field when only direct sound travelling unobstructed from the source to the listener is considered. Direct sound is therefore considered as the pure sound received from the source not having been distorted or affected by the room.

As sound pressure levels drop-off from the radiating source within a room, they eventually fall to a relatively constant level which is strongly determined by the amount of reflected sound within a room which in turn is determined by the sound absorption of the room boundaries. This lower limit is called the reverberant field.

[3] Frequencies refer to the centre frequency of each octave band.
[4] The sound sources utilised in underground stations VA systems (loudspeakers) can be approximated as point sources in the far field due to their relatively small dimensions.

The extent of the reverberant field depends on the size of the room and the size and characteristics of its reflective surfaces (figure 2:2).

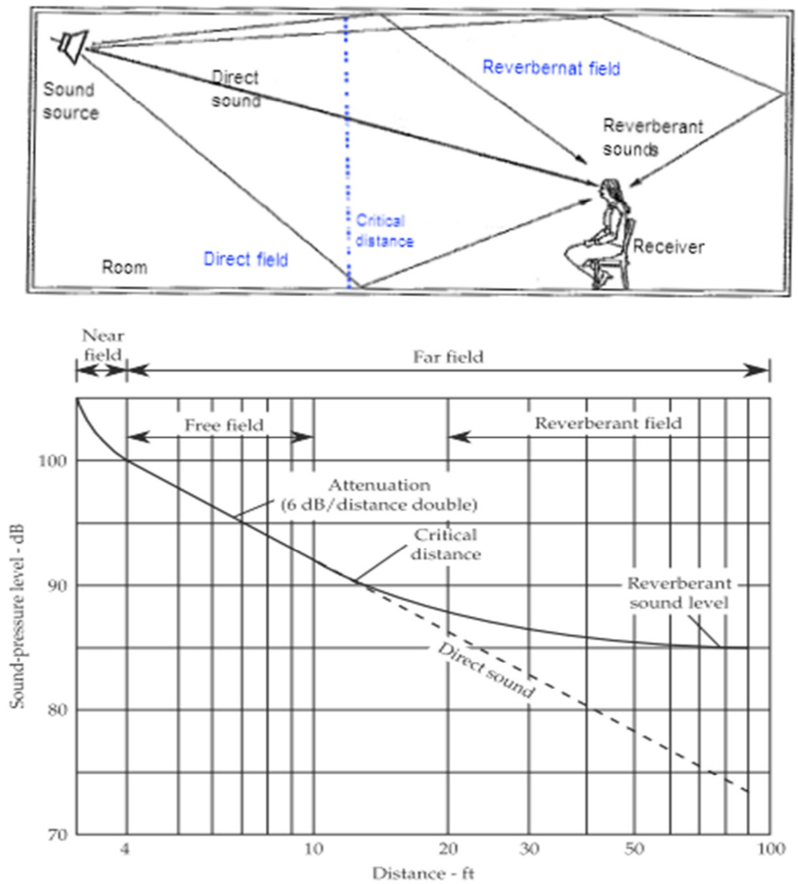

Figure 2:2 Sound fields in a room (top) and corresponding graphical representation of measured SPL's as a function of distance from the source (below)

In practice, the sound fields present in most of rooms are formed by direct and reverberant regions whose extents vary mainly dependent on the absorption provided by the room surfaces. The critical distance (or room radius) is the distance in metres at which the direct sound energy is equal to the reverberant sound energy.

$$Critical\ distance = 0.057\sqrt{\frac{V}{RT_{60}}}$$

Equation 2:2 Room critical distance

where V is the room volume in m^3, RT_{60} is the reverberation time in seconds (see 2.1.2). Further away from this point, the reverberant field dominates and the SPL's become constant, independent of location and only dependant on the amount of absorption in the room.

2.1.2 Sound field acoustic parameters

Sound waves incident on the surfaces of a room are partially absorbed and partially reflected back. The proportion of sound energy absorbed and reflected will depend mainly on the angle of incidence and on the absorption coefficient of the surface materials. Reflections can be either specular of diffuse.

If the size of a flat reflective (or acoustically hard) surface is larger than the sound wavelength then the reflection will be specular following the Snell's optics law which states that the angle of incidence is equal to the angle of reflection. In a diffuse reflection the reflected wave is scattered into smaller components (wave fronts) appearing to bounce off in a range of angles and therefore spatially diffused (figure 2:3).

Sound absorption in objects occurs due to frictional losses between the air particles and the material airways (porous/fibrous materials and resonators) or due to sound energy transferred into mechanical movement or vibration (e.g. diaphragmatic absorbers).

The sound absorbing property of a material is expressed by the sound **absorption coefficient, α** (equation 2:3), defined as the fraction of incident sound energy which is not reflected. This coefficient is a function of the frequency and angle of incidence. Values of α range from 0 (total reflection) to 1 (total absorption).

$$\alpha = \frac{absorbed\ sound\ energy}{incident\ sound\ energy}$$

Equation 2:3 Absorption coefficient

Scattering coefficient, s, is a ratio of sound energy scattered in a non-specular manner to the total reflected sound energy (figure 2:3). Scattering mainly depends on frequency (wavelength), angle of incidence, surface dimensions, surface roughness and edge of the surface exposed to sound waves.

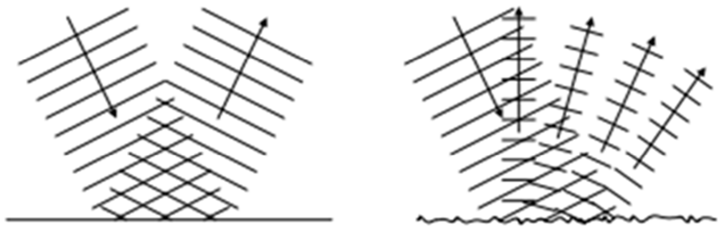

Figure 2:3 Specular and diffuse reflection due to surface roughness (after James et al, 2001)

The main purpose of this coefficient is to characterize surface scattering for use in geometrical acoustic room modelling programs.

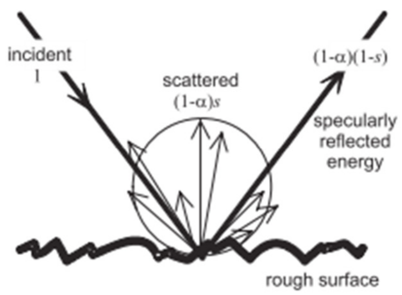

Figure 2:4 Separation of reflected energy into scattered and specular components (after Vorländer and Mommertz (2000))

The energies of reflections (normalised with respect to a reflection from a rigid reference plane) in figure 2:4 can be expressed in terms of:

$$E_{spec}=(1-\alpha)(1-s) \underset{=}{\rightarrow} 1-\alpha_{spec}$$

$$E_{total}=(1-\alpha)$$

Equation 2:4 Specular energy as a function of absorption coefficient where *s* is the scattering coefficient, α is the absorption coefficient, E_{spec} the energy specularly reflected and E_{total} the total reflected energy. The coefficient α_{spec} is the specular absorption coefficient. From these equations the scattering coefficient s can be determined by (Cox et al, 2006):

$$s = \frac{\alpha_{spec} - \alpha}{1 - \alpha} = 1 - \left(\frac{E_{spec}}{E_{total}}\right)$$

Equation 2:5 Scattering coefficient

Nowadays, most advanced simulation software typically include surface scattering in terms of scattering coefficient. Other scattering mechanisms are also considered involving surface size and edge diffraction. ISO 17497-1:2004 provides a method for measuring the random-incidence scattering coefficients in a reverberation chamber of surfaces as caused by surface roughness.

Background noise (or noise floor) is any sound present other than the sound of interest being monitored at given location and given time. The background noise level (BGNL) is measured as SPL in dB. Depending on the contributing noise sources, BGNL may vary with time and in frequency content. To assess the influence of the background noise on acoustic parameters, the spectral content and variation with time must be considered. On underground spaces the main contributing noise sources are people, mechanical ventilation systems, escalators and train noise.

Signal to noise ratio (SNR) expressed in dB is the amount by which the sound pressure level (SPL) of the sound of interest (signal) exceeds the background noise level. A noise in the presence of a sound signal does not contribute appreciably to the overall combined level if its level is ≥10dB[5] below the signal level. If the signal level is ≥15dB above the background noise level at each octave band, the noise contaminating contributions can be considered negligible (Bistafa & Bradley, 2000),

[5] In the case of a voice alarm system a signal to noise ratio of at least 10 dB is desirable to achieve intelligibility of speech messages (BS 5839-8:2008)

(Mapp, 2002a), (Bowden & Wang, 2007) and the situation for speech intelligibility calculations and assessments is considered “noiseless”.

Reverberation is the persistence of sound in an enclosed space resulting from repeated sound wave reflections off the room surfaces after the sound source is ceased. The reverberant sound dies away with time, as the acoustic energy is absorbed by multiple interactions with the surfaces of the room. Reverberation is widely regarded as the most complete and representative descriptor of the acoustic nature of an enclosure. Subjectively reverberation gives perceptions of a space with regards to quality, comfort and well being.

The reverberation time (RT) has been defined as the time for the sound to die away to a level 60 dB below its original steady level (RT60). This drop of 60 decibels, regardless of the initial level of the sound, corresponds to the acoustical energy dropping to one-millionth of its original sound energy which approximately equates to a level when sound signal becomes inaudible (figure 2:5). Very long reverberation times usually exist in large spaces (characterised by large volumes) with acoustically hard surfaces and minimal sound absorption. RT60 is a function of room volume, surface area and sound absorption coefficient (α) of materials and finishes in the room. Due to usual limiting background noise floor level present in real measurement situations, the RT60 is normally calculated from the analysis of the energy decay rate observed on the curve fitting line of shorter measurement ranges (from -5dB to -25dB for RT20 or from -5dB to -35dB for RT30). The values derived from these shorter measurement ranges are doubled or tripled respectively to normalise them to the RT60 definition (figure 2:5 and 2:6). The first 5dB of the decay are excluded to avoid the influence of the direct sound and strong early reflections.

An adequate reverberation time will depend on the type of space and the intended use. Since absorption of material varies according to the sound frequency, reverberation time also is directly dependent on frequency. Hence that RT is normally indicated in octave or 1/3 octave frequency analysis.

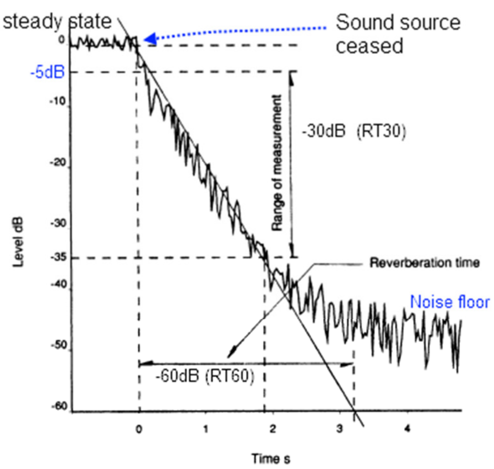

Figure 2:5 Reverberation Time determination showing RT60 and RT30 corresponding measurement ranges (after BS 6259:1997).

The Early Decay Time (EDT) formulated in 1970 and defined in BS ISO 3382-1:2009, is the rate of sound decay in the first 10dB portion of the integrated impulse response decay curve. The slope of the decay curve in that portion is determined from the best fit linear regression line. The EDT is calculated by multiplying the time taken to drop 10dB by a factor of six. This early decay is strongly affected by the relative position of the receiver respect to the source, sound source directivity and surface materials of the objects or boundaries near the source. The EDT is usually of more subjective significance in the design of PA/VA systems than the statistical decay time (RT60). If the EDT is shorter than the RT60, the speech clarity of a sound source is normally increased over what it would be if the early energy followed the same rate of decay as the late energy. The EDT has been shown to relate better to the subjective judgment of reverberation when compared to conventional RT60 (Barron, 2009), (BS EN ISO 3382-1 :2009), (Eggenschwiler & Machner, 2005). Likewise EDT has been reported to be more closely correlated to speech intelligibility (Bradley,1998), (Eggenschwiler & Machner, 2005) and to STI (see below) (Bradley, 1998), (Bradley et al,1999) than the reverberation time RT30.

A **diffuse sound field** is established when it satisfies the following conditions:

- the sound energy flow in the room is equally probable in all directions
- the sound energy density is equal everywhere

The reverberation time in a diffuse sound field is independent of the receiver location (equal everywhere). In a perfect diffuse sound field the time-averaged sound pressure is equal everywhere in the room, the reverberation decay curve is a straight line (when plotted in a logarithmic y-axis scale) and the derived EDT (figure 2:6), RT20 and RT30 values match perfectly (i.e. EDT / RT = 1 or 100%). Deviations from perfect linearity and the shape of the decay usually convey information about the acoustical space and various acoustical processes valuable in diagnosis and assessments.

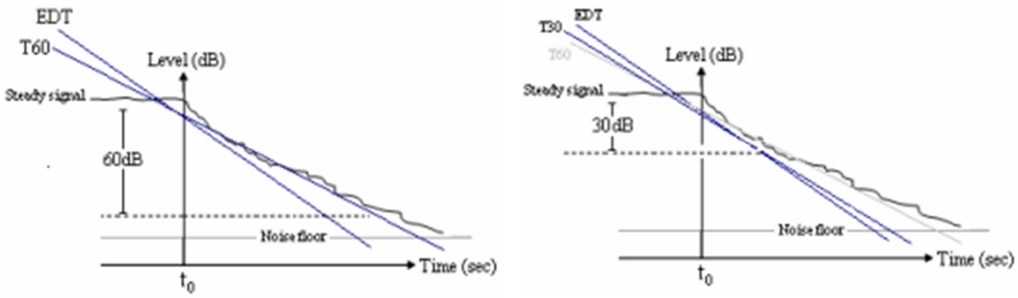

Figure 2:6 Calculation of RT and EDT in two different measurement range conditions (60dB and 30dB).

Diffuse field condition is the premise assumption for the "classic room acoustic theory" on which many practical concepts and simplified predictive formulae are based.

2.1.3 Sound fields prediction calculations

In classic room acoustic theory (or statistical theory) the sound field is assumed to be diffuse. On this basis, W.C. Sabine developed in 1898 an empirical formula to calculate the RT of a room from the knowledge of its volume and total amount of sound absorption. This formula was given by:

$$RT_{60} = \frac{0.161\,V}{A} = \frac{0.161V}{S\overline{\alpha}}$$

Equation 2:6 Sabine RT calculation

where *RT* for the frequency of interest is the reverberation time in seconds, V is the volume of the room in m^3, *S* is the total internal surface area, $\overline{\alpha}$ is the average absorption coefficient of the room's surface materials. *A* is the total random incidence absorption of the room in m^2. When the absorption coefficient of the air *x*, is taken into account the formula is expressed as:

$$RT_{60} = \frac{0.161\,V}{A} = \frac{0.161}{S\overline{\alpha}+xV} \qquad \textit{where} \qquad \overline{\alpha} = \frac{1}{S}\sum_i \alpha_i S_i$$

Equation 2:7 Sabine RT calculation including air absorption

Sabine's calculation theory assumes diffuse field conditions. This fundamental premise limits the validity of his formula to spaces where the absorption is uniformly distributed, the average absorption coefficient is less than 0.2 and the geometry shape is proportional promoting homogeneous mixing of sound within the room. Other statistical formulas have been developed from Sabine's initial works, attempting to account for some of these limitations (e.g. Eyring and Millington formulas).

Within the domain of classic acoustic theory given a room where a sound source of sound power level L_w (dB) and directivity factor Q establishes a steady state condition (sound energy equilibrium in the room); the total SPL in dB received at a distance r from the source can be calculated as the combination of the direct field (L_D) and reverberant field (diffuse field) (L_R) level contributions. The equations 2:8 and 2:9 show the calculation of the direct and reverberant fields contributions. The combination of the two contributions yield the total SPL at any position in the room given by equation 2:10.

$$L_D = L_w + 10log_{10}\left(\frac{Q}{4\pi r^2}\right) \qquad L_R = L_w + 10log_{10}\left(\frac{4}{Rc}\right)$$

Equation 2:8 SPL's (dB) calculation of direct (left) and reverberant fields (right) components

being *Rc* the Room constant given by the formula:

$$Rc = \frac{S\overline{\alpha}}{1-\overline{\alpha}}$$

Equation 2:9 Room constant

where *S* is the surface area of the room (m^2) and $\overline{\alpha}$ the average absorption coefficient in the room.

$$SPL_{total} = L_w + 10log_{10}\left(\frac{Q}{4\pi r^2} + \frac{4}{Rc}\right)$$

Equation 2:10 Calculation in (dB) of SPL_{total} at a receiver position at r from the sound source

The critical distance d_c (or room radius) is the distance in metres at which the direct sound energy in dB is equal to the reverberant sound energy. The d_c is determined by the Volume (m^3) of the space and its RT as expressed by the equation 2:11

$$d_c \approx 0.057\sqrt{\frac{V}{RT_{60}}}$$

Equation 2:11 Room critical distance

Direct to Reverberant ratio (DRR) is the proportion of direct sound to reverberant sound received by a listener. This concept is fundamental in understanding how reverberant sound contributions affect speech intelligibility (see 2.2). However this broad concept does not account for the beneficial effects of early reflections (part of the early reverberation), hence it remains as a concept and not an established metric.

2.1.4 Impulse response and energy based parameters

Appendix B.1 presents a summary of the impulse response (IR) theory and its main applications in acoustic measurements. From the measurement and analysis of the room impulse response (RIR) various acoustic parameters such as the EDT and RT20 and RT30 can be quickly and accurately calculated. Other metrics have been developed from the analysis of the RIR to study different acoustical qualities of rooms.

Definition (Deutlichkeit), D50, is an early-to-late arriving sound energy ratio quantifying the effects of the room on the definition of the received sound. When reflections are received within 50ms of the direct sound, the hearing system combines these contributions with the direct sound so the combination is a homogeneous single sound. This beneficial combination of direct and useful reflections is perceived as if the clear, original sound has been amplified relative to the later, reverberant energy. This parameter is calculated from the squared impulse response and deriving the proportion of early (useful) reflected energy within 50ms of the direct sound to the total energy arrived (figure 2:7). Other delimitation integration times are utilised for other purposes and applications; e.g. 80ms for music.

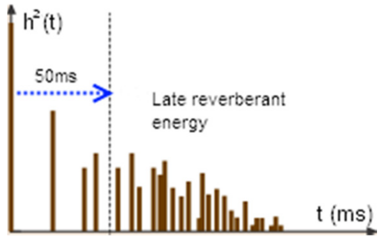

Figure 2:7 D50 energy ratio delimitation on an echogram of an impulse response

D50 is given by the equation 2:12 below and conveniently expressed as a percentage. High percentage indicates high definition of the arrived sound.

$$D_{50} = \frac{\int_0^{50ms} h^2(t)dt}{\int_0^{\infty} h^2(t)dt} * 100\%$$

Equation 2:12 D50 theoretical calculation

D50 has been found in various works (Kuttruff, 2000), (Mapp,2002a) [6] to be closely related to speech intelligibility.

Clarity Index, C50, is a very similar parameter to D50 which derives from the squared impulse response the proportion of early reflected energy within 50ms of the direct sound to the proportion of energy arrived after that time window. It is given by equation 2:13 below expressed in dB.

$$C_{50} = 10log_{10}\left[\frac{\int_{0}^{50_{ms}} h^2(t)dt}{\int_{50_{ms}}^{\infty} h^2(t)dt}\right]$$

Equation 2:13 C50 theoretical calculation

Similar to D50, C50 is used to relate results to conditions for speech intelligibility (see below). Both parameters are directly related by the following relationship:

$$C_{50} = 10log_{10}\left(\frac{D_{50}}{1 - D_{50}}\right)$$

Equation 2:14 Conversion relationship between C50 and D50

D50 and C50 results are calculated and presented in octave band frequency analysis. Despite that both parameters are well correlated with speech intelligibility measures in high SNR (noiseless) conditions, they do not account for signal-to-noise ratio and therefore should not be used as sole predictors of potential speech intelligibility in situations where background noise is significantly present.

[6] (Bradley,1998) and (Mapp,2002a) reported high STI correlations with C50 parameter in noiseless conditions; however D50 and C50 are equivalent and mathematically interchangeable using equation 2:14 (Bradley et al,1999), (Dalenback,2007),

2.2 Speech intelligibility

When designing a VA system, the most important factor to consider is the intelligibility of the required speech signals (BS 6259:1997). Speech is a continuous varying sound waveform defined in three domains namely sound pressure, time, and frequency. This waveform signal is of a wide frequency range starting from below 125Hz and extending to above 8kHz. Its fundamental frequencies typically ranging between 100Hz and 400Hz, and depends primarily on gender. These fundamental frequencies are the foundation for the series of changing harmonics created called formants. The energy in the speech is carried mainly by vowels (corresponding to low frequencies content) while most of the discerning and comprehension cues of the speech message is attributed to lower level consonant sounds (high frequencies). Figure 2:8 shows these patterns of frequency bands approximate contributions to overall speech energy (indicated in SPL) and to intelligibility (indicated in %).

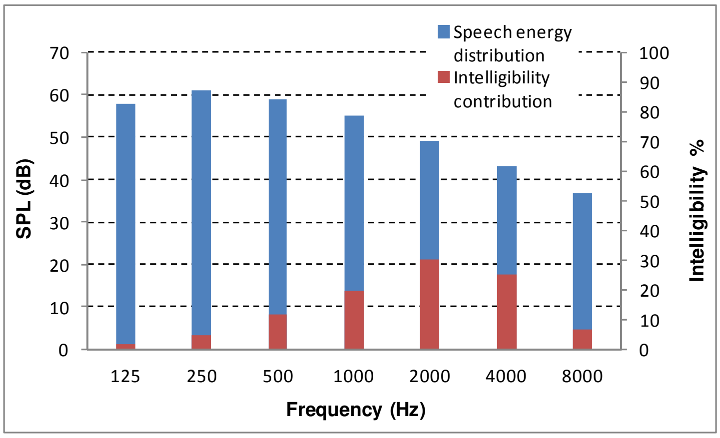

Figure 2:8 Time averaged speech spectrum relative level distribution and relative octave band percentage contributions to intelligibility (after Ballou, 2005).

It is important to highlight that 2 kHz and 4 kHz frequency bands provide most of contribution to total speech intelligibility (30% and 25% respectively) while the 1 kHz octave contributes a further 20%. These three bands therefore provide approximately three quarters of the available spectral intelligibility content.

Common durations for vowels and consonants are approximately 100ms and 60ms respectively. These considerations follow that vowels tend naturally to mask over the consonants.

Speech intelligibility is defined as the measure of the proportion of the content of a speech message that can be correctly understood (BS 5839-8:2008), it is therefore a subjective derived metric rather than a physical quantity. Satisfactory speech intelligibility requires adequate audibility and adequate clarity. Clarity is the property of a sound which allows its information-bearing components to be distinguished by the listener being related to the freedom of the sound from any type of distortion (BS 5839-8:2008). The clarity is negatively affected by non-linear distortion, frequency response imbalance in the transmission channel (source-room-receiver) and by echoes and reverberation (time domain distortions). In other words clarity describes the ability to discern correctly the structure of a speech sound into vowels and consonants.

Audibility is the property of sound which allows it to be heard among other sounds therefore it relates to the physical ability to hear the sound and is well quantified by the Signal to Noise Ratio (SNR). This property depends on the spectral distribution and variation with time of the signal and noise. Noise has the effect of masking or obscuring the voice signal. However humans are able to tolerate a great deal of noise before intelligibility diminishes appreciably.

In many situations the speech signal is degraded by the signal path or the transmission channel between talker and listener, resulting in a reduction of the intelligibility of the speech at the listener's location. When speech is transmitted through a reverberant space temporal distortion (echoes and reverberation) affect the speech waveform by smearing it in time. The lingering reverberant energy of one syllable (late reflections) overlap and mask the start of the following one rendering a reduction of intelligibility (figure 2:9). In a similar fashion, if the speech signal level (i.e. loudness) is not sufficiently higher than the background noise (SNR), parts of syllables become buried (or blurred) by the noise and the intelligibility of the message diminishes (figure 2:10).

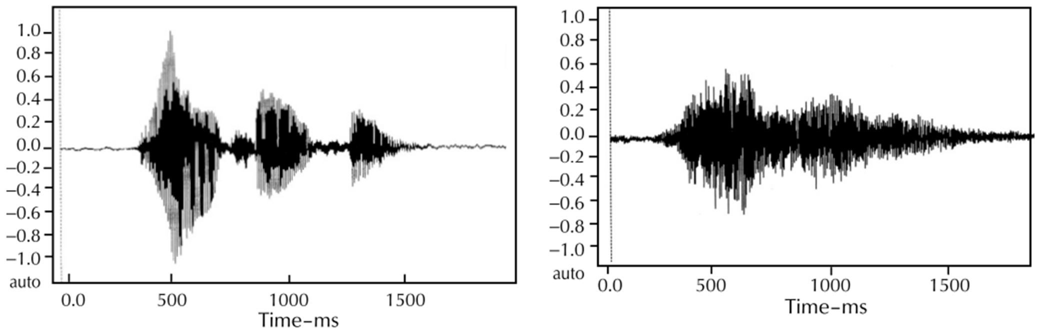

Figure 2:9 (Right) Speech signal received as direct sound in anechoic and noiseless condition (pure); (Left) and received in noiseless but reverberant space (smeared) (after Ballou, 2005).

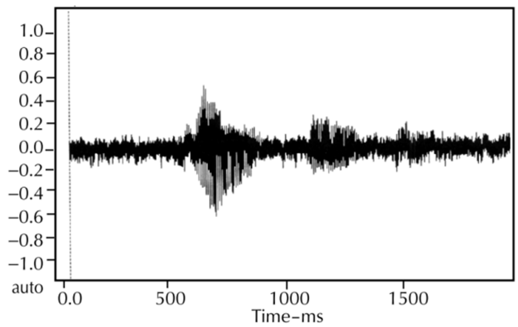

Figure 2:10 Speech signal received in a space with low reverberation but high background noise (after Ballou, 2005).

In essence the two main fundamental factors affecting speech intelligibility in a natural (human to human) or reproduced (through a sound system) process are the room reverberation and speech signal to noise ratio experienced at the listener position. Specific elements contributing to these two primary factors include:

- Volume, size and shape of the space.
- Sound absorption and scattering of surfaces and fittings
- Source-receiver distance
- Loudness
- Power and directivity of the source and receiver
- Transmission channel frequency response / tonality balance
- Background noise character/spectrum
- System linearity (non-linear distortions)
- Talker annunciation / rate of delivery
- Listener acuity

2.3 Measures of speech intelligibility

Speech intelligibility combines cognitive and psycho-acoustics processes. The methods developed for speech intelligibility assessment and measurement can be divided into two main groups: Subjective and objective. Subjective tests (e.g. PB word scores, Modified Rhyme Test (MRT)) involve the use of trained speakers, speech test materials (word/sentence scores) and a representative panel of listeners. Objective (or quantitative methods) tests utilise measurements of physical acoustic parameters of the transmission channel (source-channel-receiver) that correlate closely with robust subjective measurements of speech intelligibility. Although subjective methods are considered generally the most accurate, they can exhibit low repeatability and reproducibility and are considered to be complex, expensive and time-consuming to set up for large scale practical applications. Objective methods are widely used in industry due to their practical implementable qualities (time and cost efficiency, simplicity, consistency) which among others allow the prediction of parameters by various calculation techniques.

Various objective speech intelligibility metrics were developed in the last century which attempt with different degrees of accuracy, to assess the potential speech intelligibility of a communication channel.

The early **Articulation Index (AI)** (ANSI S3.5 1969) proposed a measure of the intelligibility of hearing speech in a given noise environment. It does not account for room acoustics (reverberation) or non linear distortions. The percentage **Articulation Loss (%ALcons)** method expresses loss of consonant definition and is based on measurements only on the 2kHz octave band. It does not account for vowel components, signal-to-noise ratio, background noise spectrum, distortion, late reflections or echoes or system frequency response. %ALcons was never standardized. **The Speech Intelligibility Index (SII)** method for prediction of intelligibility (ANSI S3.5 1997) is based on the early Articulation Index (AI). SII considers reverberation, noise and distortion, all of which are accounted for in the modulation transfer function. The method is not robust for late-arriving reflections, echoes, compression and does not take non-linear phase into account, however it can provide measurements using high resolution frequency bands. It has been widely used in the American industry although in recent years has been replaced by the more widely accepted Speech Transmission Index (STI).

Speech interference level (SIL) is a simple calculation method to predict or assess intelligibility in direct communications (person to person) based on the speech to noise level difference. Due to its simplicity, the use of SIL is only recommended in situations where other assessment and prediction methods of speech intelligibility cannot be applied (BS EN ISO 9921:2003).

The Speech Transmission Index (STI) method is an advanced objective metric which predicts the intelligibility of speech transmitted from talker to listener by indicating the degree to which the transmission channel (e.g. PA/VA systems) degrades speech intelligibility. The STI method and its subsets (see appendix B.2) can be used to determine the potential intelligibility of a speech transmission channel at various locations and for various conditions. In particular, the effect of changes in the acoustic properties of spaces can be assessed. The method is based on the modulation transfer function concept (MTF) which analyses the intensity modulations loss of the original speech signals caused by the transmission channel. The result of the final analysis is an index that ranges from zero (unintelligible) to 1 (perfectly intelligible). The STI method accounts correctly for band-pass limiting, noise, reverberation, echoes, and some degree of non-linear distortion.

The first work leading to the development of the STI were published by Houtgast & Steeneken in 1971 and 1973. Since then and until recently (BS EN 60268-16:2011) the STI method has been expanded, refined, and extensively validated, such that for the past 15 years it has become the de-facto world industry and research standard for measuring and predicting intelligibility of transmission channels including VA systems. STI has been shown to correlate closely with subject based measures of speech intelligibility for adult listeners with normal hearing (Houtgast & Steeneken,1973), (Houtgast & Steeneken, 1985). (Anderson & Kalb,1987), (Steeneken & Houtgast,2002).
This method was first internationally standardised in 1988 (IEC 60268-16) and since then three major revisions have been published (BS EN 60268-16:1998:2003 :2011).

The STI method has also been adopted in several national and international codes of practice (BS EN ISO 9921:2003),(BS 6259 :1997),(BS 5839-8 2008).

International standards (BS EN 60268-16:2003 and BS EN ISO 9921:2003) include a standardised five point rating scale to determine the quality of the speech communication (Table 2:1).

STI	Speech Intelligibility
0.00 – 0.30	Bad
0.30 – 0.45	Poor
0.45 – 0.60	Fair
0.60 – 0.75	Good
0.75 – 1.00	Excellent

Table 2:1 STI scale and equivalent subjective perception rating of speech Intelligibility

There are two methods to measure STI: the direct methods using modulated test signals (full STI, STIPA, RASTI, see appendix B.2) and the indirect methods based on the system's impulse response (RIR) using the Schroeder method (1981) assuming linear and time invariant (LTI) conditions (see appendix B.1). The indirect method is widely used in experimental measurements and in computer modelling where the RIR of the room is simulated for later analysis (see appendix B.1) (BS EN 60268-16:2003 :2011).

If diffuse field conditions are assumed, the STI can be theoretically calculated from the knowledge of the reverberation (EDT), signal and background noise spectra (see appendix B.2), (ISO 9921:2003), (BS EN 60268-16:2003 :2011).

The last revision of the STI standard was released in late 2011 (BS EN 60268-16: 2011). However measurements and investigations of this research were taken when previous revision was current (BS EN ISO 60268-16:2003). For consistency the same revision version was followed throughout the project. For the sake of completeness and referencing the main relevant differences and additions incorporated in the last revision are presented in the appendix B.2.

2.4 Acoustic simulation

Acoustic simulation provides a close representation and characterisation of the sound field of a space prior its existence or renovation. In large and reflective rooms, such as underground spaces, the satisfactory performance of a VA system depends on its interaction with the challenging acoustics of the space where it is installed. Acoustic simulation can be a valuable tool to account for these complex interactions and investigate design iterations of new or renovated spaces. Classic room acoustic theory (see 2.1.3) offer formulae to predict some basic parameters under several limiting assumptions and approximations. However these predictions are normally not suitable for spaces with complex geometries, interior architectures and multi source electro-acoustic installations.

2.4.1 Acoustic simulation frequency range delimitation

Sound field behaviour in rooms can be broadly characterised in two ways depending on the room dimensions and frequency, namely: wave and geometrical behaviour. Schroeder (1962) defined a relationship (equation 2:15) to conveniently characterize a room as (acoustically) small or large in terms of its frequency response. In acoustically small rooms and at low frequencies, natural room modes dominate the behaviour of the room. For this type of behaviour wave methods in either the time or frequency domain can be used to predict the room response. In acoustically larger rooms higher densities of overlapping modes are present allowing room reflections and statistical behaviour to be employed. These reflections and localised reverberation can be considered by means of geometric modelling methods.

The Schroeder (1962) equation (equation 2:15) provides a guide to delimit modal behaviour and overlapping modal behaviour (Van Buuren, 2009) in terms of a frequency ($f_{Schroeder}$) (Hz) which in turn delimits the applicability range of the modelling approach to be used.

$$f_{Schroeder} \approx 2000\sqrt{\frac{RT}{V}}$$

Equation 2:15 Schroeder frequency delimiting room acoustic behaviour in the frequency domain

Where RT is the reverberation time in seconds and V the room volume in m^3.

2.4.2 Acoustic simulation techniques

2.4.2.1 Wave acoustics

In order to acoustically model small rooms, solutions to the wave equation can be implemented analytically or by using elements methods such as the Finite Element Method (FEM) or the Boundary Element Method (BEM). These methods are often not practical in room acoustics because the number of modes in a room increases rapidly to an unwieldy value as frequency increases. As a result these methods are restricted to small simple rooms at low frequencies.

2.4.2.2 Physical scale modelling

This technique studies the sound fields generated in a scaled-down recreation of a space. This technique takes account of wave effects and can yield accurate predictions in the full frequency range. Hence this approach is still used when high degree of visual representation and prediction accuracy is required during the design process (e.g. auditoria). However the technique is time consuming and expensive as compared to computer simulations. With the rapid development of computer technology over the past thirty years, physics and acoustics researchers have been able to efficiently implement complex mathematical algorithms to calculate acoustic parameters in ample variety of spaces and scenarios with comparable accuracy to physical scale modelling. Computer simulation offers the great advantage of providing higher degree of flexibility in the design/assessment process.

2.4.2.3 Geometrical acoustics

The simulation of acoustically large ($f >> f_{Schroeder}$) spaces is reliably approached using geometric acoustics in which emitting sound from a source is simulated by rays or sound particles carrying proportional energy. These rays or particles propagate in straight lines being reflected off the room boundaries where they lose energy at each hit according to the acoustic properties of the boundaries (absorption and scattering coefficients). The energy (intensity) of the ray is additionally reduced by the inverse square law of spherical propagation, since each ray represents a portion of the spherical wave front. However, geometrical models cannot adequately model wave effects, such as diffraction and phase interference, which can be significant at low frequencies. The room geometry is represented by a three-dimensional (3-D) computer-aided design (CAD) model.

Image source method (ISM)

The image source method is a geometrical and deterministic technique based on the concept that a sound ray that is reflected from a planar surface can be thought of as originating from a virtual source (S') which is the mirror image of the original sound source (S) as formed by the planar surface (figure 2:11). The mirroring process can be extended to enclosures with many planar surfaces, yielding higher order reflections due to mirror image sources until a predefined order is reached. The incident ray intensity is reduced on reflection to account for the energy lost due to surface absorption. Energy loss due to distance propagation is also accounted for.

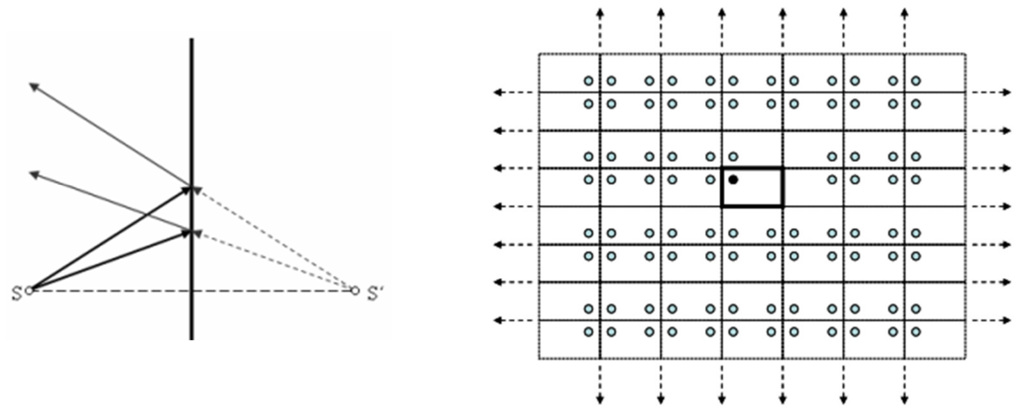

Figure 2:11 Image-source principle and mirroring room sources and boundaries process. Right rectangular room (dark box), source (black dot) and image sources (light dots).

In theory, the final infinite pattern of images represents all reflections from the surfaces in the original room. The sound field in the room can be found by summing the contributions of all free-space images source. If the original sound source generates a Dirac impulse, this process yields the impulse response of the room (see appendix B.1). This technique exhibits the following limitations: surfaces must be perfectly reflective and planar, exponential growth with length of impulse response, difficulty to implement for complex room shapes. In spite of these limitations, the method is very accurate and the impulse response can be calculated with a high time resolution (Vorlander, 1997).

Ray tracing

This stochastic technique involves the representation of the room using a 3D geometrical model. The sound source emits a large number of rays of certain intensity, derived from the sound power, in many directions according to the source directivity pattern. The rays propagate around the room losing energy in transit due to air absorption coefficient, distance propagation and at each reflection according to the absorption coefficient of the surface being hit. Rays are reflected at each surface in a direction according to the scattering coefficient of that surface. Receivers are small volumes which record the number, direction, time and energy history of each ray passage since leaving the source. The ray continue to propagate until its energy is exhausted by surface and air absorption or until a predefined ray truncation time limit is reached (figure 2:12). On completion of the tracing for all rays, the impulse response at the receiver positions can be derived from the tracing history information recorded.

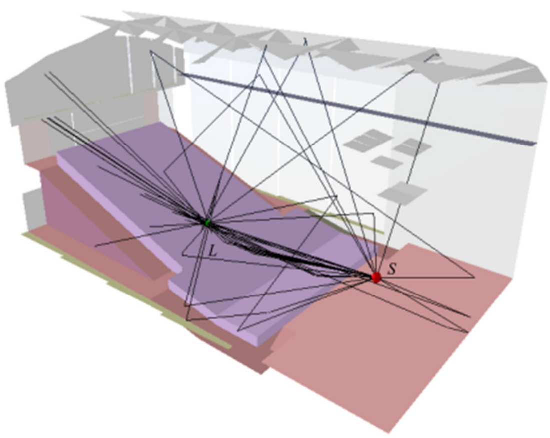

Figure 2:12 Ray tracing computer simulation showing first and second reflection paths in a auditorium (after Savioja,1999).

This technique is suitable to simulate rooms of complex geometries and shapes. An important advantage of this technique is that it can efficiently handle surface scattering. The chosen number of emitted source rays, has a direct impact on accuracy and calculation time. The more rays the longer the computational time. More rays are necessary for large spaces and rooms with numerous surfaces when high accuracy is required. However, ray tracing models exhibit more limited temporal resolution as compared to the image source method

Radiosity technique

The principle of this technique is that the reflected sound from a surface is represented by a large number of point source covering the surface and radiating according to some directivity pattern, usually a random distribution of directions (Lewers, 1993). This method has also been used as an efficient way to model the scattered part of the early reflections.

Hybrid methods

Hybrid methods are geometrically approaches to overcome the limitations of both image-source and ray tracing techniques, while maintaining and maximising their respective advantages. These methods may contain variations on the general methods (e.g. cone, beam or pyramid tracing as opposed to ray-tracing) and are usually optimised to speed up computation time without significant loss of accuracy and vice-versa (figure 2:13). Hybrid algorithms can include a visibility test which uses ray-tracing to determine valid image sources (Long, 2006).

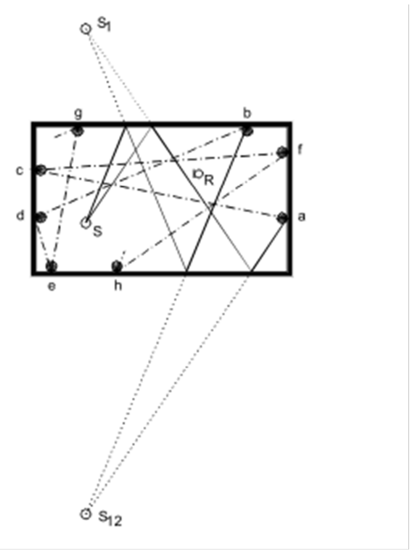

Figure 2:13 Hybrid model principle. The rays create image sources for early reflections and secondary sources on the walls for late reflection (after Rindel, 2000).

Hybrid techniques frequently divide the analysis into early and late reflection with some transition order defined (Christensen,2008a),(Dalenbäck,2007). Often the image source method is used for the generation of the early part of the impulse response (along with a visibility test), while some type of ray tracing for the later part of the impulse response. Some hybrid techniques are complemented by a stochastic simulation run time (Dalenbäck, 2007). This hybrid method of prediction have been adopted and extensively advanced by most established commercial computer simulation developers (CATT-Acoustics, Odeon, EASE).

In a comparative study of commercially available computer simulation programs (aka Round Robin) Bork (2000) found that hybrid method programs produced the most accurate results. The hybrid technique is able to accurately simulate the RIR at receiver positions from which room acoustical parameters including RT,EDT,D50,MTI,STI can be calculated.

2.4.3 CATT Acoustics

CATT-Acoustics (Dalenbäck, 2007) (CATT) is a complete computer room acoustic simulation program first developed in 1989 by Dr. Bengt-Inge Dalenbäck. CATT together with Odeon (see below) are two of the most advanced, reliable and highly regarded commercial programs available. A substantial part of this research is based on the utilization on these two programs.

CATT is a hybrid program which provides the prediction, post-processing, convolution and multiple source addition capabilities. It requires a 3D geometrical description of the space and the input information related to source, receivers and surfaces. Geometrical models can be constructed using external CAD drafting programs (Google SketchUp, 2010) and imported into CATT by means of a intermediate program (SU^2CATT) (Rahe-kraft, 2006), thus avoiding the complex and time consuming process of using plane-corner Cartesian coordinates.

Following the creation of a working model (including the geometrical model, source receiver and surfaces complete definitions) a number of calculations can be performed to produce a wide variety of parameters and charts, graphs and visual displays (figure 2:15). Similar to absorption coefficient selection and assignment, the program uses scattering coefficients assigned by the user in octave bands normally estimated based on general guidance. The program features the option of automatic edge diffusion as well as an automated calculation of scattering coefficient for smooth objects dependent on their size and sound frequency.

CATT can utilise three types of calculations which suit different modelling applications. For certain specific applications where only the early part of the echogram is of interest, the direct sound and early part of the echogram is modelled using the Image-Source (ISM) technique. This technique is also used to calculate the 1st order specular and diffuse reflections as well as the 2nd order specular reflections. The use of ISM in the speech critical early part of the echogram allows a high degree of qualitative detail. One of the distinctive features of CATT it its way of handling diffuse reflections. For 1st order reflections individual sources are applied to the diffusive elements, with the density of these sources determined by $(1-\alpha)\delta$. The emitted power is determined by Lambert's law (Kuttruff, 2000), being proportional to $(1-\alpha)\delta$, where δ is the frequency-specific scattering coefficient and α the absorption coefficient. With diffuse reflections above the 1st order, a random number method for scattering is employed, where the scattered ray direction is determined by Lambert's law.

The full detailed calculation prediction uses Randomized Tail-corrected Cone-tracing (RTC) for the echogram calculation in octave bands. This method assumes that the reflection density growth is quadratic for closed rooms without coupling effects. The ISM is used for the early part of the echogram, as described above. Then the RTC algorithm is used for higher order of reflexions where cone directions are randomised so that the diffuse reflections can be taken into account. Cone tracing is similar to ray tracing except it uses cones instead of rays, As cones do not cover the surface exactly, they must be overlapped (figure 2:14). An algorithm must also be implemented to weigh the energy so that the multiple contributions produce (on average) the correct sound level. Only the centre ray of the cone is traced in the calculation (Ballou,2005).

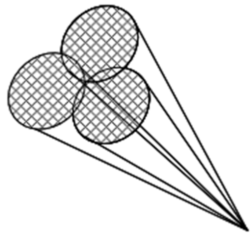

Figure 2:14 Cone tracing principle showing cone overlapping

One advantage of the RTC is that requires very few assumptions about the statistical properties of the room under consideration. If the room is closed, reflection density growth is assumed to be quadratic over time. If the room is partially open, the algorithm makes no assumption about the reflection growth. For coupled rooms and rooms shapes where more late reflection definition is required (e.g. L shape rooms) the developer recommends to disable the RTC algorithm so the late part of the ray trace is traced conventionally in detail, with the consequent increase in computational time.

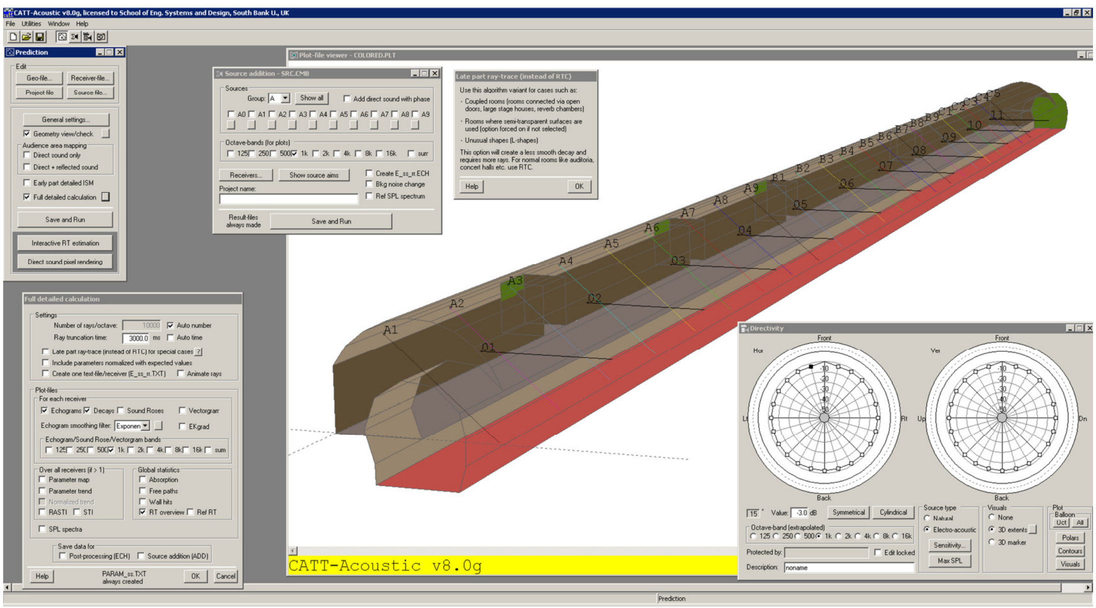

Figure 2:15 CATT v.8.0g main interface showing an underground platform model with a distributed VA system and several calculation settings and control panels

2.4.4 ODEON

Odeon (Christensen, 2008a) is an advanced room acoustic computer simulation program. The first commercial version of the program was released in 1991 and since then has been continuously developed and improved. In a room acoustic Round Robin in 1995 (Vorländer,1995) Odeon was together with CATT one of only three programs out of 16 to be judged *'Unquestionably reliable in the prediction of room acoustical parameters'*.

Odeon basic operations, calculation capabilities and functionalities are very similar to CATT, although Odeon generally exhibits shorter computation times and is considered to be more intuitive and to have a superior user interface. Odeon is based on a hybrid approach (figure 2:16) which combines the image source with a special ray-tracing/radiosity method.

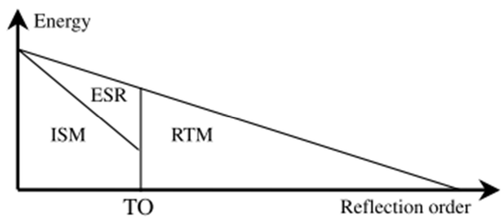

Figure 2:16 Summary of the hybrid calculation method used in Odeon (after Christensen, 2008a).

Early reflections below a user selected transition order are calculated using a combination of the image source (ISM) method and early scattering rays (ESR). Reflections above the transition order (TO) are calculated using a special ray-tracing (RTM) process generating diffuse secondary sources at the collision points between walls and the rays traced. The reflections are calculated in terms of time of arrival, strength in eight octave-bands and angle of incidence. The information on size of the reflecting surfaces and absorption coefficients are also included as a part of the calculation and used to determine whether a phase shift should be applied to the reflection.

Another feature of Odeon in terms of calculation algorithms relate to the user selectable scattering calculation method (figure 2:18). In Odeon scattering is caused by surface roughness and diffraction from surface edges. The primary scattering method is the conventional full scatter, Lambert method. In addition, two more techniques developed by Odeon are available aimed to refine the scattering processes and improve accuracy of predictions of late reflections, these are explained as follows:

Oblique Lambert methods allows frequency dependent scattering in late reflections for point response calculations. Depending on the scattering coefficient the area radiation provided by late secondary sources is tilted towards the specular direction. If the calculated scattering coefficient is zero the secondary source will point in the specular direction and if it is one then it will point in the normal direction of the surfaces as a traditional Lambert source. Oblique Lambert produces a shadow zone where no sound is reflected (figure 2.17). Because this technique will make part of the *Oblique Lambert* source point out of the room, a compensation factor to is applied in order to avoid energy loss.

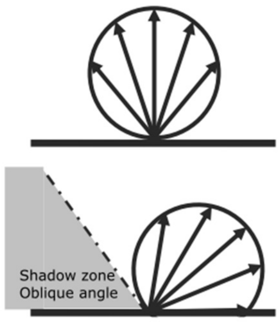

Figure 2:17 Traditional Lambert energy scattering directivity (top) and oblique Lambert (bottom).

The Reflection based scattering method automatically takes into account scattering occurring due to geometrical properties such as surface size, path lengths and angle of incidence. In this method due to different angle of incidence, one surface will provide a higher degree of scattering at some locations than in other. In other words calculates a scattering coefficient that is unique to each reflection.

Even though only one scattering coefficient value is entered for each surface, this coefficient is expanded into values for each octave band, using interpolation or extrapolation to cover the whole frequency range, using frequency interpolation functions.

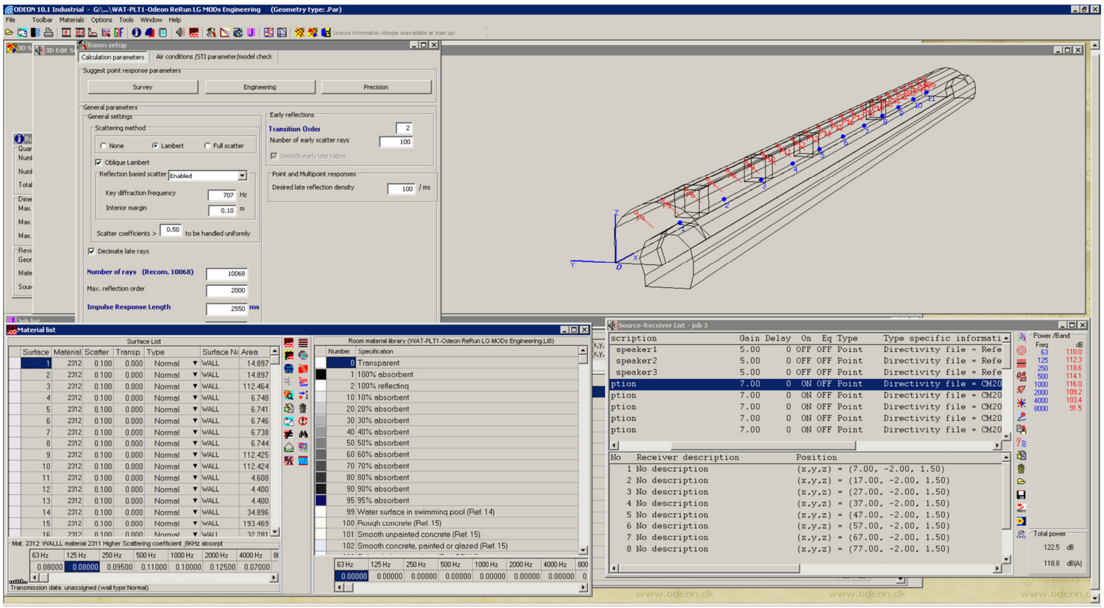

Figure 2:18 Odeon v.10.1 Main interface showing an underground platform model with a distributed VA system and several program settings and control panels

Chapter 3 Voice Alarm systems in underground stations

3.1 Introduction

In order to achieve a satisfactory electro-acoustical performance from underground stations Voice Alarm systems (VA) it is necessary to understand the particular environment, practical limiting factors and acoustics interactions of these systems in those spaces. This chapter presents a summary of specific background knowledge on VA, underground station acoustics and characteristics relevant to their electro-acoustic design and performance. A summary of the relevant standards, guidance, performance specifications and criteria is also provided. A critical discussion on the performance specifications documents highlights potential weaknesses and a novel design parameter is proposed for design performance assessment. The objective of the chapter is to equip designers, management and other stakeholders with specific knowledge and practical understanding of the topics above to attain the performance targets required.

3.2 Voice Alarm systems

3.2.1 Definitions

A Public Address systems (PA) is defined by BS 9921:2003 as *"an electro-acoustic system that is used to address a group of people in one or more environments".*
Within London Underground, PA systems installed in subsurface stations[7] are classed as Voice Alarm systems since they are integrated into the station's fire alarm and emergency evacuation system (VA system specifications, 2005). VA systems in

[7] Subsurface stations refer here to stations situated below the street level. These station spaces can be either at deep level where there is no direct exposure to outdoors or at low level below street (aka cut and cover) where direct or indirect exposure to the outdoors can occur.

that environment form the communication part of the statutory requirements under Fire Precautions Sub-Surface Railway Stations regulations 1989. Overground[8] stations do not require fire alarm evacuation systems thus electro-acoustic sound systems installed on over-ground stations are referred as PA systems. Subsurface stations are also referred as "section 12 stations" in reference to the classification made in section 12 of the Fire Precautions Sub-Surface Railway Stations Regulations 1989.

A Voice Alarm system is defined "*as a sound distribution system that broadcast speech messages and / or warning signals in an emergency*" (BS 5839-8:2008). British standard 6259:1997 defines a sound distributed system as "*A sound system designed to distribute audio signals from a source in one or more locations to a number of other locations through a multiplicity of loudspeakers".*

London deep platforms are subsurface platforms which run typically at 20m below surface and therefore have not direct exposure to external acoustic influences. They are characterised by having a single track and curved concave ceiling of a circular cross section (figures 3:1 and 3:3) and are the focus of this research.

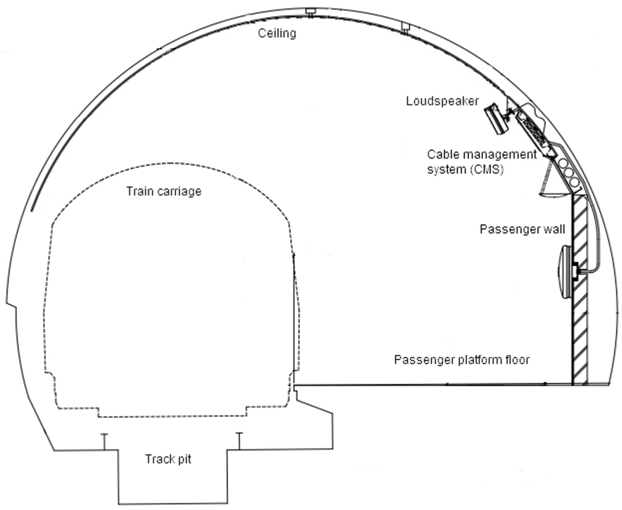

Figure 3:1 London Underground station deep platform cross section

[8] Overground stations are at or close to street ground level and therefore are acoustically exposed to open air and outside background noise.

3.2.2 The importance of VA systems in underground stations

The main purposes of VA systems in underground stations are to broadcast intelligible speech announcements as an aid to the safe evacuation strategy under emergency conditions (e.g. fire, terrorist attack), control passenger flows and provide safety warning on platforms (London Underground, 2004), (Aldred & Gorasia, 2007). A function considered secondary since it is not life critical, is to provide regular travel/customer information (Aldred & Gorasia, 2007).

Speech based communications systems have been found to be the second most effective warning systems (Tom et al, 2001) after visual based systems. The primary reason for using speech announcements instead of coded audio warnings (tones) given by sounders, is to reduce the time taken for those at risk to recognize that an emergency exists, and to give clear instructions on what to do next (EN 54-24:2008).
However, announcements may be ignored if the message is not intelligible (Tom et al, 2001). Hence a VA system loses its intended purpose if it is unintelligible and it can become even counterproductive in case of an emergency. Conversely, an intelligible system broadcasting with high speech quality not only will facilitate a safe and efficient evacuation, it will also reassure the feeling of safety and well being of occupants as well as enhance the quality of service provided.

3.2.3 VA system structure

A voice alarm system is fundamentally a one way speech communication channel consisting of several consecutive and distinctive parts. A typical VA system structure can be thought as a chain system in which the final output is determined by the weakest link in the chain. The basic structure consists of the input section, signal processing, amplification and the electro-acoustic section (figure 3.2).

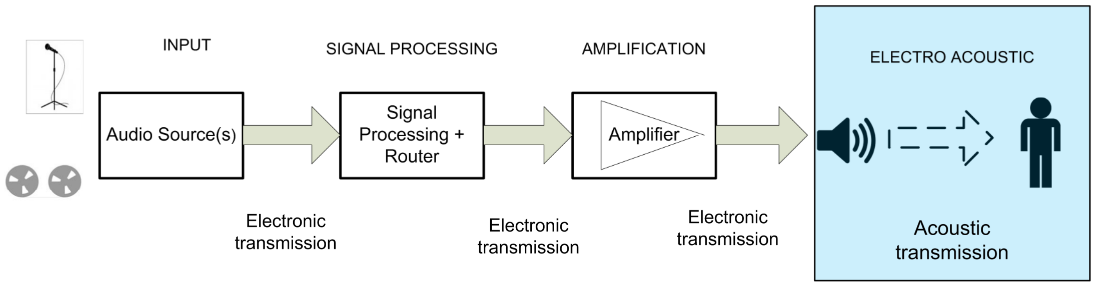

Figure 3:2 VA system basic structure and signal transmission path

The input section receives the speech signal either live from an announcer through a microphone or reproduces electronically stored pre-recorded announcements which are directly fed into the next section. The signal processing section prioritises, routes and processes the incoming input signal(s) before it is sent to the next stage where its level is amplified. The system amplifiers increase the voltage of the signal to enough level to excite the distributed loudspeaker system (or array) which will reproduce acoustically the signal to audible levels into the underground station area(s).

The last part of the system named as electro-acoustic transmission section, comprises three elements: the loudspeaker array (sources), the room space (acoustic transmission channel) and the listener (receiver) (figure 3:2). It is in this last section of the chain where the performance of the whole system is delivered (perceived at the listener's ears or measured at a microphone). The investigations of this research are confined within the electro-acoustic transmission section. The acoustic transmission channel is the space defined between the loudspeaker array and the listener (figure 3:2 and 3:3).

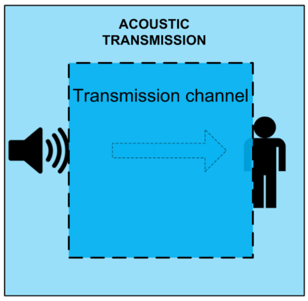

Figure 3:3 Electro-acoustic transmission section of an underground platform (left) and representation of the acoustic transmission channel (right).

3.2.4 Speech intelligibility degrading factors along the chain

Speech intelligibility is the most important performance requirement in attaining the purposes of underground stations VA systems (London Underground, 2004), (VA system specifications, 2005), (BS 6259:1997) and is the central focus of this research work.

The performance of VA system relating to its intelligibility characteristics is a relatively complex phenomenon and depends on multiple interrelated factors. Speech intelligibility depends on two speech characteristics: audibility and clarity. Audibility is related to the SNR at the listener or receiver position. Clarity is a function of distortions introduced along the transmission path which degrades the fidelity of the original speech signal. As in a chain structure the contributing degrading factors are cumulative along the signal transmission path.

The quality of the speech input signal in the first section of a VA system will have a determinant effect in the final intelligibility of the signal (or announcement). If the signal is pre-recorded electronic distortions and recording quality are the main factors. However if the signal is from a live announcer, factors such as message content, diction and rate of delivery together with microphone quality and acoustics effects of the announcer's room will be main contributors.

The potential degrading factors from the input, signal processing and amplification sections are mostly of an electronic nature which include electrical noise, non-linear distortions and limited bandwidth. The control and mitigation of the electronic degrading factors of the first three sections is relatively simple to attain. However speech signal degradation in the electro-acoustic transmission section is more difficult to control and reduce. In this part of the system chain, complex electro-acoustic interactions occur between the loudspeaker array and the room space. Consequently this section of the VA system is generally the most critical for achieving satisfactory speech intelligibility performance. These difficulties become particularly challenging when the loudspeaker array is installed in large enclosed spaces of complex geometry built with acoustically hard and reflective surfaces such underground spaces (figure 3:3)

3.2.5 Electro-acoustic section

The scope of this research concentrates on the electro-acoustic section of the VA system. For that purpose, it is assumed that listeners share the same first language as the announcer, the hearing ability of receivers is normal and the complete three previous sections of the system operate in optimal conditions.

The primary potential factors affecting the speech intelligibility performance in the electro-acoustic section of VA systems in underground stations are summarised below.

a) <u>Room reverberation:</u> Reverberation is the principal contributor to reduction of intelligibility in deep platforms. There are limited effective means to control reverberation particularly after the station has been built.
b) <u>Early to late reflected energy ratio.</u> This indicative ratio of reverberant energy is fundamental in the determination of intelligibility performance and design. It becomes highly correlated to intelligibility in noise-free situations (see D50). This factor can be partially controlled and optimised at the design stage in new and renovated systems.

c) Background noise: The main noise sources making up the station background noise are passenger activity, remote ventilation/fan noise, and train noise[9]. There are very limited effective means to control this factor.

d) Speech signal to Noise Ratio (SNR): This ratio is fundamental in the determination of intelligibility performance and design. This factor can be optimised retrospectively by design and other means.

e) Volume, proportions and shape of the space: These determine the distribution of the sound fields and the creation of late reflections or focusing effects. These geometrical factors also determine a) and b). There are no practical or feasible means to control these factors.

f) Loudspeaker-receiver distance: This is a fundamental variable to control and improve b) and d) available at design stage in new and renovated systems.

g) Loudspeaker frequency response: For satisfactory speech intelligibility, loudspeakers should be capable to reproduce the fundamental frequency range of human speech (from 125Hz to 8kHz). This reproduction should be free of distortion and of approximately of equal level across the frequency range (flat response). This factor can be partly controlled by loudspeaker selection at design stage in new and renovated systems.

h) Loudspeaker directivity: This is a fundamental loudspeaker characteristic used to concentrate sound energy on the listeners' plane. This factor can be controlled by careful loudspeaker selection

i) Loudspeaker distortion: Distortion is a form of noise masking and degrading the original speech signal therefore affecting the integrity the original speech modulations. This factor can be controlled by careful loudspeaker selection

As it can be expected and it is demonstrated in chapter 7, the most effective means to control high reverberation and consequently improve intelligibility performance in underground stations is introducing acoustic absorption. This remedial measure would control factors a),b),c),d). However the dramatic benefits of this solution have been traditionally rejected on grounds of increased costs, fire hazard, installation and maintenance difficulties and aesthetics.

[9] Normally, announcements are not broadcast when the train is approaching, leaving or at the platform. Train noise is not considered as a noise source for the purposes of this research.

Loudspeakers commissioned to be used in London underground stations must satisfy strict minimum performance specifications for optimal speech reproduction, safety, fire and dust ingress resistance, aesthetics and other mechanical and installation requirements (VA system specifications, 2005), (London Underground, 2004), (BS 5839-8:2008), (BS EN 54-24:2008). Those requirements limit the selection of loudspeakers commercially available.

In a VA system renovation without the option of introducing acoustic treatment, factors relative to the loudspeaker array become the only controllable design variables left to overcome the inherent acoustic difficulties of the space and achieve the required system performance. However even those variables can be severely constrained by practical installation and maintenance priorities (cabling routes, vandalism protection, accessibility). It can be safely and reasonably assumed that LU approved loudspeaker types (see appendix F) are fit for VA system purposes by being able to reproduce adequately the full speech frequency range of speech announcements at maximum SPL of 102-106dBA without significant distortion (see appendix F). Provided the above assumption, the adverse factors to speech intelligibility on platforms are reduced to reverberation and background noise related effects, that of factors a),b),c),d). This conclusion is critical as it focuses the research on finding the most efficient design strategies to achieve higher speech intelligibility performance.

3.3 Underground station acoustics

Most of London Underground subsurface station spaces can be classified into two typical distinctive geometries: 1- Spaces approximating to a parallelepiped of proportional dimensions and 2- Spaces of long, short and narrow volumes. Most ticket halls (TH) and concourses circulation spaces fall within to the first type (figures 3:4 and 3:5) while platforms and corridors fall into the second (figure 3:6). None of these spaces are acoustically completely closed, they contain openings (e.g. entrances, exits, ventilation shafts, tunnel openings) connected to other spaces. Depending on the amount and cross sectional size, these interconnecting apertures can act as effective acoustic absorption in the source room, create coupling effects and/or convey background noise from remote areas.

Traditionally subsurface circulation spaces have not considered acoustic quality or VA speech reproduction. Hence the limited remedial acoustic measures available tend to be costly, difficult to implement and usually have limited effectiveness.
Internal surfaces in stations are selected for durability, hygiene and safety reasons. Consequently, they are acoustically characterised by being large, flat, smooth, hard and highly reflective. These qualities promote the formation of standing waves, echoes, long reverberation times and increase background noise. Within the public circulation spaces there are usually very few fittings or objects in the way of the sound waves which could block, diffract or scatter sound and hence minimise the above detrimental acoustical effects.

Ticket halls and concourses can be regarded as approximating diffuse field based on their proportional geometry and homogeneity of reflective surface materials. Thus conventional classic acoustic theory approaches can be applied in the design and performance prediction of new or renovated VA systems.

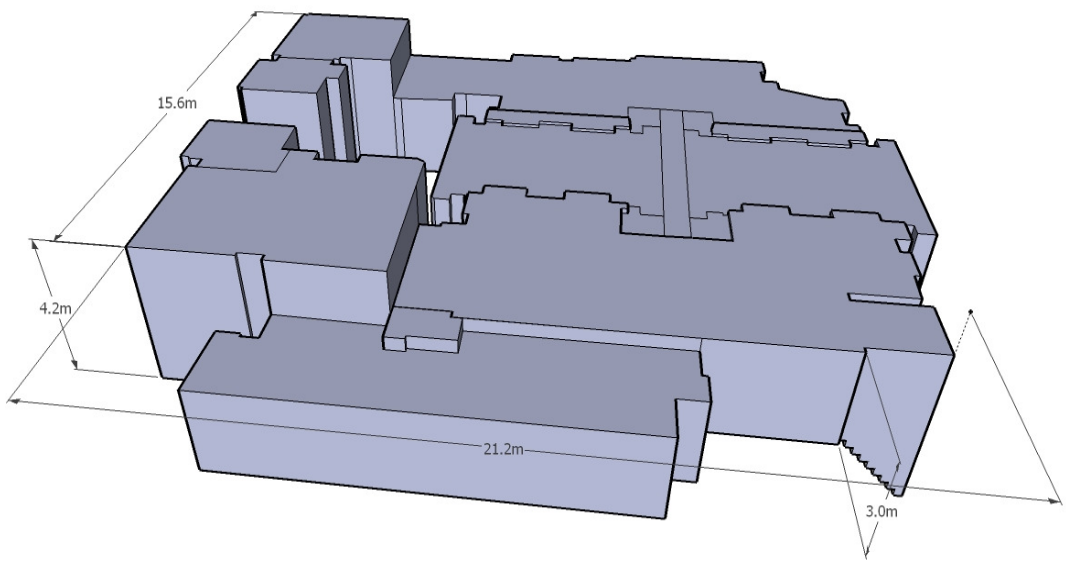

Figure 3:4 Main dimensions and shape of a small sized ticket hall

Volume and dimensions vary significantly among ticket halls and circulation spaces depending on station capacity, building location, and other architectural reasons. Ticket halls volumes can range from 300m^3 to 4000m^3. Large halls are typical for stations interconnecting with railway lines or airports.

Figure 3:5 Examples of four ticket halls of different sizes and geometries.

Underground station platforms and long corridors are disproportionate geometries acoustically considered long spaces in which one dimension (length) is much greater than the other two (width and height), although the other two are still relatively large with respect to the acoustic wavelengths of interest (Kang,1997b) (figures 3:6 and 3:7). Due to the platform's large dimensions relative to the acoustic wave length, acoustic wave theory in ducts is not applicable (Doak,1973). The most notable acoustic characteristic of long spaces is that the sound field generated by a single source is not diffuse; consequently classic room acoustics theory is invalid in long enclosures (Kang, 1996a). A long space cannot be characterised readily by a fixed length/cross-section ratio since the sound field is also affected by other factors, such as the width/length ratio and boundary conditions.

Hence Kang (1997b) suggested a criterion by which long space theory should be applied if the length of the enclosure is greater than six times of the width and the height.

As it will be shown in the next chapters, the acoustic field in a long enclosure excited by a single sound source is very different to the more complex field created by a multisource arrangement as it is the case of VA loudspeakers distributed systems. This fact is central in the potential design and performance prediction approaches to be used.

The prime acoustical characteristic shared by all deep platforms is their long reverberation time caused mainly by their large volume and prominence of acoustically hard and smooth surface materials. Typical deep platform dimensions are: 117m long (head to back wall), 6.4m wide (at widest point), 4.9m high (highest point from bottom of the track pit). The typical total internal surface is approximately of 2590m^2 and internal volume 3060m^3 (figure 3:6).

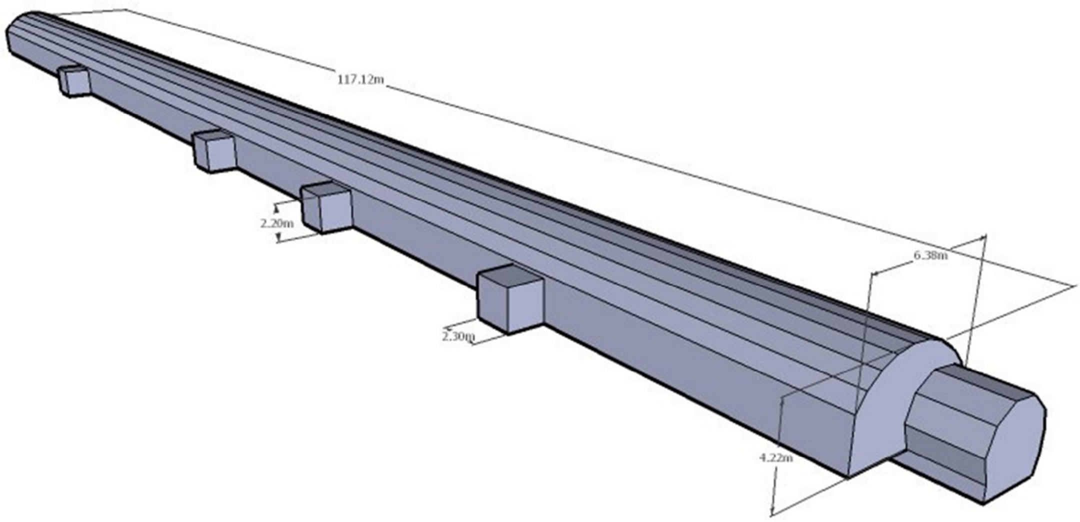

Figure 3:6 Main dimensions of typical London Underground deep platform

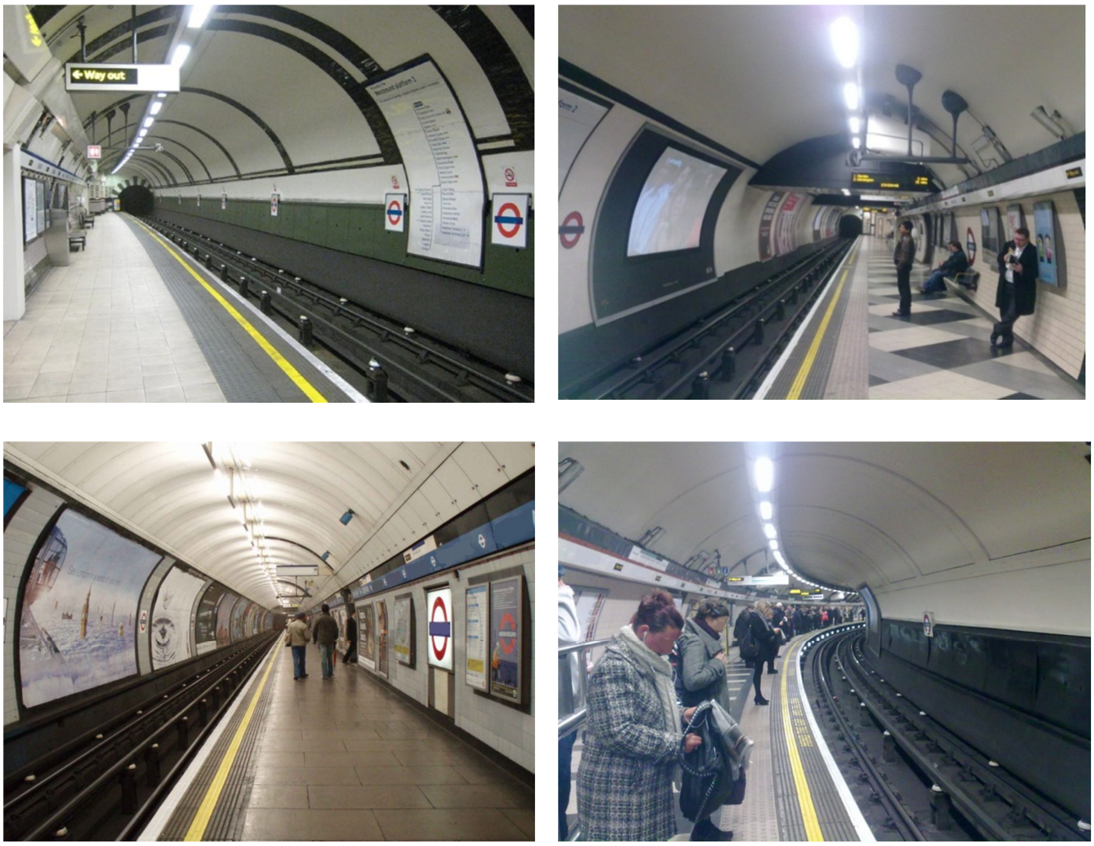

Figure 3:7 Examples of four London underground deep platforms.

The ceiling of most of deep London Underground platforms approximates to a concave dome. Under certain geometrical conditions, concave surfaces in an enclosure can give raise to focusing effects which can degrade speech intelligibility and coverage uniformity (Shield & Yang,1999a). However, the asymmetrical cross-section geometry of the platform space and the relative position of sources and receivers makes it highly improbable the formation of significant focusing effects on the listeners' plane (Shield & Yang,1999a). A small proportion of deep platforms have their ceiling covered by metallic panels for cabling containment, lighting and aesthetical purposes. In chapter 6 is shown that this type of dropped ceiling arrangement conveniently absorbs low frequencies within the speech frequency range by acting as a diaphragmatic absorber.

On highly reverberant subsurface spaces a distributed VA sound system approach is recommended (BS 6259:1997), (BS 5839-8:2008) and is implemented by metro operators around the world (London Underground, 2004), (Harrison, 2001). This conventional arrangement consists of a loudspeaker array, formed from equally spaced rather directional and low powered identical units, distributed in reverberant spaces such as concourses, ticket halls, long corridors, or station platforms. Loudspeakers in this conventional arrangement are all connected in parallel without delay, installed at approximately 3m height on the standing platform wall and aimed at the listening plane or ear height, (figure 3:8).

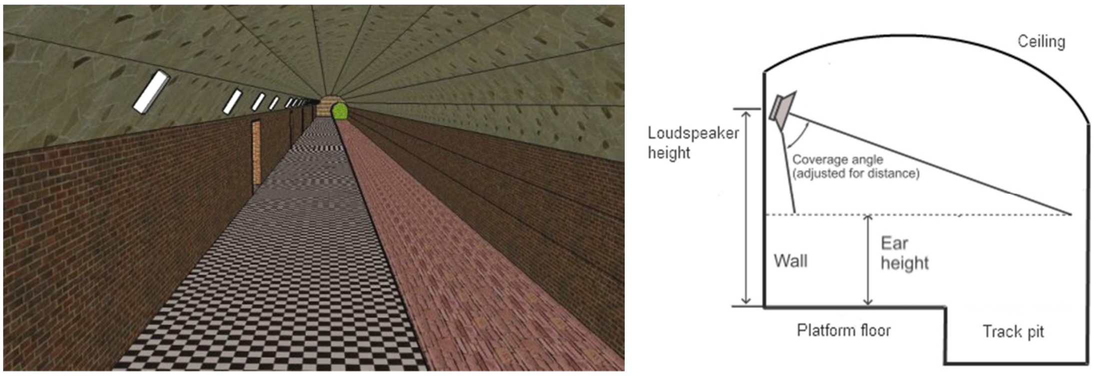

Figure 3:8 Conventional VA loudspeaker arrangement on LU deep platforms

This conventional VA loudspeaker arrangement on underground platforms aims to provide short source-receiver distance to achieve high direct to reverberant ratios (DDR) at the receivers. This also has the benefit of minimising acoustic energy spillage to the reverberant field. However, this approach is confronted by the need of high quantity of loudspeakers to provide coverage uniformity and practical issues regarding minimum height, cabling, cost, maintenance and aesthetics.

Subsurface ticket halls and deep platforms normally present steady and low levels of background noise across the speech frequency range during engineering hours[10] (figure 3:9), excepting when there are works occurring at those hours. Night time station background noise is normally from remote plant and ventilation systems, buzzing electrical systems as well as occasional remote traffic noise partially conveyed through ventilation ducts and platform tunnels. Figure 3:9 shows the average and standard deviation (σ) of background noise measurements of 20

[10] Engineering hours are between 01:30am to 05:30am to cover the time when stations are closed to the public, no trains run and engineering works are undertaken. For valid RIR measurements during engineering hours exclusive access to the station is required to avoid presence and noise from other trades at the time of measurements.

platforms and 12 ticket halls during engineering hours. Average of overall levels in ticket halls is $LAeq_{2min}$= 55dBA with σ= 4.2dB and $LAeq_{2min}$= 52dBA σ= 3.1dB on platforms.

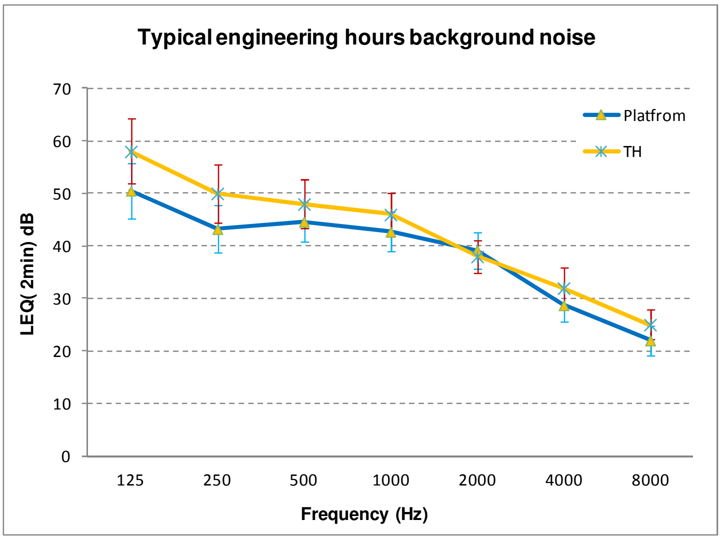

Figure 3:9 Typical subsurface platforms and ticket halls engineering hours background noise spectrum; bars indicate standard deviation.

The low level background noise during engineering hours allows a valid Room Impulse Response (RIR) measurements to be taken (see appendix B.1), (BS EN ISO 3382-2:2008), (BS ISO 18233:2006). This condition is referred as "noiseless" since the interference of the background noise components in the derived RIR acoustic parameters is nil. Measurements taken during engineering hours also facilitate the fulfilment of the LTI system conditions on which RIR are based (see appendix B.1), (BS ISO 18233:2006). The necessary test conditions for reliable RIR acoustic measurements (see appendix B.1) and the avoidance of aural disturbance to passengers by the loud test signals are the major reasons to conduct acoustic testing during engineering hours.

Considering RT and background noise as the two major degrading factors for VA speech intelligibility on underground spaces and assuming the remaining factors are fixed and optimal, one of the worst case scenarios leading to poor speech intelligibility will occur under minimum occupancy conditions. These are also the conditions during engineering hours. STI calculated from RIR measurements (see appendix B.2) taken during engineering hours is therefore obtained under maximum reverberation although noiseless conditions.

In prediction simulations the degrading effect of high occupancy noise can be accounted for by synthetically contaminating the noiseless STI with representative background noise (BS 5839-8:2008). In the case of underground spaces rush hours[11] traffic occupancy noise characteristics can be used. However it is important to note that in this attempt of simulating real conditions, the influential effects of reduced space volume, increase sound absorbency and diffusion provided by the occupants are not accounted in the noisy STI calculation which can lead to significant prediction errors.

Due to the high variability in occupancy noise conditions, for consistent comparative purposes within this research and in line with the sub contractor design and systems' commissioning procedure, STI values reported in this research were obtained from measured and simulated assessments in noiseless conditions.

On subsurface ticket halls and platforms the specified 10dBA SNR for adequate speech intelligibility from VA systems (see 3.4.3 below), is ensured by means of an Ambient Noise Sensor (ANS) system connected to the VA system. This system automatically and continuously monitors the overall background noise level (dBA) in the area of interest and adjusts the power amplifier gain accordingly to maintain announcement levels on the listening plane 10dBA above the existing background noise (see 3.4.3) (near to noiseless conditions). This system intends to provide a dynamic balance of adequate audibility for satisfactory intelligibility and acoustic comfort for most background noise situations[12] occurring during traffic hours (e.g. fluctuation of background noise due to rush and non rush hours). In case of emergency situation the ANS system is automatically disabled and the VA amplifiers automatically provide the calculated gain in the design necessary to reach a maximum announcement level of 90dBA in the station area of interest (see 3.4.3). This adaptive system therefore approximates real conditions to noiseless.

[11] Rush traffic hours is the time period of maximum occupancy of passengers/users in the station areas
[12] Except when trains are entering, at or leaving the station, then broadcast is normally disabled (VA system specifications, 2005)

3.4 Legislation, standards and performance criteria

3.4.1 Introduction

In the UK except for subsurface underground stations (see below), there is no legislation defining the requirements for the use, the specification and performance compliance of PA or VA systems. However National and European standards exist, which provide detailed codes of practice and recommendations on extensive range of aspects of those systems. These standards do not explicitly indicate when a system is required. The need for voice alarm system is normally determined by the relevant building licensing authorities or on completion of a risk assessment by the owner (BS 5839-8:2008).

Relevant standards in the PA and VA industry are widely adopted and implemented as reference for compliance purposes and occasionally for litigation or court cases. These standards are regularly upgraded, expanded and updated, suggesting the increasing importance given to life-critical sound systems as an essential evacuation and/or crowd management aid.

This section summarises the national and international standards directly applicable to subsurface stations VA systems. System specification documents laid out by London underground and appointed contractor company are outlined in detail. A critical review is provided on aspects of various documents and recommendations suggested for revision and improvement.

3.4.2 Historical background to the main documents

The 1988 public enquiry which investigated the London Kings Cross fire, prompted the Secretary of State made regulation under Section 12 of the Fire' Precautions Act 1971. This action subsequently created the Fire Precautions (Sub-Surface Railway Stations) Regulations 1989. The regulations stipulated amongst other requirements that all sub-surface stations should be equipped with fire detection and evacuation systems. These systems in public areas of the station were concerned with the use of VA system capable to reproduce live or recorded speech announcements.

London Underground were given until 31st December 1990 to comply with the Regulations which had been brought in during 1989. Any sub-surface station not meeting the requirements would face the threat of closure by the LFCDA (London Fire & Civil Defence Authority).

A relevant code of practice in the form of the British Standard BS 6259:1982 was available at that time. This standard provided general guidance for the design, planning and installation of sound systems for a wide range of applications, and proposed criteria for adequate performance. The revised version of this standard (BS 6259:1997) is currently used in the sectors concerned.

In 1988 the first part of BS 5839-1:1988 entitled *“Fire detection and alarm systems for buildings. Code of practice for system design, installation and servicing”* was published. This standard established for the first time standardised guidelines regarding fire alarm systems in buildings. The earliest part of standard (part 1) provided generic recommendations concerning the placement and SPL of sounders. Sounders are loudspeakers of very limited frequency range designed for efficient emission of attention drawing audio signals. A brief mention is made in this standard for the first time regarding speech intelligibility and BS 6259:1982 standard referred as guidance for systems relying on speech messages.

The first standard (part) dedicated to VA systems was first published in 1998 entitled BS 5839-8:1998, *Fire Detection and Fire Alarm Systems for Buildings – Part 8: Code of practice for the design, installation, commissioning and maintenance of voice alarm systems*. This standard was updated through a revised version released in 2008 which remains central to technical guidelines in the VA system industry.

London Underground Chief Engineer’s Directorate published in 2004 a standard entitled London Underground 2004, *SCS-ST-0008. Public Address systems on Sub-surface railway systems*, which defines the minimum technical requirements for the

VA systems installed in subsurface stations. This document is based on requirements stipulated in the Fire Precautions-Sub-Surface Railway Stations-Regulations 1989, BS 6259:1997 and BS 5839-8:2008. At the time of field research work (2007-2009) of this project this document was used as a specification reference for sub-contractors and operator companies working on behalf of LU.

An *Installer and designer contractor company* (or *Contractor Company*) was responsible for the improvement and maintenance works of communications systems including VA on various lines of the LU network. This company produced its own performance system specification document (*VA system specifications*, 2005) which was mostly a re-definition and interpretation of the details and requirements of one of the parent documents produced by its client (London Underground, 2004). This detailed document provided complementary measurement and performance specifications based on several relevant British standards (BS 6259:1997, BS EN 60849:1998, BS EN 60268-16 :2003, BS 5839-8:1998). The document is used in turn as a reference providing electro-acoustic information, guidance and performance criteria to be followed by sub-contractor companies working for the *Contractor Company.*

3.4.3 Performance specifications and relevant standards

Table 3.1 introduces the documents which detail the required electro-acoustical performance specifications for London Underground subsurface VA systems and relevant national and international standards which are referred to.

An expanded overview of the information provided in table 3.1 is provided in appendix C by presenting a compilation of the relevant extracts from each document listed. The extracts detail performance specifications, test requirements and recommended performance targets. For consistency those versions have been used throughout the duration of this research. Documents are presented in order of hierarchy of importance and relevance. The first two documents (see documents 1 and 2 below) are the contractual guidelines and specifications and served as the basis and reference for the purposes of this research.

	Document	STI	SNR (dB) A weighed	Max SPL(A)	Frequency response	Direct field Coverage	Distortion
1	*London Underground, 2004*	0.5	10	N/A	±2dB / 120Hz-9.5kHz	N/A	5% of second harmonic distortion
2	*VA system specifications,2005*	0.5	10	90	±2dB / 250Hz-6kHz ; over 100Hz-10KHz <5dB between 1/3 octave bands	±2dBA over 90% area	Not specified; for 1KHz tone at 70% output
3	*BS 5839-8:2008*	0.5	10	N/A	as per 8	N/A	N/A
4	*BS EN 60849:1998*	0.5	N/A	N/A	N/A	N/A	sine signal <17%THD
5	*BS EN 60268-16:2011*	0.5 - 0.46	N/A	N/A	N/A	N/A	N/A
6	*BS 6259:1997*	N/A	N/A	N/A	N/A	N/A	N/A
7	*BS EN ISO 9921:2003*	N/A	N/A	N/A	N/A	N/A	N/A
8	*BS EN 54-24 2008*	N/A	N/A	N/A	for loudspeakers; limits shown in a figure (appendix C.1)	N/A	N/A

Table 3:1 Summary of specified and recommended values of electro-acoustical performance parameters as detailed in documents 1 to 8 (see appendix C).

3.4.4 Discussion and recommendations

Establishing clear, suitable and feasible electro-acoustic performance specifications and guidelines is crucial in all stages of the VA design process to ensure that the system will be fit for purpose. This section provides a critical review and recommendations with special focus on the first two documents of table 3:1.

Firstly, the determination of BGNL is critical in the estimation of total sound power required by the system, the calculation of SNR and consequently the predicted speech intelligibility (STI). The specification of SNR in documents 1 and 2 (see appendix C.1) appears incomplete and ambiguous. It is suggested that the determination BGNL is furthered specified indicating the measurement parameter to be used, test duration, test locations, time of the day (rush or non rush hours) and other practical influencing factors (e.g. train arriving/leaving the station).

Secondly, the calculations of SNR based on overall A-weighed SPL single values are prone to produce unrepresentative and misleading SNR values depending on the temporal and spectral variability of the background noise being considered. It is recommended that SNR calculations are performed in octave bands at least covering the speech frequency range (125Hz- 8kHz). If an overall figure is to be used to determine SNR levels, the C weighting is more suitable as this weighting correlates more closely to perceived loudness (audibility) for the typical spectrum and levels broadcast on platforms (figures 3:10 and 3:11).

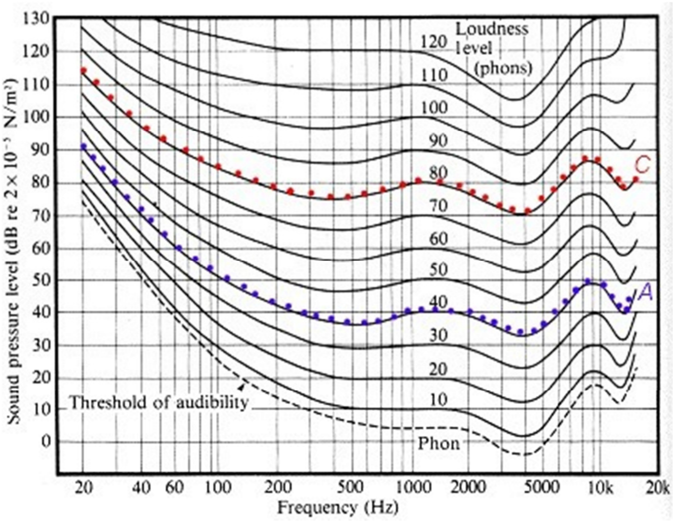

Figure 3:10 Fletcher-Munson equal loudness curves showing highlighted curves on which A and C weightings are based

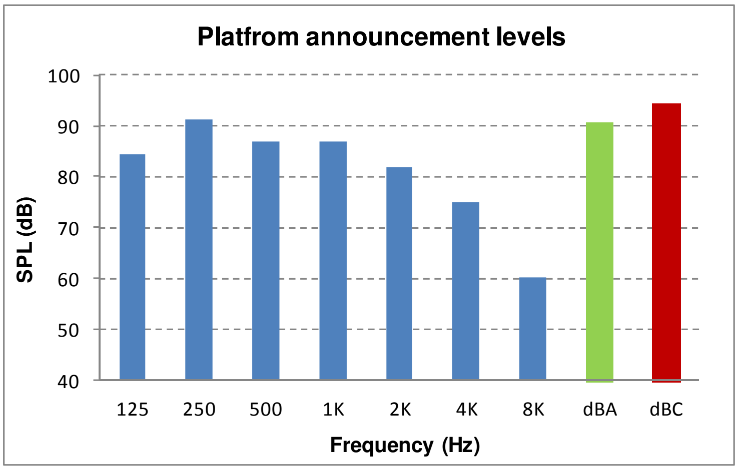

Figure 3:11 Platform VA system maximum announcement levels using speech shape noise input signal

Thirdly, consistent with BS 5839-8:2008 the primary documents require a minimum 10dBA speech to noise ratio (SNR) on platforms for acceptable intelligibility. However the author considers this ratio is not truly representative of effective audibility in relation to speech intelligibility on real situations since the measured speech signal at the receiver positions will be mostly comprised of degrading reverberant sound.

Fourthly, the specified direct level coverage given in document 2 is considered an unrealistically stringent requirement. To meet that specification on a deep platform it would require an unreasonable number of loudspeakers with no significant improvement in performance. In cases where loudspeaker installation constraints are imposed the coverage specification should be relaxed to reflect this condition. Moreover knowledge of theoretical direct field level is not indicative of prospective speech intelligibility or suitable to calculate realistic SNR since this field becomes dominated by the reverberant field under realistic situations. If knowledge of useful sound coverage is needed for the strategic placement of loudspeakers, a direct plus early reflections field level *(DERL)* coverage indicator is proposed by the author as a more relevant and realistic indicator of useful energy coverage.

DERL is defined as the SPL (dB) resultant from the useful speech energy registered at the receiver during the time window comprising the direct sound arrival and the subsequent 50ms of early reflections. From the impulse response this parameter could be calculated as expressed in equation 3:1:

$$DERL = \int_{-\infty}^{0} p^2(t)dt + \int_{0}^{50} p^2(t)\,dt \qquad \text{(dB)}$$

Equation 3:1 Definition of proposed DERL parameter derived from the RIR

Where p^2 is the *square of the instantaneous sound pressure and t is time.*

Fifthly, the operational frequency response (and flatness of its curve) as required by document 2, is considered unnecessarily stringent given that a significant degree of non-flat response is tolerable in speech intelligibility perception (BS 6259:1997). This requirement cannot be met on deep platforms due to the inherent acoustic limitations and performance of certified type of loudspeakers. This argument is in line with lower specifications suggested for equivalent sound systems in the literature (5839-8:2008),(BS 6259:1997),(BS 7927:2011). It is recommended by the author that the flatness specified in document 2 be indicated in dB instead of dBA and be lowered to ±3dB over the frequency range 250 Hz and 6kHz. A relaxation in the flatness specification should allow other important parameters to achieve their targets under the restrictions imposed in the design (e.g. architectural, cost). It is also suggested to limit the frequency range of broadcast signals (announcements and test signals) within the speech frequency range (125Hz to 8kHz octave bands) so to avoid unnecessary upward masking, strain of loudspeakers and save in power amplification requirements.

Sixth, the determination of the distortion parameter in document 1 is not consistent with document 2 and appears ambiguous. Harmonic distortion does not degrade intelligibility significantly, while inter-modulation distortion is often very destructive to speech intelligibility. Hence the author suggests considering inter-modulation distortion, rather than harmonic distortion, as the preferred measure of system linearity (BS 6259 :1997).

Seventh, subsurface VA system are designed and assessed for unoccupied conditions (i.e. engineering hours) (see 3.3). However document 2 and BS 5839-8:2008 (see appendix C) implicitly consider the detrimental influence of occupancy noise level during operation hours (traffic hours). However these documents fail to consider the potential dramatic effects of occupancy on reverberation, echoes, SNR and STI. This factor could be taken into account at design stages by means of acoustic simulation and integrate it in the requirements as a performance variable to consider.

Eighth, it is recommended to replace the inflexible STI target (0.5) indicated in documents 1 to 4 with the purpose-specific and tolerant targets suggested in the latest revision of the STI method BS EN 60268-16:2011 (see table 3:1 and appendix C).

Ninth, it is suggested that the speech shape noise signal weighting provided in document 2 be replaced with the weightings given in BS 7827:2011.

Finally, due to the interrelation among VA performance parameters in deep platforms, attempting to meet all performance specifications as laid in documents 1 and 2 can prove unattainable (see chapter 5 and 7), (Harrison, 2001). A full revision and update of documents 1 and 2 is strongly recommended to improve the design and performance delivery of subsurface VA systems in London Underground. The following objectives are proposed for the suggested revision:

- Avoid inconsistency between documents and ambiguity by providing clear and detailed specification determination.
- Review specifications in light of latest standards.
- Consider practical architectural, environmental and acoustic considerations when establishing the intended targets.
- Balance carefully the specification target setting by considering level of importance of each parameter (prioritisation/compromise) and impact on other performance parameters (competing parameter targets).
- Allow a tolerance band to each requirement specification.
- Specify lower parameter targets for special situations (e.g. train pulling in / out station, emergency fans in operation, installation constraints).
- Review regularly specification documents in light of new guidance (standards), measurement methodologies and feedback from designers and other stakeholders,

Considering the critical importance of VA systems in subsurface areas and that a substantial proportion of London Underground users fall within of a group of disadvantage listeners in terms of speech intelligibility (English as a second language[13], old age[14] or degree of hearing loss[15]), the author based on latest guidance (BS EN 60268-16:2011) recommends to upgrade the STI target to achieve qualification band E (0.56 to 0.6) for subsurface public areas. Current STI target for surface station areas is 0.6. In order to achieve this suggested target a financial and

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[13] Almost one quarter (23%) of all Londoners aged 16-34 have a first language other than English (Spence, L. 2006). According to a 2007 NIACE survey two out of five adults in London do not speak English as their mother tongue (Luddy, 2008)
[14] At 2009, 15% of Londoners were 60 years old or older, The over-65s are likely to increase by 33% (over 280,000) to reach 1.15 million by 2031 (London higher 2011)
[15] Within greater London, 25,290 people were registered in 2010 as deaf or hard of hearing (National Health Service 2011).

strategic re-prioritisation of the role of VA systems during renovation and upgrade works would be required by project management and other relevant decision makers.

Currently national and international standards providing VA electro-acoustic guidelines (see documents 3 to 8) appear disconnected and do not give complete and clear information on performance test methodologies. Guidance provided frequently overlaps and occasionally is not consistent across standards. Harmonisation and standardisation is especially needed for objective speech intelligibility measurement instrumentation widely performed with the STIPA method.

It is assumed that conventional deep platform VA system are well configured and exhibit optimal frequency response and distortion levels so they are not influential parameters on speech intelligibility. For the purposes of this research and for consistency with data obtained during the time of field research, the four most important VA performance parameters and their criteria targets as specified in documents (1 and 2) are considered in the following chapters. The parameters and performance targets are summarised in table 3:2.

Parameter	**Performance Target**
SNR	10 dBA
MaxSPL	90 dBA
Direct sound Coverage uniformity	±2dBA 90% area
STI	≥ 0.5

Table 3:2 Performance parameters and criteria targets as specified in the documents 1 and 2.

3.5 Conclusions

This chapter has provided electro-acoustical and practical background knowledge specific to VA installed in subsurface underground stations with the aim to equip designers and other stakeholders in the delivery of effective VA systems. Architectural and acoustic limitations as well as electro-acoustical factors have been particularly highlighted.

A comprehensive review of the applicable performance specifications and relevant standardised guidance has been undertaken A critical review has shown that contractual performance specifications were ambiguous, conflicting and occasionally unsuitable. Suggested practical and realistic revisions to the appropriate standards and guidance have been given to produce clear and balanced specification targets. The suggested reference document improvements will facilitate a more efficient and effective VA system performance delivery.

Chapter 4 Literature review

4.1 Introduction

This chapter provides a literature review of the relevant research works. In addition, to improve readability, a further literature review is provided specific to each chapter. Research works on calculation algorithms and prediction models are critically discussed with comments on their potential accuracy and reliability. The review is divided in three sections. In each section chronological order has been followed except in certain cases to aid readability.

4.2 Acoustics of long spaces

This section reviews the literature concerning research on the acoustics of long spaces excited by a single source or multiple sources in various types of long spaces including long built up city streets, indoor corridors, traffic tunnels and underground station platforms.

4.2.1 Single source

The acoustics in long spaces has been studied from the early 1960's (Yamamoto 1961, Davies 1973, Said 1981, Redmore 1982, Kuttruff 1989). These studies were dedicated to determining the special acoustics observed in long spaces mainly concerning the sound propagation along the length and deriving sound field prediction formulas. An application of those studies was the prediction of sound attenuation in long corridors. Based on various assumptions a number of formulae were developed although their application range was limited.

Reverberation time (RT) in long spaces was first considered by Schroeder (1973) who took extensive RT measurements of long town streets, which can be considered in effect as long enclosures with a highly absorbent ceiling. Kuttruff

(1975) considered RT and absorption in the derivation of formulae developed to calculate noise levels in city streets. The reverberation caused by street buildings as a contributor to the total traffic noise levels in towns was investigated by Steenackers et al (1978). In their study they observed that decay curves were not representative of a diffuse field. Sergeev (1979) used an image source model to derive simple formulas to estimate the propagation of sound in city streets and long tunnels, where he noticed that RT spatial distribution was fundamentally different than that of diffuse spaces. However this study on RT was restricted and did not yield measurement results or a usable relationship. An approximate theoretical relationship for calculating the RT in rectangular tunnels was developed by Hirata (1979) assuming that the source-receiver distance was small compared to the peripheral length of the cross-section. In this work he also noticed the RT characteristics to be fundamentally different from that of the diffuse field.

Westerberg (1986) measured the effect on sound propagation and RT of adding acoustic treatment in an underground platform station in Stockholm. It was observed that RT increased with distance from a single source in both cases with and without acoustic treatment applied. The addition of treatment reduced the RT at all frequencies by more than half and this reduction was even more pronounced at low frequencies.

From a series of underground station platform measurements of SPL and RT using a single source, Barnett (1994a) observed that the acoustic field of station platforms show characteristics different from diffuse spaces where classic acoustic holds true. RASTI predictions were provided from an empirical formula based on the early MTF algorithm (Houtgast et al, 1980). In this formula input values were derived from SPL attenuation and RT predictions provided from simple empirical formulae. Validation of RASTI prediction results showed agreement with measurements although results against full STI were not reported. These simple formulae have not been utilised or further developed possibly due to the lack of robustness, detail in the method and formulae construction or commercial value.

The above early studies were all based on simple and limited conditions where significant assumptions were made. Although those studies formed the precursor step for further knowledge developed years later, they were not sufficiently advanced to be utilised effectively and confidently in practical design applications.

Extensive and rigorous research was undertaken by Kang (1996a,b,c) to determine the acoustic field of long spaces and design factors involved. In this initial research a series of measurements and theoretical analyses using the image source method demonstrated systematically that the sound field generated by a single source in long enclosures, whether with geometrically or diffusely reflecting boundaries, is far from diffuse and classic formulae are unsuitable in long enclosures. It was theoretically and experimentally demonstrated that in long enclosures the sound pressure level decreases continuously and with the increase of source-receiver distance while the RT increases steeply to a maximum and then diminishes slightly. The decay curves are convex, especially in the near field. Kang (1995) also experimentally demonstrated through measurements taken in a 1:16 scale model of a London underground subsurface platform that diffusers placed on the boundaries are an effective technique for increasing sound attenuation along the length of long spaces.

These first single source works set the basis for further research also undertaken by Kang where multiple sources were utilised to determine the sound field of underground platforms equipped with distributed sound systems along the platform (e.g. VA system). These early theoretical works and the derived formulae were based on geometrical reflection and they were mostly limited to prediction of sound propagation and RT along rectangular long spaces of uniform hard and smooth boundaries.

Computer models utilising the ray tracing technique were created by Yang and Shield to predict and study the acoustic field in long enclosures. Models were first developed using a single source with the aim of predicting propagation, acoustic indices and speech intelligibility in long enclosures particularly underground station platforms (Yang et al, 1996), (Yang & Shield, 1998), (Yang & Shield, 2000), (Yang & Shield, 2001). The prediction model developed by Yang and Shield (2000) could predict SPL, EDT and RT parameters in rectangular cross section spaces taking into account diffraction effects of the tunnel portal and source directivity. In a further developed version of the model (Yang & Shield, 2001) platforms of semi-circular cross section could be exactly described analytically and the impulse response could be calculated from which a number of acoustic and speech related parameters were derived (SPL, RT, EDT, D50, C50, STI). The algorithm for calculating the STI was based on the ray tracing method of Rietschote et al (1981), which was modified to allow for the correct calculation of the MTF's in a non diffuse long enclosure by also taking account of the air absorption and background noise. The resulting models were validated respectively against scale model measurements of a curved ceiling London underground platform and measurements of an actual Hong Kong station platform of rectangular cross section using in both cases a single source. Prediction results along the length of the platforms showed general close agreement with measurements. Inherent inaccuracies were observed in the 125Hz octave band of EDT due to wave interference effects and in the source near field due to source and receiver sizes relative to the wavelength. Reported predictions of the overall STI figure were particularly accurate with an error less of ±0.03 and ±0.01 in the near and far field respectively. The model as acknowledged by the developer is only suitable for an overall investigation of the sound field. The model requires simplification of the space and other reliability and accuracy weaknesses included the lack of diffuse reflection and absorption angle dependence. Air absorption was only accounted for by a single combination of temperature and relative humidity. The calculation procedure to predict the STI was based on the early MTF method (Houtgast et al, 1980) and naturally did not include auditory masking, absolute reception threshold, gender and redundancy factors incorporated in BS EN 60268-16: 2003.

Imaizumi et al (2000) investigated the characteristics of sound propagation and speech transmission along a tunnel with a branching “T” shape intersection using both the conical beam prediction method and actual measurements. His studies although somewhat related theoretically to the acoustics of long enclosures bear limited relevance to underground platforms with multisource systems.

Reverberation in rectangular long enclosures with diffusely reflecting boundaries was systematically analysed by Kang (2002) using a computer model developed based on the radiosity technique. Results from the computational analysis showed that RT30 increases continuously and the EDT increases rapidly until it reaches a maximum and then decreases slowly: the decay curves are concave in the near field and then become convex. With diffusely as opposed to geometrically reflecting boundaries, the sound attenuation along the length was notably greater, the air absorption was more effective with regard to both reverberation and sound attenuation. However, the conclusions offered can be of limited relevance and low applicability since the radiosity method is only suitable for high frequencies. Moreover, results were calculated from a single omni-directional source condition and the model was not validated with real measurements.

The effect of phase and interference of multiple sound rays reflections from boundary surfaces was ignored in all the incoherent models works mentioned to this point. This factor was incorporated by Li & Iu (2004) in their prediction models by coherently summing the contributions from the image sources, although only sound propagation from a single source was investigated. Li & Lam (2005) extended the coherent image source approach to evaluate the impulse response of sound in a rectangular cross section long enclosure from which RT30, EDT and STI could be derived. The model was validated in three long enclosures of different impedance boundaries. Results showed that the model technique can predict slightly more accurately than incoherent models especially for narrow frequency band analysis.

More recently, related research has focused on the study of the sound field and sound propagation in long enclosures generated by a single source in the presence of multiple branches (Liu & Lu, 2009, 2010). To take account of wave effects in these studies they utilised scale models and developed a combined prediction method based on wave theory analysis. These studies showed that the sound field of long enclosures with multiple branches is more complex and inhomogeneous than that of the long enclosures without branches or with one branch.

4.2.2 Multiple source

Studies on the acoustics of long spaces have consistently demonstrated that the sound field when excited by a single source is not diffuse and therefore classic acoustic theory is not applicable. Hence new theory has been formulated for several cases and geometries with a single source. This knowledge has provided the basis for establishing the rather different and more complex sound field generated when the space is excited by a multisource system. In most real world applications a multisource system is typically utilised for speech communications (e.g. PA, VA). However there is much less knowledge and research available in the literature on sound fields in long enclosures created by multisource systems.

The first research work found in the literature on multisource systems in long enclosures was a specific empirical study comparing the relationship between objective and subjective speech intelligibility measurement methods, obtained from VA systems installed on London underground platforms (Barnett,1993), (Knight & Barnett, 1993). In these investigations speech intelligibility on three subsurface platforms was assessed by using STI and RASTI scores derived from RIR measurements and utilising word scores methods. The validity of the jury test using binaural recordings to accurately determine speech intelligibility was shown. The PB word scores test was found to be more reliable than MRT test method. It was also shown that full STI measured results on the three platforms consistently correlated closely with PB word scores test; this finding is very important since it validates the use of the indirect STI method as an accurate predictor of speech intelligibility on underground platforms equipped with VA systems. However RASTI measurements were considered misleading and unreliable due its intrinsic limitations.

A physical scale model of a London underground subsurface platform incorporating a VA distributed sound systems was utilised by Orlowski (1994) as a reliable tool to investigate the effect of acoustic treatment and its location on STI measurements. His measurements showed that a substantial amount of acoustic treatment suspended from the curved platform ceiling was capable of increasing STI scores on the listening plane by up to 0.3. It was found that with the loudspeaker spacing of 4m, STI spatial variability between loudspeakers was low between on axis and off axis positions, when no treatment was applied. With treatment applied STI spatial variation between speakers was slightly higher. However, in the report, no representative platform background noise appeared to have been incorporated in the determination of the STI. Although the overall findings can be valuable for the present research, it is important to note that due to the date of the study redundancy and auditory effects factors were not included in the determination of the STI values reported.

Based on previous investigations with a single source, Kang expanded his research into the acoustic of long spaces featuring multisource arrays with particular attention to the case of underground station platforms equipped with a distributed loudspeaker system. A theoretical and computer hybrid model, MUL, was developed by Kang (1996d) for calculating acoustical indices of multiple sources in long spaces by summing the contributing effect of each source. The calculation principle considers that if the sources are equal and the boundary conditions are uniform along the length of the long enclosure, the effect of any source on a given receiver can be determined from obtained data of a single source. Hence the acoustic parameters of multiple sources at a receiver can be calculated by summing the sound energy from all the sources. The model was tested in a hypothetical rectangular enclosure and validated by measurements of RT, sound distribution and the STI in a 1:16 scale model of a London underground station. The computations showed that the reverberation at a receiver can be significantly increased by adding sources beyond a single source and could be decreased by adding sources between a single source and the receiver. RT can be reduced and STI increased with additional absorption, although excessive absorption can be detrimental to STI. The STI calculation error for 5m loudspeaker interval was found to be 0.08. MUL was later utilised to investigate the effectiveness of some architectural treatments on the STI by introducing as input data series of single source acoustic measurements carried out in a 1:16 scale model of a London underground station platform (Kang,1996e). This investigation suggested design guidelines when using the proposed architectural treatments. It showed how the STI from a distributed loudspeaker system in a long enclosure can be improved by strategically placing porous absorption, ribbed or Schroeder diffusers, reflectors or a combination. Of all the treatment options considered, only treatment along the length was reported as quantitatively effecting STI. However these results were derived from theoretical model predictions based on single source measurements. RT results were only reported for 630Hz. From the date of the study and the references provided it is assumed that redundancy and auditory effects factors were not taken into account in the calculation of the STI. Reported STI values appeared to be calculated using the RASTI method and only at high SNR's. The MUL calculation procedure exhibited the following significant simplifications which can lead a considerable uncertainty in real world applications: single source decay curves were assumed to be linear, end walls were assumed completely absorbent, boundaries conditions were supposed uniform, absorption coefficient was taken as angle independent and based on geometrical reflection, multisource EDT approximated from decay SPL calculated from simple level summation of contributing sources, diffraction effects and diffuse reflection on hard surfaces were not considered, calculation of the STI did not include auditory effects. Based on predictions and investigation utilising the MUL model, design guidelines were developed on effectiveness of architectural treatments on platform spaces for the improvement of STI from multisource systems. Although the guidelines are valuable in nature, most of the specific solutions proposed are unlikely to be implemented in real underground platforms due to various practical impediments (e.g installation, maintenance, safety, cleaning, cost, fire resistance, aesthetics). Kang also developed a semi empirical calculation method for predicting acoustic indices in long enclosures of multiple loudspeaker PA systems (Kang, 1997f). The method preliminarily calculates the acoustic indices of a single source by existing theories (Yamamoto,1961),(Kang,1996c) or by computer simulation models (Yang et al, 1996). The possible errors caused by the preliminary calculation simplifications are then corrected using an empirical database based on results from two physical scale models. With the corrected data of the single source, the acoustic indices of multiple sources can be calculated. The model was validated in a rectangular double platform Hong Kong MTR station. It was acknowledged by Kang that the calculation method should be improved by developing a more accurate algorithm of simple conditions and by establishing a more detailed empirical database. The use of the RASTI[16] instead of the full STI method and the undefined correction process could potentially lead to unreliability in the predicted results. In a similar study Kang (1998) investigated in detail the effectiveness of a series of architectural acoustic treatments for improving the STI from a multiple loudspeaker system. Measurements were undertaken with a single loudspeaker in a 1:16 scale model of a rectangular Hong Kong MTR station island platform simplified as a shortened long parallelepiped. The single source data was used with MUL to produce multisource acoustic indices and the investigations of interest. Practical conclusions and guidelines drawn on the use of architectural treatments agreed with his previous work. Results showed that STI values were in good agreement with the measurements for receiver positions on or off axis of loudspeakers and that ribbed

[16] RASTI method is now formally obsolete and its use has been recommended to be discontinued due to its proven inaccuracy as a method to predict speech intelligibility from PA/VA systems (see chapter 2).

diffuser, membrane absorbers and strategic designs of obstructions can be used to improve the STI.

Although this semi empirical investigation provided practical design guidelines when using the proposed architectural treatments, full reliability and applicability of results cannot be assumed in real world practical designs for the following reasons: reported acoustic indices including STI values were calculated using the MUL model using single source measurements and without interfering background noise, the scale model utilised was an approximation of a specific platform type, single source measurements were taken for octaves up to 2kHz which suggests that RASTI was calculated instead of STI calculation was limited up to that frequency, end wall conditions were not representative of real conditions.

The ray tracing model developed by Yang & Shield (2001) was adapted for multisource predictions and validated using multisource arrangements (Shield & Yang, 1999b) by comparing predictions against measurements taken by Barnett (1994b) of a distributed loudspeaker system in a real London underground station. Good agreement between measured and predicted STI values was reported although no data was provided. The model was also used to investigate the effect of several factors on the predicted STI from a five and ten loudspeaker arrays in a hypothetical simplification of a semi cylindrical underground platform. Predictions of SPL and STI in the semi cylindrical platform agreed with the findings of other researchers (Barnett,1994b), (Orlowski,1994) whereby values were almost constant along the length of the platform covered by the multisource array as also reported by Kang (1996d) which resembled diffuse field conditions. Other practical findings were drawn from analysing predicted STI along the length by varying combined or individually the following factors: number of sources (loudspeakers), background noise level, platform end absorption and listening plane absorption. Most of the findings on the effect of those factors on the STI, fundamentally agreed with results obtained by Kang using the image source prediction technique (see above). From the findings obtained some design guidelines were provided to improve the STI by determining the optimum number of sources as a function of the prevalent background noise. An interesting solution was proposed involving the dynamic reduction of number of active platform VA loudspeakers depending on occupancy and the existing background noise in an attempt to reduce the reverberant effect created by distant loudspeakers. However this concept in principle would not conform to the specification criteria for coverage uniformity (VA system specifications, 2005). The principle on how multisource acoustic indices were calculated was not explained. The model did not consider diffuse reflection and the STI calculation did not include redundancy and auditory effects factors. The validation of the multisource model was rather limited and not sufficiently documented. The model has not been further developed and is not available for use.

Yang (1997) briefly proposed an analytical description and theoretical explanation of the sound field in a long enclosure created by a multisource array formed of equal sources evenly spaced. Utilising her ray tracing model in a cylindrical cross-section simulation of a station subsurface platform with a multisource array, it was observed that predicted SPL and STI values were almost constant along the length of the platform. These observations agreed with measurements by Barnett (1994b), Orlowski (1994), calculations by Kang (1996d). This sound field homogeneity was also seen inside a cross section of the same platform. These observations led her to suggest a *Quasi diffuse sound field theory* by which the sound field of a long enclosure of hard surfaces excited by an evenly distributed multisource system is analogous to a quasi diffuse field where classic room acoustic approximately applies. This conjecture was validated by comparing RT, SPL and STI calculations based on the Quasi diffuse theory against ray tracing predictions and actual measurements, as reported by Barnett (1994b). Although the theory was not systematically expounded and extensively validated, it constituted a conceptual and useful framework to better understand the sound field created by a multisource array in long enclosures. In chapter 8 this theory is verified and expanded.

Kang & Orlowski (2001) presented some practical acoustic design guidelines regarding the installation of acoustic treatment based on the specific research and design work for the new subway platforms in Hong Kong. Previous theoretical models (Kang,1996c), a simplified RT prediction formula (aka Kang-Orlowski equation) for underground platforms and scale model measurements were utilised for producing the design guidelines. Due to the constrained validity conditions of the Kang-Orlowski RT formula and limited applicability of the recommendations specific to those platforms, the guidance provided is not useable for subsurface platforms found in the London underground.

An empirical/deterministic hybrid prediction method was proposed by Harrison (2001) with the aim to remove the need to describe station platform geometry as required by the image-source and ray tracing techniques. The method is a development of the early calculation models by Houtgast et al (1980) by applying the quasi-diffuse theory introduced by Yang (1997). In this method only the distance between the receiver and the loudspeaker contributing most to the local broadcast level is used in the calculation and is also assumed that speech from distances longer than the reverberation distance acts as if it were uncorrelated background noise. For its validation, measured RT and background noise data of a London Underground platform reported by Yang (1997) were introduced in the calculation to predict RASTI values along the platform. A good spatial agreement was found, with a RASTI maximum error of 0.05. The method was presented as a convenient design tool to test loudspeaker spacing to provide required RASTI coverage on the listener plane. However for practical industrial applications the need for RT and background noise measured data to perform the speech intelligibility predictions offset the benefits of this simple model[17]. The reliability and robustness of the method for industrial purposes is questionable due to the following: reported results are from the unreliable RASTI method, the calculation principle is based on two significant simplifications not extensively validated and cannot consider platform end wall reflection effects, the STI calculation method did not incorporate auditory effects, the method has not been further validated for different type of electro-acoustic and architectural factors encountered in real subsurface platform VA system installations.

A computer prediction model run on a web browser was implemented by Dance (2007) based on a simple prediction model (CISM) (Dance, 2003) with the aim to offer a fast and freely available acoustic prediction resource. This image source based tool is capable to simulate hypothetical long spaces with multiple sources requiring minimal input information and yield octave band predictions of SPL, EDT, RT and overall STI values. The CISM model was tested in an imaginary noiseless

[17] Simple models refer to simplified prediction simulations and unlike commercial models require minimal input data and skill to yield key output acoustic parameters. They require low computational power, short computation times and are normally inexpensive or freely available.

120m long parallelepiped including a ten multisource array against results predicted by commercial[18] advanced prediction program CATT-Acoustics v.8f (CATT) (Dalenback, 2007). A good EDT agreement was seen at all frequencies although significant discrepancies were observed on STI values. Moreover multisource STI measurements on a curved ceiling London underground platform (Barnett, 1994b) were compared against predictions by CATT and CISM programs, both simulating the actual platform geometry as a simple long parallelepiped with a distributed array of omni-directional sources. It appeared that CISM consistently overestimated the STI respect to measurements and CATT predictions. The CISM model in its current development version exhibited the following weaknesses limiting it accuracy for practical purposes: Only omni-directional sources are accepted, reflection order limited to 14, number of sources limited to 18, unsuitable to model semi circular cross section or more complex platforms geometries, diffuse reflection not included.

A preliminary comparison was undertaken (Dance et al, 2008) to establish the viability of utilizing simple models in the prediction of key acoustical parameters on deep underground platforms. The simple models considered in the comparison included the Sabine formula, Kang-Orlowski formula, CISM, an early version of the empirical model presented in this thesis in chapter 5 and CATT v8.f. In a hypothetical station platform, CISM model predicted EDT along the length agreeing closely with calculated Sabine values. These results were consistent with Yang's quasi-diffuse theory. Simple models predictions of RT and STI were compared against measurements of a curved-ceiling London Underground platform (Barnett, 1994b). RT measurements agreed closely with Sabine and CISM calculations, however Kang model underestimated RT significantly. Overall, the combined operation of prediction models showed acceptable errors in the prediction of STI. The empirical model using CISM or Sabine predicted RT were the most accurate calculation model combinations. From the models' limited simulation flexibility, capabilities and unreliable prediction accuracy it appeared that none of the simple models or their combined operation was yet sufficiently developed to provide the

[18] Commercial computer simulation programs or "commercial models" refer to acoustic software commercially available widely regarded as advanced, highly accurate and reliable for the prediction of room acoustics or/and electro acoustic system performance. Their effective utilisation requires geometrical description of the enclosure and several other detailed input data together with skilled usage. They tend to be costly and require relatively long computational times in complex cases.

satisfactory performance required in real world deep platform applications. However, that pilot comparative study introduced the concept and potential of utilising simple prediction models as preliminary prediction or diagnosis tool for representative straightforward cases under certain applicability conditions.

4.2.3 Other related research

Traffic tunnels are long enclosures normally equipped with VA alarms systems which exhibit some similar characteristics and challenges as those found on underground station platforms.

A procedure was developed (Wijngaarden & Verhage, 2001) to predict the STI metric in traffic tunnels as an aid to design. The main STI factors considered were loudspeaker type, loudspeaker position relative to noise sources, and loudspeaker spacing. The procedure makes use of the commercial hybrid acoustic simulation software Odeon (see chapter 2),(Christensen,1998a) and a post-processing stage to calculate the STI index contaminated synthetically with representative noise. A strong correlation (r=0.89, n=154) was obtained between the set of predictions and measurements. Furthermore predictions and actual measurements closely and consistently agreed at specific points along the tunnel validation. These prediction results suggests the potential reliability of the utilisation of advanced commercial hybrid simulation programs in similar long enclosures such as underground platforms equipped with VA systems.

Wijngaarden & Verhave (2006) developed a simple model approach to provide an easy to use and quick STI prediction tool to aid the non-acoustic specialist in the design of VA systems in traffic tunnels. The model is based on fixed non-linear regression statistically deriving prediction functions from measured data and Odeon predictions based on different traffic tunnels. The preliminary prediction validation stage showed an average absolute error of 0.05 STI. Then the non-linear regression prediction approach was compared to solely Odeon based predictions and found that predictions matched those of the Odeon model with an average absolute discrepancy of 0.04 STI. This attempt to produce a valid simple model specific for traffic tunnels exhibits limitations and weaknesses in its reliability which may caution the concept against being transferred for underground platforms. The method comprises of the following reliability weaknesses: the calculation does not incorporate STI level dependence factors despite the expected high absolute signal levels of VA systems, geometry plane curvature were approximated by planes, predictions validated by another prediction technique, the calculations did not consider conditions with background noise which is the main STI degrading factor in traffic tunnels, applicability is limited to specific type of loudspeaker configurations, tunnels geometry and within certain input ranges. These limitations were acknowledged by the authors.

Odeon v6.01 was also employed for the design and optimisation of the installation of acoustic treatment on three new rectangular cross section underground station platforms and ticket halls for the Ankara (Turkey) metro system (Sü & Çalışkan, 2007). Acoustic parameters including RT20, SPL and STI were predicted based on a grid response on the listener plane. Absorption material type, quantity and locations were mainly investigated as part of the optimisation process to achieve the performance targets specified in the relevant Turkish regulations. The use of an advanced commercial prediction program was reported by the authors as an invaluable flexible tool for an efficient design of complex geometries.

4.2.4 Discussion and conclusion

Only three authors in the literature have reported acoustic and speech intelligibility related experimental measurements on underground spaces equipped with multi source systems. Of those only two provided data of London Underground subsurface platforms. However the data provided was very limited in detail and scope. These facts indicate the clear lack of acoustic measured data of underground stations and in turn suggest the little research undertaken on the subject at hand.

A significant amount of research has been undertaken into the sound field of long enclosures generated by a single source (generally assumed omni-directional). Although this knowledge has laid the basis for more complex cases, findings and derived guidelines from those studies are not directly applicable or suitable for the underground station platform case which normally incorporates a directional multi-source array (VA system).

Only a few studies have been found in the literature pertaining to the acoustics and speech intelligibility created by multi-sources in long enclosures. None of these works described the sound field created by multisource array as rigorously and extensively as expounded for single source studies. A simple energy addition approach was employed to derive multisource acoustic field parameters from the knowledge of one source of the multisource array.

Almost no studies have been found investigating the acoustics or the acoustic performance of multisource PA or VA systems in station ticket halls. This lack of research attention is due to the assumption of those spaces being acoustically less complex, thus allowing conventional classic acoustic approaches to be applied.

Practical investigations utilising scale models of underground station platforms were conducted with the main aim of VA system design and optimisation. Although this prediction technique is regarded as highly accurate, its use is impractical in real design of a large set of underground stations due to the high cost, labour and the associated lack of design flexibility.

Much of the relevant research on long enclosures with special attention to underground platforms was undertaken by Kang. Two pioneering prediction image source models were specifically developed to model the acoustics of long enclosures. These models were capable of describing the basic characteristics of a multisource array and yield multisource based acoustic and intelligibility predictions. These models were then utilised to provide an extensive investigation and subsequent guidelines on the effects of varying design factors on the acoustics and predictors of speech intelligibility of rectangular cross section platforms with multisource systems.

However the image source technique may appear restricted for modelling some underground platforms by its inherent limitations in handling curved surfaces, large number of planes and representing the reverberation tail. The image source technique is most suitable for spaces of relatively simple geometry or geometrically simplified. Multi source prediction models by Kang have not been further validated or developed, and not available for use.

Conversely the ray-tracing and hybrid methods do not require geometric simplifying assumptions and can predict the sound field of any space regardless of geometrical or architectural features. Generally ray tracing modelling technique possesses a higher applicability and versatility required in iterative industrial investigations. However, to date only Yang has developed a ray tracing prediction computer algorithm dedicated to the study and prediction of the acoustic in long enclosures. It was shown that the ray tracing technique developed for long enclosures required adjustments to allow for the correct calculation of the MTF's from a single source in a non diffuse long enclosure. Yang briefly proposed the Quasi diffuse field for underground platforms which can prove very useful in the practical design of platform VA systems. Chapter 8 offers a verification of the theory and expanded knowledge derived from it.

A few simple models have been tested for predicting the acoustics and speech intelligibility on multisource underground platforms. They were utilised in an attempt to simplify the acoustic prediction and design processes of subsurface platform VA systems. Although their purpose would find useful application in the early design stages and in situation of similarity of spaces, their level or development shown could not offer the design flexibility and sufficient prediction reliability for industrial requirements.

All the models and prediction approaches presented above rely on significant simplifications, have not been sufficiently developed or extensively validated for the level of accuracy and modelling flexibility required in real world subsurface station projects. Moreover, except the CSIM, none of the above prediction models is available for use.

From the information found during the literature review, it is here hypothesised that commercial hybrid models are the most suitable and reliable prediction tools for investigation, design and optimisation of VA systems in subsurface platforms. The robust prediction and flexible design capabilities of these models can satisfy the contractual requirements of real world metro projects. This hypothesis will be validated in Chapters 6 and 7.

Although commercial simulation programs are used in the industry for the design of VA systems on underground station platforms, no study has been found in the literature validating, assessing or comparing their descriptive and predictive capabilities in underground stations. Similarly, no research on the acoustic characterization or practical design guidelines of VA systems on underground platforms obtained through utilization of commercial simulation programs have been found. These identified knowledge gaps are filled in chapters 6 and 7.

Chapter 5 Platform empirical prediction model development

5.1 Introduction

This chapter presents extensive measured acoustical data and architectural surveyed details of a large number of deep platforms of the London underground network. This large set of valuable data has been compiled into a database with the purpose of providing a baseline of acoustic conditions, architectural details and speech transmission quality found in a large sample of the above platforms. The second objective of this chapter is to develop a pilot empirical simple model from the above database able to predict platform speech transmission quality from conventional VA systems with a minimal amount of input data. As a purely empirical based approach, the simple prediction model developed will have the advantage of fully relying on experimental data and therefore offer a strong confidence in the predictions yielded. However the convenient alternative of introducing input predicted values instead of measured ones is also considered.

The simple prediction model is named as *Deep Platform Empirical Prediction Model* (*DPEPM-1*). The model requires minimal input data, is quick to run and learn, and so can be utilised by the non-specialist. The tool could be used for the early assessment of the effects of design changes on speech transmission quality (as indicated by STI) as well as a screening or diagnosis tool when testing VA system performance on platforms.

The objective is to produce, with further development, a model which could eventually remove the need to systematically utilise commercial acoustic simulation programs to design and predict platform VA system performance in platform of similar conditions.

The prediction performance of *DPEPM-1* is validated against measurements and tested against other applicable prediction models from the literature.

5.2 Methodology

This section details the process followed to develop a database of a large sample of deep platform acoustic and architectural survey data from which statistical baseline information can be extracted. The procedure for the creation and validation of the empirical simple model based on the above database is also described.

5.2.1 General approach

An extensive database was created from the information provided by a scheme of station platform surveys on several lines of the London Underground network. The surveys provided platform speech transmission quality data (STI), acoustic field parameters (RT20), geometrical and architectural details of each platform.

This raw data was screened following suitability and simplification criteria and then compiled into a database on which analytical studies were performed.
The development of a reliable empirical prediction model was initially considered feasible due to the large sample of platforms surveyed and their apparent geometrical similarity.
Since acoustic parameters during station surveys were measured under noiseless condition this will result on the reverberation of the space to be the dominant and main degrading factor on the speech transmission quality predicted by the STI. This leads to consider the STI vs RT as the principal relationship to focus on for the determination of a predictive simple model.

The simple model development aims to predict the STI score of deep platforms equipped with conventional distributed VA systems based on the platform RT in the 500Hz,1kHz and 2kHz octave bands and certainty that the platform fulfils the model applicability conditions.

Parametric and correlation analyses are then performed on an optimised database to identify the influencing geometrical and architectural factors on the STI parameter. Since the target relationship selected is STI as a function of input RT, platform entries possessing significant level of other influencing factors are identified and removed from the database.

This influencing factor identification and platform data entry elimination process is performed successively to arrive at a final database in which the platform data entries only contain RT data as influencing factor to STI. The resultant simple model will be based on a prediction function derived from regression analysis performed on the STI and RT relationship obtained from this final database.

5.2.2 Survey data

As part of a VA system renovation scheme for a large number of London Underground stations, platform surveys were undertaken to characterise each platform acoustically, geometrically and architecturally. For the purposes of this chapter, only station survey data relevant to deep platforms of single track and circular cross section were selected (figures 5:1, 5:2 and 5:3). A total of 86 surveyed platforms fell within the above initial platform type classification. A validity screening process on the initial dataset collected was applied. A total of 12 platform data were rejected as valid entries for the database due to unrepresentative conditions during their acoustic survey, unusual geometrical / architectural features or incomplete or unreliable measured acoustic data.

Hence a total of 74 surveyed platforms produced suitable data to be incorporated into the initial complete database. This number of platform entries represents approximately 50% of the total population of deep platforms in the London underground network.

It was verified that the platform variables (see below) data ranges provided by the 74 platform sample, encompassed most geometrical and architectural features found in deep platform of the London Underground network.

The information surveyed from the platforms and collected in the initial complete database included platform features and characteristics (variables) which were considered likely to have an effect on the STI parameter at receiver positions.

Surveyed platform data entered in the database consisted of three categories: Acoustic measurements, geometrical measurements and architectural details.

For the acoustic surveys the exponential sine sweep open loop technique was utilised to obtain the room impulse response RIR measurements. The application of this technique involved the test signal being reproduced through the existing station VA distributed loudspeaker system and recorded on a portable sound analyser for later post processing (Gomez et al,2008) (see appendix B.1 and H). From the RIR RT20 and STI were derived. All the acoustic measurements were undertaken during engineering hours.

For each platform, RT20 (250 Hz to 4 kHz) and STI values measured at the specified five test positions along the middle line of the platform length were averaged (VA system specifications, 2005) and introduced in the initial complete database. It was observed that deep platforms of single track and circular cross section had rather similar geometrical characteristics, however those platforms can possess significantly different architectural features (variables). It was initially hypothesised that certain degrees of dissimilarity in those characteristics and features significantly influence the acoustic (RT20) and speech intelligibility (STI) parameters as measured on platform positions.

The geometrical data collected during the platform surveys included the following variables: platform length, width, height and curvature observation details. Architectural details consisted of the measurement and count of acoustically open and closed over bridges, count and measurement of cross passages, count and measurement of ventilation outlets as well as the description of platform surfaces materials. Details of acoustically open over bridges, ventilation outlets, cross passages and tunnel openings were taken in the survey as they offer surface areas of high acoustic absorbency; those surface areas are referred in this chapter as “open area”.

5.2.3 Database construction

The platform survey data for the selected 74 platforms was compiled in a large MS Excel 2007 database. This initial complete database incorporated all the relevant platform information and variables arranged in the spread sheet as data field columns.

Data fields included: station name and reference number, RT20 in octave bands, platform length, width, height, count and measurements of cross passages (figures 5:1 and 5:3), count measurement and description of over-bridges (figures 5:1 and 5:2), description of type of ceiling cover (figures 5:1 and 5:2), count and measurement of vents (figure 5:2), curvature description (figure 5:2), loudspeaker spacing, description of surface materials, description of other open areas (figure 5:3) and other relevant surveyor remarks.

Open area absorption contributions were speculated to be a strong influential factor. It was also initially hypothesised that the metallic panels clad on the ceiling featured in some platforms could act as diaphragmatic absorbers providing low and mid frequency sound absorption.

Figure 5:1 Left: platform of large volume with ceiling metal cladding. Right: platform with two blocked over-bridges.

Figure 5:2 Left: Curved platform. Right: platform showing vents outlets and ceiling metal cladding.

Figure 5:3 Left: platform with over bridges of acoustically transparent sides; Right: platform showing two central wide cross-passages.

In a process of database optimisation and simplification for later analysis, some data fields (columns) were combined or removed if that data field appeared invariant throughout the complete database. Invariant data fields removed from database included: cross section aspect ratio, loudspeaker spacing, height and aim, surface materials and over bridge dimensions.

Platform geometrical variables length, width and height were combined into a single data field of calculated volume arranged in a single column and named as *volume*. Likewise cross-passages, tunnel openings and ventilation outlets open surface areas were added together and presented as single data field column named as *openings absorption*. Relevant surveyor's notations and description comments on the visual inspection of architectural details were incorporated by estimating an approximate numerical value in the relevant data field. This was the case for end wall equivalent open area, metallic ceiling coverage area and over bridges side wall estimated open area. Platform curvature was considered acoustically significant if any receiver position did not have direct line of sight from any of the loudspeakers on the platform. This curvature criterion was input as *yes* or *no* in the corresponding database field.

All the resultant data fields were arranged in columns and are referred hereafter as variables or factors. The resultant database as this stage was named the simplified database (table 5:1).

	RT20(sec)					RT20aver		Major potential factors					
Platfor	250Hz	500Hz	1KHz	2KHz	4KHz	Average 500/1K/2K H	STI	Volume (m^3)	Overbridg	Overbridge Closed	Ceiling Clad coverage	Platform Straigl	Openings Absorption (m
PLT1	3.5	4	3.3	2.6	1.7	3.30	0.43	4422.6	NO	-	HALF	YES	25
PLT2	2.2	2.4	2.1	1.8	1.1	2.10	0.44	3046.7	NO	-	HALF	YES	37
PLT3	3.1	3	2.6	2.3	1.6	2.63	0.44	2812.3	2	NO	-	YES	73
PLT4	3.6	3.1	2.6	2.3	1.6	2.67	0.430	2812.3	NO	-	-	YES	40
PLT5	3.64	3.3	2.8	2.5	1.7	2.87	0.420	2812.3	NO	-	-	YES	26.5
PLT6	3.1	2.6	2.2	2.2	1.9	2.33	0.44	2812.3	2	YES	-	YES	26.5
PLT7	3.5	3.5	2.9	2.5	1.7	2.97	0.390	2812.3	NO	-	-	YES	25.96
PLT8	3.7	3.3	2.8	2.5	1.6	2.87	0.43	2812.3	2	NO	-	YES	41
PLT9	3.04	2.79	2.40	2.16	1.29	2.45	0.450	2968.6	NO	-	-	YES	31.48
PLT10	3.5	3	2.65	2.35	1.6	2.67	0.440	2968.6	NO	-	-	YES	31.48
PLT11	3.6	3.5	3	2.6	2.1	3.03	0.420	2812.3	NO	-	-	YES	30.46
PLT12	3.4	3.5	3.1	2.7	2.2	3.10	0.420	2812.3	NO	-	-	YES	30.46
PLT13	2.89	2.88	2.53	2.23	1.58	2.55	0.430	2812.3	NO	-	-	YES	24.5
PLT14	3.92	3.77	3.08	2.59	1.78	3.19	0.4	2812.3	3	NO	-	YES	39.5
PLT15	3.88	3.77	3.07	2.73	1.76	3.15	0.390	2812.3	NO	-	-	YES	24.5
PLT16	3.61	3.37	2.86	2.53	1.74	2.92	0.38	2812.3	NO		-	YES	24.5
PLT17	2.87	2.86	2.68	2.32	2.08	2.62	0.41	2968.6	2	YES	-	YES	24.5
PLT18	3.00	2.84	2.97	2.51	2.20	2.77	0.430	2968.6	NO	-	-	YES	24.5
PLT19	1.29	1.68	1.89	1.79	1.73	1.78	0.5	2812.3	NO	-	HALF	CURVED	27.4
PLT20	1.49	1.86	2.24	2.15	2.04	2.08	0.46	2812.3	NO	-	HALF	YES	33.9
PLT21	2.00	2.42	2.75	2.34	2.03	2.50	0.45	2756.5	NO	-	FULL	YES	29
PLT22	1.71	2.12	2.29	2.07	2.00	2.16	0.46	3133.5	NO	-	FULL	YES	36.1

Table 5:1 Extract of 22 platform entries from the simplified deep platform database showing filtration buttons in each data field.

5.2.4 Database analysis

The simplified database was analysed to produce baseline information for acoustical (RT20) and speech transmission quality (STI) characteristics on platforms

For the analytical purposes JND of RT20 and STI parameters are taken as maximum threshold reference value, 10% and 0.05, respectively, see chapter 6.

Measured octave band RT20 data is shown in different data fields as useful analytical information. A single number, $RT20_{aver}$, calculated from the average values of RT20 at 500Hz,1kHz and 2kHz bands is presented in a separate data field column as a representative and convenient single figure of the overall reverberation conditions.

To ensure that the correlation relationship between RT_{aver} and STI was independent of other factors (i.e. geometrical and/or architectural), parametric analysis on the simplified database was necessary to identify platform data entries containing influencing factors on the STI that could contaminate the relationship of interest.

A total of six geometrical and architectural factors were analysed sequentially by using filtered databases which isolated a factor and STI data to assess its influence. A factor was considered to be of influencing significance if changes on the STI due to factor variation were higher than 1 JND or if the correlation coefficient between the two was higher than R=0.75. Platform entries possessing influential factors on STI other than RT_{aver} parameter were eliminated from the simplified database.

To provide additional validity on the analysis results, influential factors identified in this chapter were contrasted with the findings of similar analyses based on tests using commercial simulation programs in chapter 6 and 7.

The simplified database included the following factors: RT20 in octave bands, $RT20_{aver}$, Volume, Number of over bridges, Over-bride closeness condition, Ceiling cladding coverage, Platform straightness and Opening absorption (area). As above, these factors were arranged as data fields columns for multilevel filtration (table 5:1 and appendix D.1).

Through the process of influencing factor elimination, a pre-final database was determined presenting RT_{aver} values with their corresponding STI scores independent of any other factor or parameter.

A preliminary prediction function was then derived from a regression based function relating the $RT20_{aver}$ and STI datasets. Data entries in the correlation analysis which fell outside a validity band of ±1 JND STI about the regression function were rejected from the pre-final database. After performing this last screening criterion, the final version of the database was established from which the final regression function was derived and formed the basis of the prediction model.

The initial hypotheses considered above in the development process were tested during the database analysis. If a hypothesis was found *not true* it would be invalidated and corrected accordingly at the corresponding process stage (figure 5:4). Figure 5:4 illustrate the approximate model construction structure.

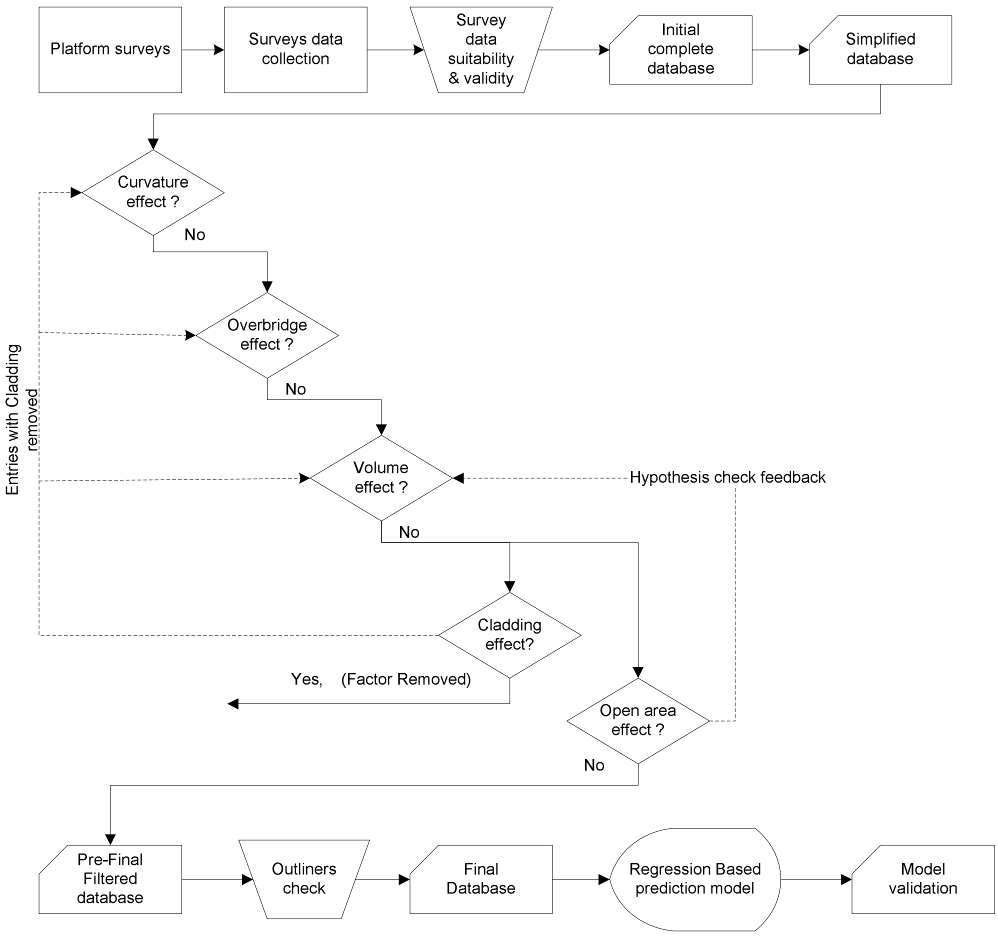

Figure 5:4 Empirical prediction model construction flow chart.

5.2.5 Error analysis and prediction accuracy

In the regression analysis below, Person's correlation coefficient R is used to indicate the interdependence between factors (or variables) and establish the influence of some of them on STI. The RMS error and the overall absolute mean difference are used to express the theoretical prediction accuracy of the final regression based model. The RMS error is calculated from the equation:

$$RMS\ error = \sqrt{\sum \frac{x^2}{n}}$$

Equation 5:1 Root mean square error

Where x is an individual data point error and n is the number or data points.

In section 5.4, model prediction error (error) is defined as the arithmetical difference between predicted and measured values. Reference (ref) STI is the average of scores measured on the platform.
For prediction performance evaluation purposes the maximum allowable prediction error set is 1 JND for STI and RT_{aver}.

5.3 Results analysis

The simplified database (table 5:1) is analysed in this section with the two approaches described in section 5.2, to deliver the two main objectives of this chapter.

5.3.1 Deep platform baseline conditions

In this subsection data from the simplified database is analysed statistically to provide an overview on the baseline acoustic and speech transmission quality conditions found on a sample of London Underground deep platforms. The results of the analyses are presented in the table 5:2 and figures 5:5, 5:6, 5:7 and 5:8.

	RT20averg	*STI*	*Volume (m^3)*	*Overbridges per platform*	*Overbridge Closed*	*Ceiling Cladded*	*Curved Platform*	*Total openings Absorption (m^2)*
MIN	1.78	0.37	2540.2	0	-	-	-	20.9
MAX	3.60	0.54	4422.6	4	-	-	-	133.8
RANGE	1.81	0.17	1882.44	-	-	-	-	112.93
MEDIAN	2.65	0.43	2919.4	-	-	-	-	31.5
MODE	2.67	0.43	2812.3	-	-	-	-	24.5
AVERAGE	2.63	0.43	3037.5	-	-	-	-	37.0
STD	0.40	0.04	347.7	-	-	-	-	18.0
Total in sample	-	-	-	19	12	16	3	-
% of sample	-	-	-	25.7	16.2	21.6	4.1	-

Table 5:2 Statistical summary of platform characteristics for the platform sample of the simplified database (n=74)

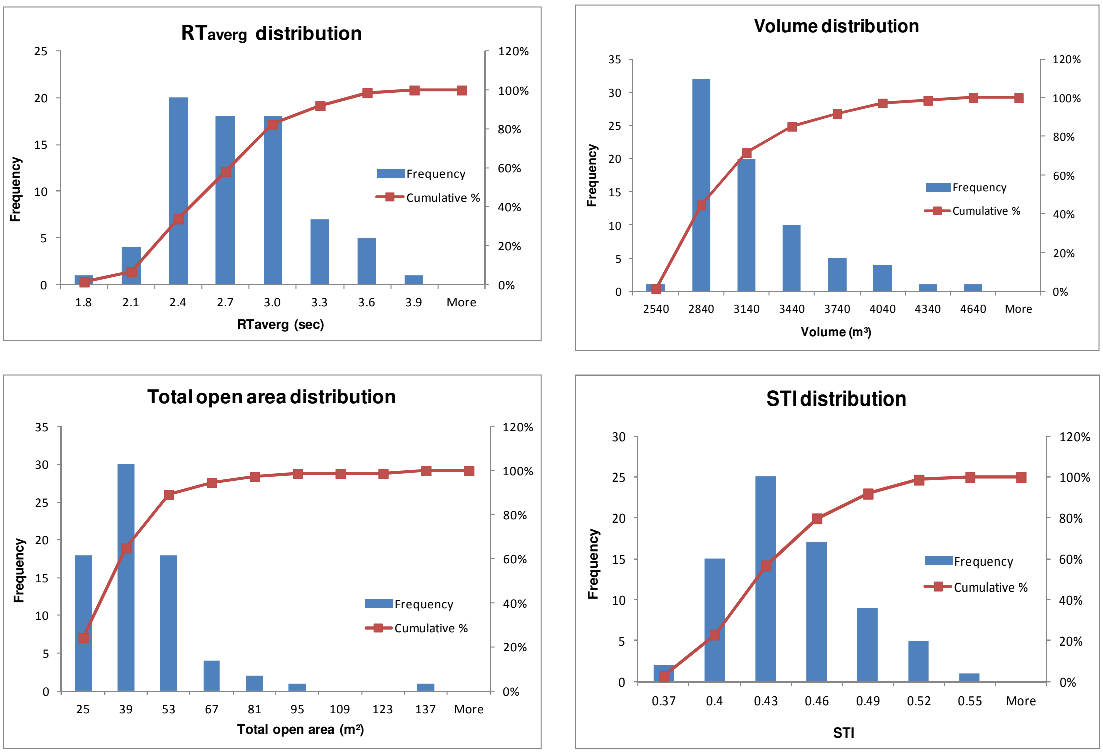

Figure 5:5 Histograms of the platform main variables and STI score

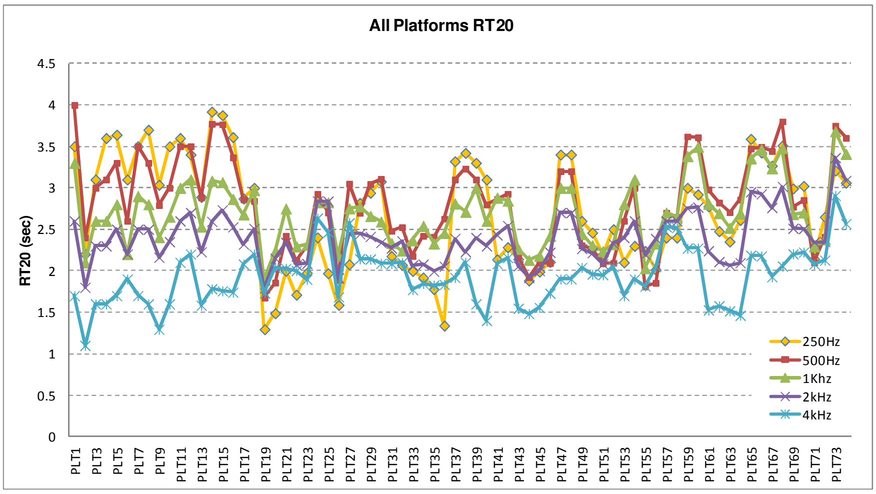

Figure 5:6 All platforms octave band RT20 data variability

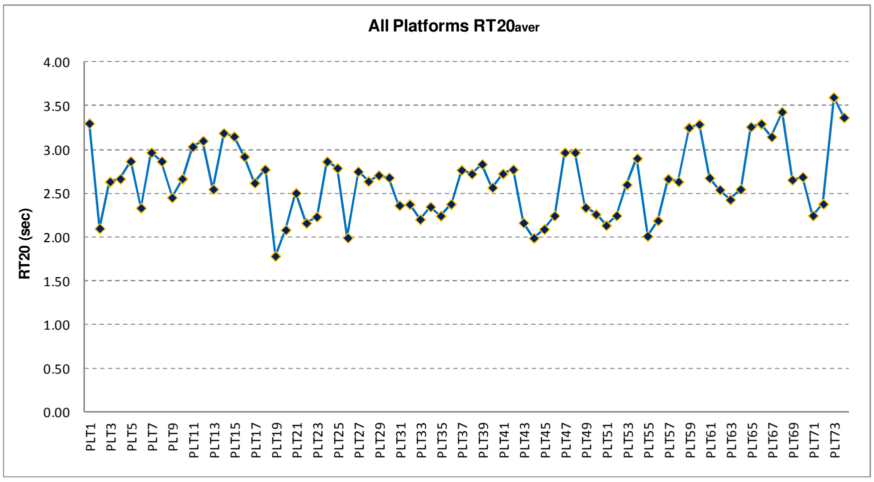

Figure 5:7 All platforms RT20aver data variability

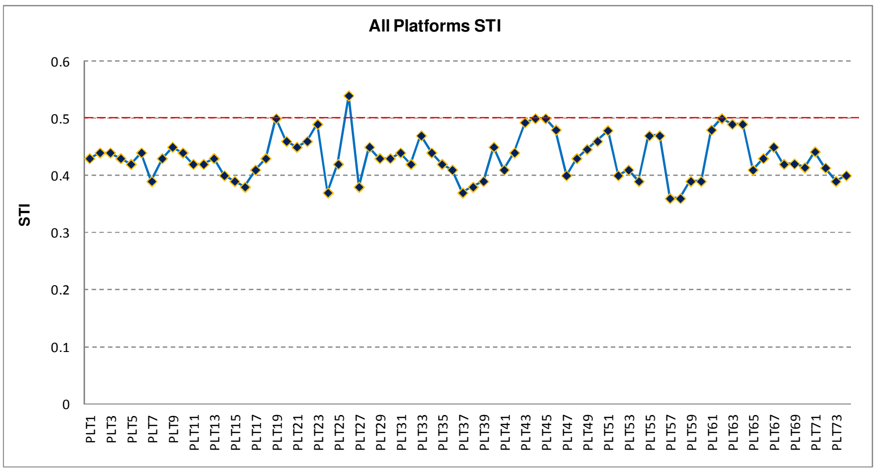

Figure 5:8 All platforms STI data variability and target criteria (red line)

5.3.2 Influential factors determination

The simplified database is analysed parametrically using multilevel data filtration for identifying the influential factors on the STI. The first of the six factors tested was the effect of platform curvature on the measured STI. Values of measured STI on platforms exhibiting pronounced curvature were compared to straight platforms possessing very similar or equal values of all other factors so to avoid contamination from other potential influential factors in the analysis. From the analysis result shown in figure 5:9 it is suggested that platform curvature does not affect the STI.

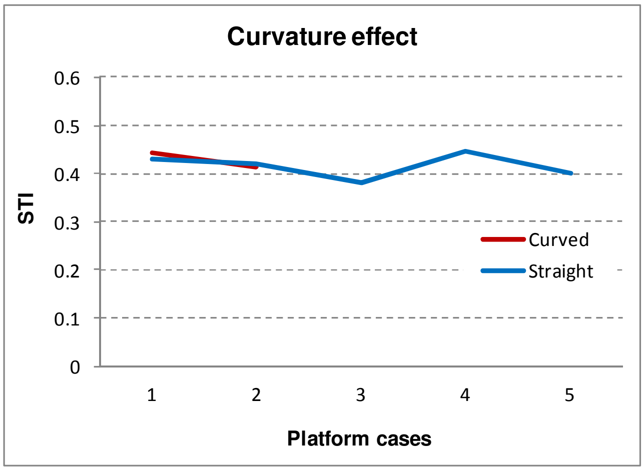

Figure 5:9 Comparative effect on STI values between curved and straight platforms

Thus curved platforms will be kept in the database for analysis of subsequent factors.

The second factor tested in the sequence was the potential effect of over-bridges on the STI. In a similar process as above comparisons were performed on pairs of same station platforms of very similar factors but differing in presence of over bridges.

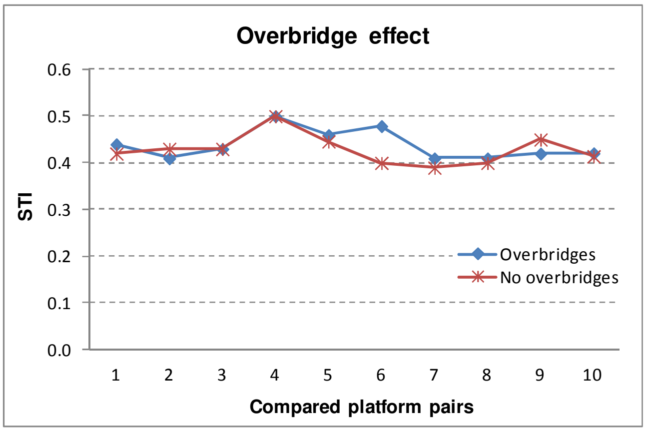

Figure 5:10 STI scores comparisons between pairs of same station platforms only differing in the over-bridges presence.

The analysis results summarised in figure 5:10 shows STI differences between pairs of platforms within 0.03. Pair 6 showed a difference of 0.08 for reasons unlikely to be related to the over-bridges presence, as the other cases demonstrate. Considering these differences in relation to the STI JND, it can be deduced that presence of over-bridges on a platform has no significant effect on the STI. This deduction is consistent with analytical results obtained in chapters 6 and 7. Hence entries featuring over-bridges remain in database for the next factor test.

The next factor tested was the platform volume. The database was initially filtered to reject platforms entries possessing significant values of the remaining suspected influencing factors (opening area and ceiling cladding). Platforms possessing ceiling cladding or opening area falling out of an average range between $21m^2$ and $54m^2$ (table 5:3) (corresponding to 0.8% and 1.9% of the internal total platform surface area) were considered of potentially significant open area and were filtered out.

The dependence between Volume and $RT20_{aver}$ was additionally analysed for a wider understanding of the next Volume and STI relationship. Figure 5:11 shows a scatter plot analysis of platforms' volumes and their corresponding measured $RT20_{aver}$ values. A second order polynomial relationship derived from the plotted data is shown. As it can be observed, there is a weak relationship between the two variables (R=0.33) which suggest that platform $RT20_{aver}$ is not sensitive to significant variations in platform volume.

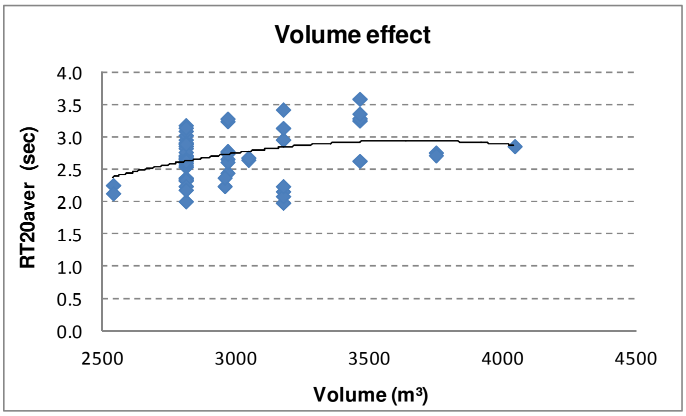

Figure 5:11 Scatter plot of platforms' RT20aver and corresponding volumes. Black line is the polynomial best fit function of the data plotted.

Figure 5:12 shows a scatter plot analysis of platforms' volumes and their corresponding measured STI values. A second order polynomial relationship derived from the plotted data is shown. The relationship between the two variables (R=0.30) was weak which suggest that STI is not sensitive to significant variations in platform volume.

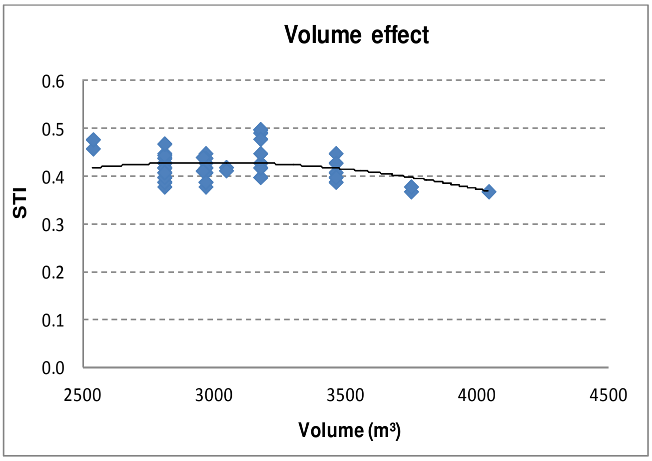

Figure 5:12 Scatter plot of platforms' STI and corresponding volumes. Black line is the polynomial best fit function of the data plotted.

The influence of ceiling metallic clad on STI was evaluated on a specific filtered database which allowed platform entries with factors of similar values to those of entries featuring metal cladding. Figure 5:13 shows a comparison of two groups of nine platforms with and without ceiling metal cladding. It can be observed that platforms featuring metal cladding ceiling have significantly lower RT20 values at low and mid frequencies with respect similar platforms without metallic clad ceiling. The effect of lower RT20 on measured STI can be seen in figure 5:14. Clad platforms measured an average of 0.08 STI (>1.5 JND) higher than a group of similar platforms without the metal cladding. Hence it can be inferred that the presence of metal cladding on a platform is an influencing factor on the STI.

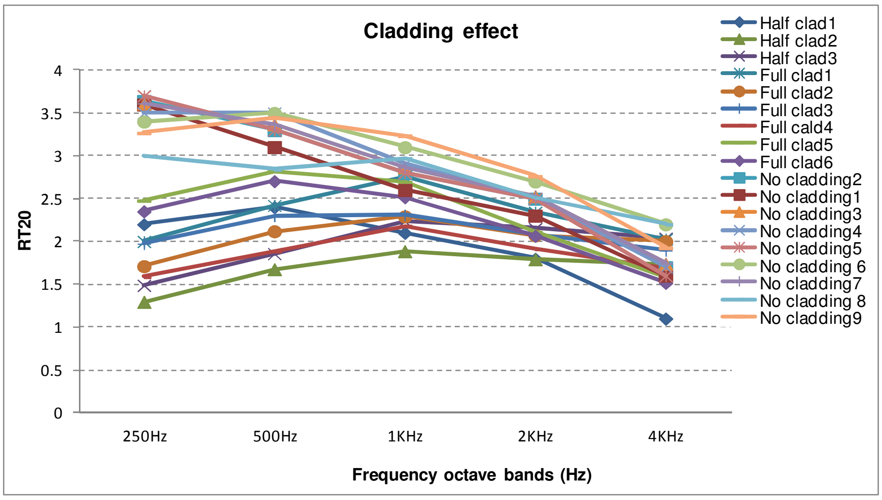

Figure 5:13 RT20 comparison in octave bands between two groups of similar platforms differing in type of ceiling cover

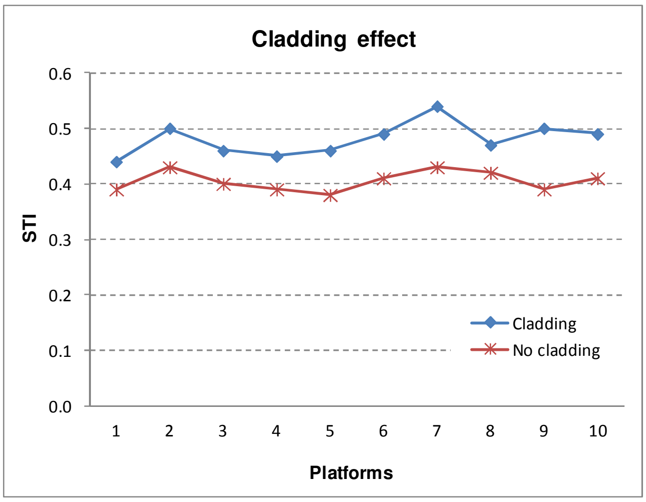

Figure 5:14 STI comparison between two groups of similar platforms differing in type of ceiling cover

For the determination of the effect of opening absorption on STI, platform entries featuring clad ceilings were filtered out from the simplified database.

Figure 5:15 shows a scatter plot relating platforms' *opening absorption* data and their corresponding measured RT20aver values as a complementary analysis. A second order polynomial relationship function reveals a weak relationship between the two variables (R=0.33). This indicates that the measured RTaver on platforms is not influenced by typical open surface areas.

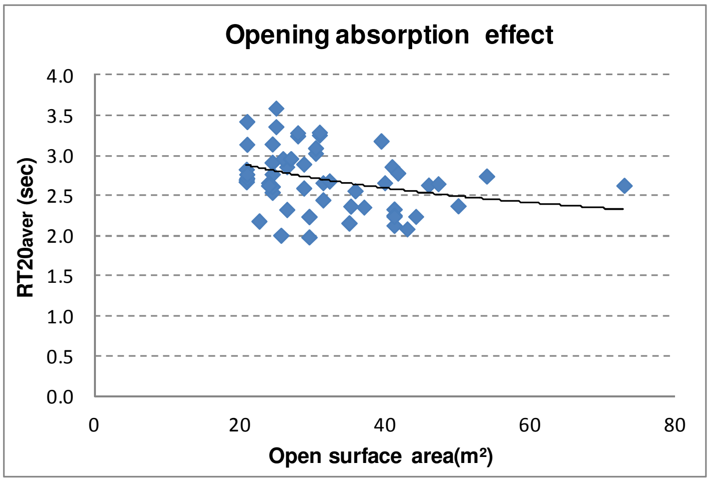

Figure 5:15 Scatter plot of platform RT20aver and opening absorption. Black line is the polynomial best fit function of the data plotted.

Figure 5:16 presents a scatter plot of opening absorption data and their corresponding measured STI values. A second order polynomial relationship derived from the data plotted shows no relationship between the two variables which indicates that measured STI on platforms is not affected by very small amount of open surface areas typical of majority of deep platforms. This finding is line with investigation results performed utilising validated computer models in chapter 7.

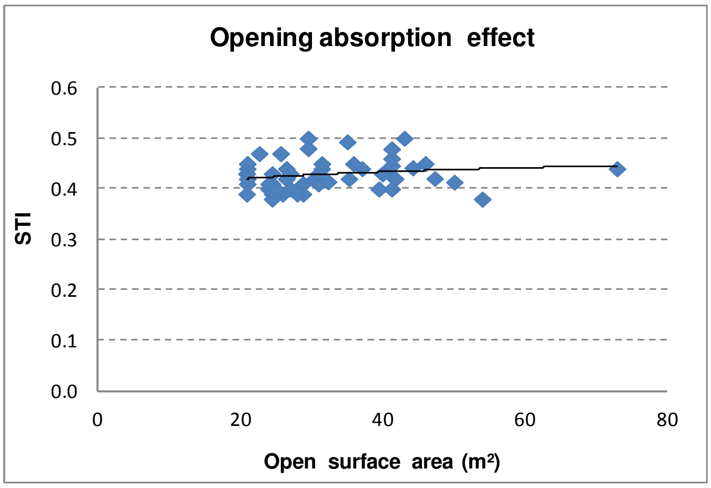

Figure 5:16 Scatter plot of platform STI and opening absorption. Black line is the polynomial best fit function of the data plotted.

5.3.3 Prediction model determination

As a result of the influential factor identification and corresponding data exclusion process, only platforms which featured metal cladding on the ceiling were filtered out from the simplified database. This gave rise to the pre-final filtered database (table 5:3)

			Major potential factors				
Platfor	RT20aver	STI	Volume (m	Overbridg	Overbridge Closed	Platform Straigh	Openings Absorption (m²)
PLT3	2.63	0.44	2812.3	2	NO	YES	73
PLT4	2.67	0.43	2812.3	NO	-	YES	40
PLT5	2.87	0.42	2812.3	NO	-	YES	26.5
PLT6	2.33	0.44	2812.3	2	YES	YES	26.5
PLT7	2.97	0.39	2812.3	NO	-	YES	25.96
PLT8	2.87	0.43	2812.3	2	NO	YES	41
PLT9	2.45	0.45	2968.6	NO	-	YES	31.48
PLT10	2.67	0.44	2968.6	NO	-	YES	31.48
PLT11	3.03	0.42	2812.3	NO	-	YES	30.46
PLT12	3.10	0.42	2812.3	NO	-	YES	30.46
PLT13	2.55	0.43	2812.3	NO	-	YES	24.5
PLT14	3.19	0.4	2812.3	3	NO	YES	39.5
PLT15	3.15	0.39	2812.3	NO	-	YES	24.5
PLT16	2.92	0.38	2812.3	NO		YES	24.5
PLT17	2.62	0.41	2968.6	2	YES	YES	24.5
PLT18	2.77	0.43	2968.6	NO	-	YES	24.5
PLT24	2.86	0.37	4044.6	NO	-	YES	44.1
PLT25	2.79	0.42	2968.6	NO	-	YES	41.8
PLT27	2.75	0.38	2968.6	NO	-	YES	54.04
PLT28	2.64	0.45	3463.3	NO	-	YES	46.04
PLT29	2.71	0.43	2812.3	2	YES	YES	20.92
PLT30	2.68	0.43	2812.3	NO	-	YES	20.92
PLT31	2.36	0.44	2812.3	3	NO	YES	37.134
PLT32	2.38	0.42	2812.3	2	NO	YES	35.31

Table 5:3 Extract from the final filtered database showing filtration buttons in each data field.

A correlation analysis between STI and RT20aver variables of the pre final database is shown in the scatter plot figure 5:17. A regression polynomial best fit curve was computed and plotted to the data. A tolerance band of ±1JND of STI centred on the best fit curve was also plotted to determine potential outlier data points to be excluded. No outlier data points were identified.

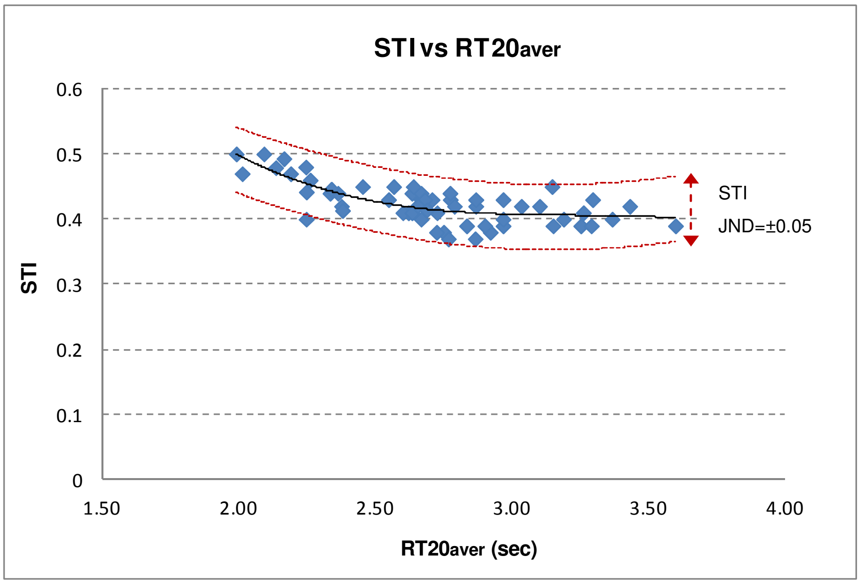

Figure 5:17 STI and RT20aver correlation scatter plot showing a polynomial regression function best fit curve and a 1JND STI tolerance band.

After the screening step for outlier points, the final database and the final prediction function were established. The regression based function fitted to the data analysed (R=0.8) was defined by the following third order polynomial equation:

$$STI = -0.0459 * (RT20_{aver})^3 + 0.4457 * (RT20_{aver})^2 - 1.4461 * (RT20_{aver}) + 1.9742$$

Equation 5:2 Polynomial relationship between STI and RT20aver derived from the final filtered database

Equation 5:2 constituted the basis for the first phase of the deep platform empirical prediction model *DPEPM-1* which can be run on a MS Excel-2007 spread sheet and be presented in a simple to use user interface as shown in table 5:4.

DEEP PLATFORM EMPIRICAL PREDICTION MODEL 1 Please INPUT Reverb Time (sec) in white boxes	
RT @500Hz	3.1
RT @1000Hz	2.72
RT @2000Hz	2.12
Average RT(sec)	***2.65***
STI =	**0.42**

Table 5:4 DPEPM-1 user interface

5.3.4 Applicability conditions and input ranges

Due to the empirical nature of the prediction model and the simplifying approach followed in its construction, several applicability conditions and limitations apply. In the first phase the prediction model is suitable for single track deep platforms of curved ceiling. The model assumes that the platform under prediction is unoccupied (see chapter 4), the ceiling has no metal cladding and the sound is distributed along the platform by a conventional underground platform VA system. Receivers on the platform are assumed to be positioned along the middle line of the platform length at ear height positions. The maximum allowable platform curvature should permit direct line of sight from any loudspeaker position to any receiver position on the listening plane. Table 5:5 summarises the input data operational ranges of the model which cover most deep platforms found in the London underground network.

Platform variable	**Min**	**Max**
Length (m)	108	150
Width (m)	5.6	9
Height (m)	3.7	4.2
Volume (m^3)	2540	4044
$RT20_{averg}$ (sec)	1.9	3.6
Overbridges (units)	0	4
Open surface area m^2	21	73

Table 5:5 Prediction model input variable ranges

5.4 Error analysis, validation and comparative performance

This section presents an error and validation analysis of the *DPEPM-1* model as well as an accuracy comparison of the model against other relevant prediction models and algorithms found in the literature.

5.4.1 Model error analysis

The Person's correlation coefficient R of the dataset analysed from which the prediction model was derived was 0.8. This value indicated a significant correlation between the two variables considered. The overall absolute mean difference and RMS error between measured data and the best fit curve were 0.017 and 0.021 (0.3 and 0.4 JND) respectively. The maximum single difference was 0.05 (1 JND), (figure 5:17)

5.4.2 Model validation and performance comparison

The *DPEPM-1* prediction performance was validated against measurements taken on two suitable London Underground deep platforms (Platform A and St John's Wood), (tables 5:6, 5:7 and appendix D.2). Prediction results from other relevant simple, commercial and theoretical models (see chapter 4) were compared against *DPEPM-1* predictions when the same input was provided (tables 5:6 and 5:7). Table 5:6 provide a summary of results for the STI validation and the STI performance comparison among various models tested using Platform-A measured RT[19] as input data (see appendix D.2).

[19] For the sake of this comparative analysis, reverberation time reported as RT20, $RT20_{aver}$ or RT30 were assumed comparable and were harmonised to a single notation RT.

		Prediction models				
	Measured (ref)	DPEPM1	CATT	CISM	Theoretical prediction	Quasi-diffuse theory + DEPEM1
RT(sec)	2.65	2.65	2.6	2.89	2.65	2.71
STI	0.45	0.42	0.44	0.41	0.38	0.42
error	0	-0.03	-0.01	-0.04	-0.07	-0.03

Table 5:6 Platform-A summary of measured, prediction and prediction error results for several prediction models

Measured RT20aver was not provided as input data to *CATT* (Dalenback, 2007), *CISM* (Dance, S. 2007) and the *Quasi–diffuse theory* (Yang, 1997) (see chapter 8) prediction models, since these programs calculated their octave band RT (see chapter 4).

The *Quasi-diffuse theory* was also utilised to provide calculated RT used as an alternative to measured RT input data value for *DPEPM-1*, creating a combined model *(Quasi diffuse theory + DEPEM-1)* not requiring measured RT input data .

STI results using the "*Theoretical prediction*" (STIr male) were obtained by applying the calculation algorithm contained in BS EN 9921:2003 and BS EN 60268-16:2003.

Another validation and prediction performance comparison was possible by utilising as a reference the measured RT and STI data reported by Barnett (1994b) on St John's Wood underground station platform[20], (see appendix D.2).

STI predictions performed by several models on this platform were reported in the literature and in table 5:7. Table 5:7 provides a STI results summary for the validation and the performance comparison among various models tested using measured St John's Wood platfrom RT as input data.

[20] This is a London Underground deep platform equipped with a distributed VA systems complying the applicability requirements of the *DPEPM-1*

		Prediction models					
	Measured (ref)	DPEPM1	CATT	Yang's	CISM	Harrison	Theoretical prediction
RT (sec)	3.3	3.3	N/A	3.3	3.18	3.3	3.3
STI	0.37	0.40	0.38	0.36	0.45	0.38	0.36
error	0	0.03	0.01	-0.01	0.08	0.01	-0.01

Table 5:7 St John's Wood platform summary of measured, prediction and prediction STI error results for several models

CATT and *CISM* models calculated their own octave RT values from the acoustic simulation of the platform, hence measured RT was not inputted in those models. Prediction results of those two models were reported by Dance (2007) while Yang (1997) and Harrison (2001) reported their prediction results on their respective works.

5.5 Discussion

The results and analyses presented in sections 5.3 and 5.4 are discussed.

5.5.1 Deep platforms baseline conditions

From the statistical analysis of platforms surveyed, it can be observed that most deep platforms present comparable geometrical and acoustic characteristics. These types of platforms tend to generally differ only in few significant architectural features such total open area (opening absorption), number and type of over bridges, curvature and ceiling surface cover type.

The typical volume of deep platforms is concentrated around a value of 3000m^3. This is indicated by the coinciding average, median and mode values (table 5:2 and figure 5:5). The large volume range value (1882.4m^3) can be misleading of the true concentration of values since the range's upper limit is skewed by a large single value of 4422.6m^3 corresponding to a rare case of a very wide platform.

Similarly the total opening absorption of most platforms appears very concentrated around the median, mode and average values while a large single value (133.85m^2) skewed the standard deviation and the range.

In the case of the acoustical characteristics represented by RT_{aver} it can be seen in figure 5:5 that 85.1% of the values fell in a 0.9 second narrow band between 2.4 and 3.3 seconds and that a relatively high concentration of the RT_{aver} was around an average value of 2.63 seconds. In figures 5:6 and 5:7, the RT20 in octaves bands followed closely the curve shape of the RT_{aver} even at octave bands not included in the RT_{aver} calculation. This provides confidence in the RT_{aver} as a single number parameter representative of a wider frequency range reverberation time.

The general high statistical concentration of RT_{aver} and architectural characteristics can be linked to the also low spread in measured STI scores. These appeared centred around the median, mode and average value of 0.43 and showing a standard deviation of 0.04 (table 5:2 and figures 5:5 to 5:8).

The statistical analysis of the platform sample surveyed showed that 93.2% of deep platforms exhibited a $RT_{aver} \geq$ 2.4sec. Based on general room design guidelines for speech communications (Smith et al,1996), (Shield & Hopkins, 2003), (BS 6259:1997), (BS 8233:1999), (Everest, 2000), (Ballou, 2005), it can be stated that reverberation times on London Underground deep platforms appear clearly inadequate for speech reproduction with only 7% of the platforms surveyed which achieved the minimum performance target of 0.5 for speech intelligibility predictor STI (figure 5:5 and 5:8). From the above baseline data and analysis it appears that speech transmission quality in deep platform is capped by the combined geometrical and acoustic conditions found in those spaces.

5.5.2 Influential factors determination

It was speculated that the pronounced platform curvature could affect the STI parameter; the curvature potentially blocking the direct sound and early reflections at some receiver positions from distant loudspeakers (figure 5:2). However the data analysis has shown that STI is not sensitive to platform curvature. This is due to the low number of loudspeakers blocked by the curvature and the prevalence in the sound field formed at any receiver of only near and medium range loudspeakers (see chapter 8). The same explanation applies for the observed insensitivity of STI to the presence of acoustically closed over-bridges acting in this case as large obstacles blocking the direct sound and reflections from distant loudspeakers (figure 5:1). The observed STI insensitivity to large platform obstacles is consistent with the findings of chapter 6 where the presence of over-bridges is investigated using computer simulation and site measurements.

The weak influence that variations of volume had on STI can be directly linked to the weak relationship between platform volume and reverberation (figure 5:11) since in noiseless conditions the STI and the reverberation are strongly interrelated. The reduced RT20 (and STI) sensitivity to the room volume may be explained by the fact that the sound field anywhere in the space is principally determined by the loudspeakers of the distributed system and nearby surfaces thus removing the relevance of the enclosure dimensions. A closer examination of figures 5:11 and 5:12 may suggest that the volume influence on RT_{aver} and on STI (although less noticeable) is capped and reaching a saturation point where larger increases in volume produce comparable influence. Similarly in figure 5:17 the STI appears to plateau which could be interpreted as the point where STI becomes limited (capped) for higher RT_{aver} values (>3sec).

Analysis from RT_{aver} clearly indicated that the metal panels forming the ceiling cladding act as diaphragmatic sound absorbers. Their absorption effectiveness was observed at low and mid frequencies being mainly determined by the mass, dimensions and air gap behind the panels (figure 5:18). This result validated the initial hypothesis formulated in 5.2.3

Figure 5:18 Platform ceiling metal clad showing the panel dimensions and air gap depth.

Although the ceiling clad is not installed for acoustic purposes it proved to provide a significant reverberation reduction resulting in STI increases above 1JND.

The regression analysis between the STI and opening absorption relationship revealed the insensitivity of STI to the typical range of open area found on deep platforms. This finding can be linked to the weak dependence between RT_{aver} and opening absorption (figure 5:15). The link between the two independent analyses, supported the initial assumption that in deep platforms and in noiseless conditions the STI and the reverberation are strongly interrelated. That apparent lack of sensitivity was actually due to the limited amount of opening absorption found on deep platforms unable to produce significant reverberation reduction to effect an appreciable increase on the STI. This conclusion agrees with investigation results found in chapter 7.

5.5.3 DPEPM-1 error analysis, validation and performance comparison

The *DPEPM-1* predicted the STI on both Platform-A and St John's Wood platform with an error less than 1JND (0.03) (table 5:6 and 5:7). That level of prediction error is equal to the STI method measurement error (Wijngaarden, 2006) and is considered as a good simple model prediction accuracy by other researchers (Wijngaarden, 2006).

The *Quasi diffuse theory* calculated RT on Platform-A with error value (0.06sec), well below the RT JND (0.26sec). This minimal calculation error provides the first validation of the Quasi diffuse theory in this research (table 5:6). Further verification of theory is presented in chapter 8.

The *DPEPM-1* under predicted the STI by less than 1JND (0.03) when the RT input value was calculated using the *Quasi diffuse theory* (table 5:6). This shows that the *DPEPM-1* can also predict STI accurately using calculated RT values which can removes the need to provide measured RT input values.

On Platform-A *CATT, CISM* and *DPEPM-1* under predicted STI although their errors were below 1JND. The *Theoretical prediction* also under predicted with error above 1JND (0.07). *Yang, Harrison* and *DPEPM-1* accurately predicted STI scores on St John’s Wood platform with error values under 1JND. *CISM* predicted with an error of 1.5 JND. The *Theoretical prediction* method and *CATT* commercial model showed a prediction error of under 1JND (0.01) (table 5:6).

5.6 Conclusions

An extensive database containing geometrical, architectural and acoustic measured data of a large sample of London Underground deep platforms has been created from data gathered during site surveys.

The database has been statistically analysed to provide a baseline overview on the acoustic and speech transmission quality conditions as well as influencing geometrical and architectural characteristics found on deep platforms.

The study has shown that acoustic and speech transmission quality parameters did not vary greatly across the large sample. Of the geometrical and architectural platform features considered only the presence of ceiling metal cladding showed a significant impact on acoustic and speech transmission quality parameters. The vast majority of platforms surveyed did not achieve the minimum speech transmission quality performance specification for deep platform VA systems.

The database was utilised as the basis for the construction of an empirical model (*DEPEM-1)* for the prediction of speech transmission quality (STI) on deep platform using measured RT as a sole input and under certain applicability conditions. The model was validated against measurements yielding a prediction error below 1 JND. The prediction performance of the *DEPEM-1* was comparable or higher than other simple models or methodologies found in the literature.

In addition, it was found that *DEPEM-1* can conveniently use RT input data calculated by the *Quasi-diffuse theory* as an useful alternative to measured RT data.

The DEPEM-1 has laid the basis for future developments (see chapter 9) to become a quick, reliable and useful prediction tool for the early design stages of VA system on London Underground deep platforms.

Chapter 6 Assessment of acoustic computer simulation programs in underground spaces

This chapter presents a validation and comparative evaluation of two widely used commercial acoustic simulation programs.

6.1 Introduction

Computer processing power now allows the utilization of sophisticated and versatile commercial acoustic simulation programs which are able to yield accurate predictions of indoor sound fields. These highly developed programs have become increasingly employed for the electro-acoustic design and optimization of distributed PA and VA systems in train station or underground spaces.

During the industrial collaboration of the author mentioned in Chapter 1, it was realized there was a need to test the reliability and suitability of commercial acoustic simulation programs when used in the electro-acoustical design of distributed VA systems in underground spaces. Presented for the first time is a quantitative evaluation on the prediction quality and suitability of two industry-standard computer acoustics simulation programs when modelling acoustically complex underground station spaces equipped with distributed VA loudspeaker configurations.

The objectives for this chapter are:

- evaluate the prediction accuracy performance of the two programs
- test their suitability as tools in the design of VA systems in underground spaces
- Compare the programs' prediction and usability performance

Several validating and comparative studies have been undertaken in the past for various commercial and non commercial simulation programs. These works typically compared programs results with one another and against measurements. The most relevant of these studies have been a series of comparative exercises known as Round Robins. These studies consisted of a precision comparison of various leading acoustic simulation packages when used by a group of independent users tasked to model and provide predictions of a given room under the same conditions provided the same input data. The results were used to determine the accuracy of models according to their conceptual nature and the acoustical characteristics simulated in different real case scenarios. In these assessments, programs and user names were kept confidential and spaces modeled were of regular shape and of a low geometrical complexity (Vorländer,1995), (Bork, 2000, 2002, 2005a, 20005b), (Christensen et al, 2008b). Other comparative works were performed independent of the Round Robin studies by specific researchers (Field & Shimada, 2005), (San Martin & Arana, 2006), (Bradley & Wang, 2007), (Astolfi et al, 2008), (Hodgson et al, 2008),(Nagy et al, 2010).

All those studies measured, predicted and compared acoustical parameters values provided by commercial computer models simulating spaces utilizing a single sound source in regular and proportional geometries such in classrooms (Astolfi et al, 2008), (Hodgson et al, 2008), concert halls (San Martin & Arana, 2006), (Shiokawa, & Rindel, 2007), (Nagy et al, 2010) and churches (Field & Shimada,2005), (Segura et al, 2011). Only some of those works provide comparative prediction quality in terms of speech intelligibility metrics such as STI.

However no validating or comparative studies have been found to date on commercial computer simulation programs predicting acoustical and speech intelligibility related parameters from multisource system in complex geometries. The evaluation presented in this chapter attempts to fill this knowledge gap.

6.2 Validation and comparison methodology

This section introduces the methodology followed to validate and compare the two acoustic simulation programs under evaluation. The criteria established to assess prediction validity and accuracy are also presented. Figure 6:3 shows the main steps followed in the methodology.

6.2.1 Programs selection

The two computer acoustic simulation programs selected to undertake the objectives of this chapter were CATT-Acoustics version 8.0g (full capabilities, build 1.01) released in Sept 2007 (Dalenback, 2007) and Odeon (Industrial module capabilities) version v9.2 released in Jan 2008 (Christensen, 2008a). For a fair comparison between programs it was ensured that both programs' versions matched closely in level of development and offered the same level of main prediction capabilities (see chapter 2).

The two programs selected above were chosen among few other competitors of the same caliber on basis of their long standing commercial availability, overall reliability and wide spread use in academic research and industrial applications. The quality reputation that these two leading programs enjoy is largely due to over fifteen years of continuous research and development. Their prediction quality and reliability has been independently validated as reported in the literature (Vorländer,1995),(Bork, 2000,2002,2005a,20005b), (Field & Shimada,2005), (Wijngaarden & Verhave, 2006), (Christensen et al, 2008b),(Astolfi et al,2008),(Hodgson et al, 2008). Only two works have been found on the acoustic simulation of geometrically extreme spaces by commercial computer simulation programs. Sü & Çalışkan (2007) described the utilisation of Odeon program as a design tool specifically for the application of acoustic treatment on underground platforms, however the reliability of the models and calculations was not validated. Wijngaarden & Verhave (2006) briefly reported good prediction accuracy (0.05 of STI) for the Odeon program in long traffic tunnels.

6.2.2 Validation and comparative initial considerations

For evaluation purposes, prediction quality will be quantified and expressed as prediction error. However, the overall quality of computer acoustic simulation programs does not involve solely that measure; some other important aspects are present during the simulation process such as usability, range of output data, data presentation, calculation time, user interface and calculation flexibility. Hence these aspects, some of them qualitative, are subjectively compared and commented upon.

The validation and comparison assessment was undertaken on a group of five representative and different London Underground station spaces (case studies). For the performance comparison between programs, a like for like approach was taken in which the same input data and same operational settings were provided to both programs for each of the five simulations. Acoustic parameter predictions were compared with each other and against experimental values.

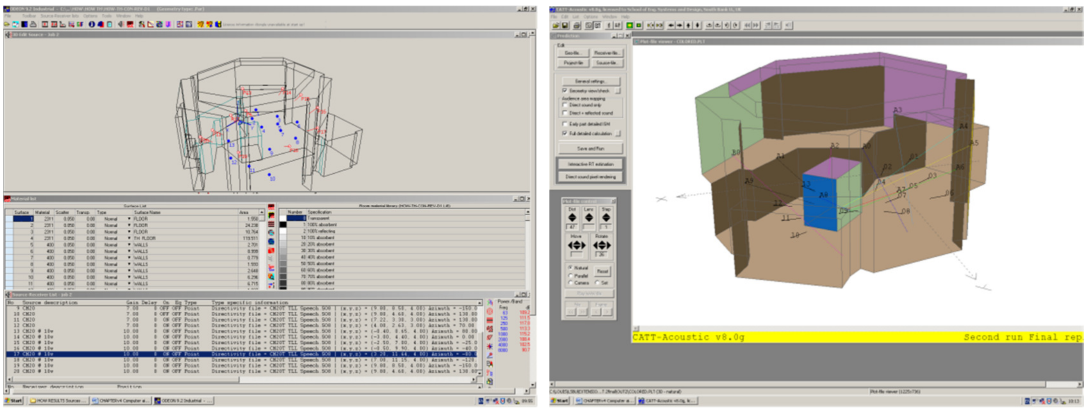

Figure 6:1 View of Odeon and CATT program interfaces showing TH1 simulation

6.2.3 Measurement data

Measured acoustic and intelligibility related parameters of the spaces under study (hereafter referred as evaluating parameters) were obtained from RIR measurements taken during engineering hour surveys prior conducting the acoustic simulations (see chapter 5). Geometrical and architectural details of each space were taken from site surveys.

For the platform measurements, five point receiver positions were taken at specific positions along the centre line of the platform length as indicated in the relevant performance specifications (VA system specifications, 2005). Receivers were positioned in between adjacent loudspeakers (off axis) at 1.5m above the standing platform floor level (listener plane) (figure 6:2).

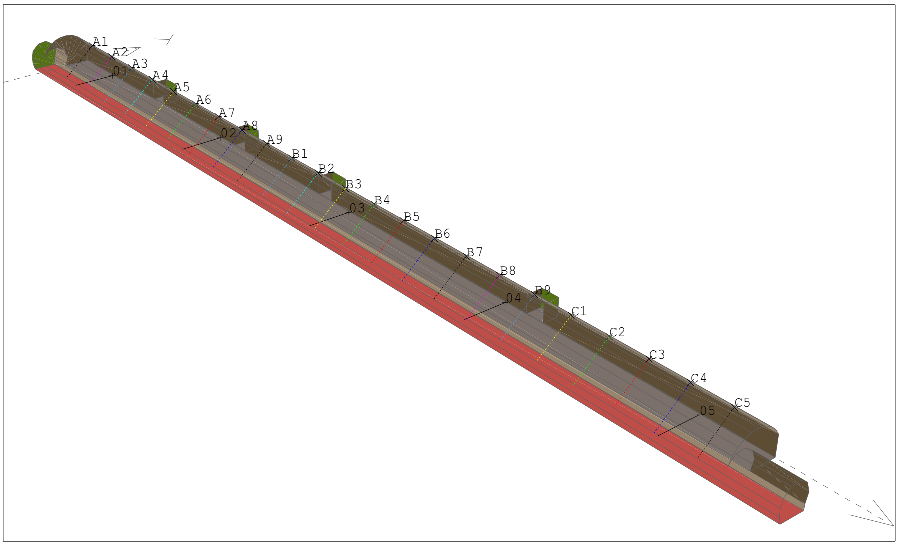

Figure 6:2 Platform five receiver arrangement showing standing platform area (grey surface)

For ticket halls measurements the above document does not specify the number of receiver points or locations. Based on BS EN ISO 3382-1:2009 recommendations a number of receivers points were positioned across the ticket hall floor area to ensure a sufficient representative coverage of the listening plane. Five and six receiver positions were positioned across TH1 and TH2 floor areas respectively. Predicted results were obtained from the point responses of the omni-directional simulated receivers. Measurement and simulated receiver positions were the same for each space examined.

The high SNR obtained during the acoustic measurement surveys (noiseless condition, (see chapter 3) was represented in the simulations. This condition removed the influence of background noise as an influential variable, making the validating and comparative analysis more effective and reliable.

6.2.4 Test spaces selection

To encompass a reasonable range of acoustic complexity found in unoccupied London underground sub-surface stations, five different extreme underground spaces were selected from a group of fourty one stations surveyed. The intended purpose was to test the simulation programs' algorithms with challenging geometries where they could show clear calculation differences and inaccuracies. The five geometries included three different platforms named Plat-1, Plat-2, Plat-3 and two different types of Ticket Halls (TH1 and TH2).

Table 6:1 shows the main significant architectural characteristics for each space (or case study) as part of the selection criteria. The definition of these criteria is described as follows:

Ground level access refers to stations with have acoustic direct line of sight with the outside open air. Regular shape is for spaces for which the main three dimensions are reasonably proportional[21]. Acoustical complexity include spaces which contain large objects, protruding edges, coupling spaces, different height levels, curved surfaces and significant non-spatial uniformity of surface material properties. A platform is considered curved if direct line of sight to any sound source cannot be obtained from any point on the listener plane.

Ticket Halls	***Volume***	***Ground level access***	***Shape regularity / acoustic complexity level***
TH 1	Small (<1200m³)	Yes	Regular / Low
TH 2	Large(>4000m³)	No	Irregular / High

Platform	***Volume***	***Large Obstructions***	***Curvature***
Plat- 1	Typical (~3000m³)	No	No (Straight)
Plat- 2	Typical (~3000m³)	Over-bridges (3)	No (Straight)
Plat- 3	Typical (~3000m³)	No	Curved

Table 6:1 Summary of main characteristics of the spaces investigated

[21] The condition for proportionality is defined here when none of the main dimensions (length, width and height) is several times (i.e up to 5 times) the magnitude of any other.

6.2.5 Geometrical model construction

For each of the five spaces, a three dimensional (3D) geometrical model was created in Google Sketch-Up Pro CAD drafting software (Google, 2010) based on two dimensional architectural plans, on-site height surveys and visual inspections.

Once the geometrical models were validated and equally optimised for subsequent acoustic modelling (e.g air tightness, no warped or overlapping planes) they were exported from the drafting software to CATT via SU2CATT format converter (Rahe-kraft, 2006) and directly to Odeon as dxf format files. The five geometrical models constituted the common geometric input data for the two acoustic simulation programs under examination. All simulations were run in turn on the same dedicated computer under the same computer operative conditions and settings. The dedicated computer was a portable PC computer *HP Compaq nc6320*; equipped with an *Intel Centrino Duo processor* of 2GHz of processing speed equipped with 2GB of RAM running on Windows XP Pro SP3.

6.2.6 Simulation construction and calibration

After the successful import of the geometrical model into the two acoustic simulation programs, all the calculation settings and computer processing conditions influencing the simulation and calculation performance for both programs were set under equal terms. Similarly and for each space simulation, the two programs were provided with the same input data which included environmental conditions, background noise levels, sound source (loudspeaker) type, locations of sources and receivers as well as source characteristics including directivity, aim and sensitivity.

Preliminary material acoustic properties (absorption coefficient and scattering coefficient) were assigned to the model surfaces based on published material libraries (Everest, 2000), (Astolfi et al, 2008), (Hodgson et al,2008), (Yang & Shield, 2001), (Segura et al, 2011) and each program material database and recommendations (Dalenback, 2007), (Christensen, 2008a). In a progressive acoustic model calibration process, the initial surface material properties were gradually adjusted separately for each program so the predicted RT30 values matched as close as possible to the RT30 measured values. The calibration target criteria was that predicted values approximate simultaneously as much as possible measured values at each of the five receiver points on the listening plane and at all frequency bands. Ideal matching target was defined to be within 10% for low frequency bands and 5% for mid and high frequencies. The adjusted absorption and scattering coefficient values were restricted to realistic ranges given in the recommended and published data (see above).

This calibration process is widely used in the industry concerned, research and academia (Field & Shimada, 2005), (Bradley & Wang, 2007), (Nestoras & Dance, 2008), (Christensen et al, 2008b), (Astolfi et al, 2008),(Hodgson et al, 2008), (Segura et al, 2011) as an accurate and efficient method to estimate absorption and scattering coefficients best suited to represent the acoustic field and therefore contribute to a realist acoustic simulation of a space.

Most user definable calculation parameters and calculation settings (i.e. reflection order, transition order and number of rays) were set as automatic as recommended by both programs upon detection of the overall geometry of the model. Only the truncation time in CATT and the impulse response length in Odeon (both settings refer to the same time) were set manually to the same value for both programs in each space simulation. Other more specific user calculation parameters in Odeon were set to engineering mode. This mode automatically determines optimum specific calculation settings according to the geometry of model to yield engineering accuracy predictions while offering reasonable processing times. The oblique Lambert scattering method was selected in Odeon to approximate the different CATT's method of handling scattering which is also based on Lambert's distribution

For each space simulation, the same loudspeaker arrangement and configuration (loudspeaker type, quantity, position/spacing, gain and aim) was input in both programs which replicated the configuration found during the corresponding site measurements. The characteristics of the loudspeakers utilised in each simulation corresponded to the loudspeakers type which reproduced the test signals during the acoustic surveys. A short column loudspeaker type named as Next2 CM20 (see appendix F.2) was used in TH1, PLAT-1, PLAT-2 and PLAT-3 while for the TH2 simulation a 5" recessed ceiling loudspeaker named as Next2 MC5 (see appendix F.3) was employed throughout the ceiling of the space.

6.2.7 Evaluating parameters

The following acoustic parameters were used as indicators for the prediction validation analysis and accuracy performance comparison: RT30 (RT)[22], EDT, total SPL, and STI. The rationale for their selection is as follows:

RT serves as an accuracy indicator of the surface properties input data and of the calculation of the late part of the decay process of the impulse response simulated by each program.

The EDT sensitivity to location and diffuse reflections serves as an indication of the quality of the internal program algorithms calculating the early part of the simulated impulse response.

Total SPL provides an indication of the calculation accuracy of the overall sound field energy levels. Total SPL measurements were not taken during the acoustic surveys, however predictions of total SPL[23] results are presented for comparisons purposes.

STI is the most relevant parameter for the purposes of VA systems in underground spaces. Its complex calculation from the simulated impulse response serves as a complementary indicator of the programs prediction and calculation abilities.

A number of non-acoustic performance indicators were selected as program output metrics to supplement the above acoustic performance indicators in the comparisons and evaluations. These metrics included processing time, calculated volume, calculated total surface, mean free path (MFP), number of rays emitted and number of lost rays.

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[22] Reverberation time measured and predicted in this chapter was RT30 however for convenience hereforth is shortened as RT

[23] In line with the relevant performance specification (VA system specifications, 2005) all the simulations were aimed to reach total SPL of 90dBA when averaged over the receiver positions.

6.2.8 Validation and comparative assessment criteria

For a quantitative and meaningful assessment of the prediction accuracy and performance discrepancy between programs, the just noticeable difference (JND or subjective limen) of each parameter was used as an error reference. Prediction errors and discrepancies were scaled relative to the corresponding parameter JND (see 6.2.9). This evaluation criteria approach is also widely utilised in the literature for similar comparative purposes (Vorländer,1995),(Bork, 2000,2002,2005b), (Astolfi et al,2008), (Hodgson et al,2008), (Christensen et al, 2008b), (Segura et al,2011).

The parameters' JND's were established by the author for the purposes of this research are determined and rationalised as follows.

BS EN ISO 3382-1(2009)[24] specifies the EDT[25] and sound strength (G) JND at 5% and 1dB respectively. However these limens were originally determined for concert halls applications under ideal conditions in which listening acuity and high measurement precision were required. Lundeby et al (1995) noted that measurements of room acoustical parameters in accordance with ISO 3382 led to a measurement uncertainty of the same or greater magnitude than the differences that would be just noticeable subjectively (σ upto10% in EDT and RT).

It is however reasonably considered by this and others authors (Hodgson,1996), (Christensen et al, 2008b), (Bistafa & Bradley,2000), that the limens indicated in BS EN ISO 3382-1:2009 can be too low for engineering and other practical applications. An engineering type level of accuracy of 10% suitable for reverberation time predictions in practical applications was suggested by Hodgson (1996) and Bistafa & Bradley (2000). In agreement with these authors and for the purposes and engineering context of this research, the EDT and RT30[26] JND are established at 10%.

BS EN ISO 3382 (2009) specifies the JND for the G parameter at 1dB. However the JND for SPL could be assumed to be that of G as suggested by several researchers (Gil-Reyes et al, 2011), (Christensen et al, 2008b), (Astolfi et al, 2008).

[24] The determination of ISO 3382-1 JND's was limited to the 1kHz octave band. In the absence of frequency dependant limen determinations those values are widely used for any frequency band.
[25] EDTJND is specified relative to a reference value.
[26] In the absence of stadarised JND for RT30, the EDT JND is widely used.

The author deems that a SPL JND of 1dB to be too stringent for predictions in an engineering context and when considering that 3dB SPL is the minimum change for a perceivable change in loudness in most practical situations. In agreement with other researchers (Astolfi et al, 2008), (Hodgson,1996) the SPL JND is established for the engineering purposes of this work to be 2dB.

In the absence of a formally established JND for the STI parameter, researchers have suggested in the literature limen values ranging from 0.03 to 0.1 of the STI scale (Bradley et al, 1999: 0.03), (Eggenschwiler & Machner, 2005; at 0.03), (Christensen et al, 2008b; at 0.05), (Gil-Reyes et al, 2011; at 0.05), (Hak & Vertegaal, 2007; at 0.1), (Wenmaekers et al, 2009; at 0.1), (Bowden & Wang, 2007; at 0.1).In a computer acoustic simulation comparative Odeon developer, Christensen (2008b), reported typical STI acoustic simulation prediction errors of around 0.1 STI which he considered as reasonable prediction accuracy.STI intelligibility qualification scale is arranged in steps of 0.15 (BS EN ISO :9921,2003) and the STI measurement error is 0.03 (Wijngaarden & Verhave, 2006). In the new standard revision on the STI method (BS EN 60268-16 :2011) the interval between adjacent qualification bands is 0.04. Based on the information gathered from the above literature review and the extensive practical experience gained by the author during the numerous station acoustic surveys, a 0.05 of the STI scale is adopted by the author as the JND for the purposes of this research work.

For the presentation on a single ordinate scale, hereafter one unit of the JND given in table 6:2 will be used as the prediction error and discrepancy threshold reference for the evaluations presented in this chapter.

Evaluating parameter	BS EN ISO 3382-1:2009	Author criteria
EDT	5%	10%
RT30	5%	10%
SPL	1dB	2dB
STI	N/A	0.05
D50	5% (0.05)	5%(0.05)

Table 6:2 Summary of JND as per BS EN ISO 3382-1:2009 and established by the author for various evaluating parameters

6.2.9 Prediction error and discrepancy definitions

The prediction performance assessment for each program was based on evaluating the absolute deviations from measured values for each parameter (prediction error) and for each simulation case study. Taking the above criteria of one unit of JND as a normalizing prediction error reference for each evaluating parameter, the following expressions can be defined to calculate relative prediction error in terms of JND:

$$error_{JND} = \frac{/Prediction - Measured/}{Measured} * 100 * \frac{1}{JND}$$

Equation 6:1 Prediction error definition normalised to 1JND used for reverberation parameters (EDT and RT30)

$$error_{JND} = \frac{/Prediction - Measured/}{JND}$$

Equation 6:2 Prediction error definition normalised to 1JND used for intelligibility related parameter STI and total SPL

The above relative errors are calculated individually for each receiver position and for each frequency octave band (where applicable).

For overall comparative purposes, an overall prediction ($error_{case(n)}$) is calculated for each program and for each of the five simulation cases as an average of relative error over the different parameters, receiver positions and frequency octave bands. The overall for each simulation case ($error_{case(n)}$) is calculated by equation 6:3.

$$error_{case(n)} = \frac{\sum_{n=1}^{P} \sum_{n=1}^{Freq} \sum_{n=1}^{Rec} (error_{JND})}{N_p * N_{freq} * N_{rec}}$$

Equation 6:3 Overall prediction accuracy for a case study

Where N_p is the number of evaluating parameters, N_{freq} is the number of octave frequency bands and N_{rec} is the number of receiver positions.

The total prediction error calculated over the five simulation cases *($error_{total}$)* examined in this chapter is calculated by equation 6:4

$$error_{total} = \frac{\sum_{n=1}^{case}(error_{case})}{n}$$

Equation 6:4 Overall prediction accuracy calculated over the five simulation cases

The relative parameter discrepancy *(%discrepancy)* between program predictions is used for the reverberation parameters (RT30 and EDT) and is calculated with equation 6:5 as the fraction of the absolute difference between two programs' predicted values over one of the values taken as the reference (equation 6:5).

$$\%discrepancy = \frac{/Prediction_a - \ Prediction_{ref}/}{Preciction_{ref}} * 100$$

Equation 6:5 Percentage relative discrepancy between two programs' predicted values

The absolute parameter discrepancy between program predictions is used for SPL and STI parameters and is defined in equation 6:6 as the absolute difference between two programs' predicted values taking one as the reference.

$$Abs\ discrepancy = / \ Prediction\ _a - \ Prediction_{ref}\ /$$

Equation 6:6 Absolute parameter discrepancy between two programs predictions

Predictions values produced by CATT were taken as the reference (*$Prediction_{ref}$*) for the equations 6:5 and 6: 6.

It is important to remember that prediction error is expected to be higher towards low frequency octave bands due to the natural higher measurement uncertainty and intrinsic accuracy limitations of geometrical acoustics in that frequency region (Bork, 2005b), (Shiokawa & Rindel, 2007). However this frequency dependency was not accounted for in determination of the accuracy reference criteria (1JND) where all frequency octave bands are assigned the same JND value (see 6.2.8).

6.2.10 Programs validation and comparison methodology flow chart

Figure 6:3 provides a generic representation of the process followed for the validation and performance comparison of the two commercial acoustic simulation programs under examination.

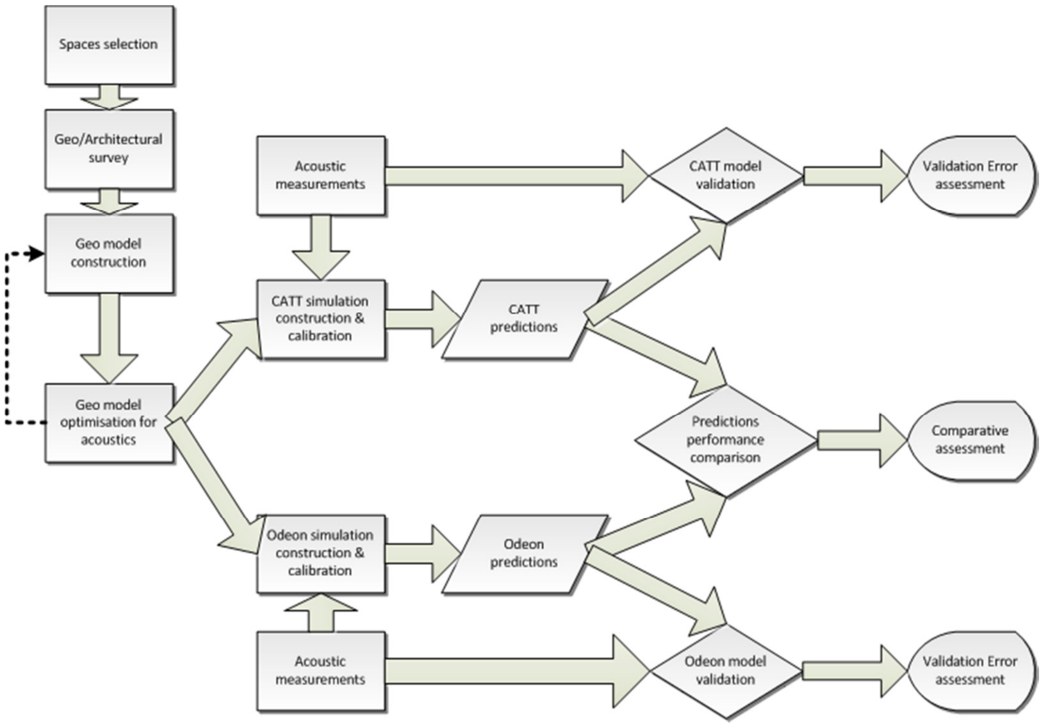


Figure 6:3 Validation and comparative assessment generic procedure flow chart

6.3 Results, analysis and discussion

6.3.1 Introduction

This section provides the prediction results by the two programs performing simulations in each of the five spaces as well as the corresponding measured values in the actual spaces. Each of the five simulations is examined in a separate subsection as an individual case study presenting results, analysis and discussion. This is followed by a cross-case examination and an overall evaluation.

Measured and prediction results are presented in graphical and tabulated (in appendix) form for each evaluating parameter at each receiver positions and for all frequency octave bands of interest (125Hz to 8kHz).This allows for spatial and frequency variations as well as trends in magnitudes to be investigated.

Standard deviation (σ) of averaged measurements is shown as error bars in data graphs as an indication of measurement uncertainty. Relative prediction errors averaged over the test positions are presented for RT30, EDT and STI parameters in column graphs incorporating one standard deviation (±1σ) error bars as an indication of the averages uniformity. Measured STI at each receiver is presented with vertical black line showing ± 1JND range.

A summary graph presents EDT, RT and SPL relative prediction discrepancy between the two programs averaged over the five receiver positions together with ±1σ error bars. STI and SPL discrepancy is provided in tabular form. EDT/RT ratios are provided in the appendix as an additional metric. Detailed data, results and analysis tables for each space can be found in the appendix E.

Additional receiver points were predicted in all the case studies as support data for the analysis and can be seen in the simulation views. However for clarity in the presentation of results this redundant data is not reported.

At certain receiver positions or whole station areas, measurements at low frequencies could not be reliably derived from the measured impulse responses due to limitations of the existing VA system or unexpected low frequency noise prevalent in the area at the time of measurement. Hence reverberation octave data is not presented in the corresponding result analyses at 125Hz band for Plat- 2 and Plat- 3 as well as 125Hz and 250Hz bands for TH2. Similar low frequencies measurement limitations have been reported and commented in other studies (Bork, 2005b), (Christensen et al, 2008b), (Dalenback,2007).

A summary table is provided for each simulation case detailing each program's calculation inputs settings and output parameters for program performance comparison purposes.

6.3.2 Ticket Hall 1

6.3.2.1 Room Description

Ticket Hall 1 (TH1) is a hall of approximately 1100m^3 of volume and 2590m^2 of internal surface area featuring a rather regular heptagon shape with the following main dimensions 14.6m (L) x 11.5m (W) x 7.9m (H). Some of the hall walls include large protruding volume irregularities into the room which could promote the creation of a more diffuse acoustic field, (figure 6:4 and 6:5). All internal surface materials are similarly large and acoustically reflective as is typical for underground stations. The hall does not present other fittings or architectural details which could be acoustically significant in the simulation except for the presence of parallelepiped ticket office kiosk of 14.5m^3 situated in the middle of the floor area acting as a large acoustic obstacle. The hall contains two open entrances at street level which have been modeled as very absorptive surface areas with very low scattering coefficients.

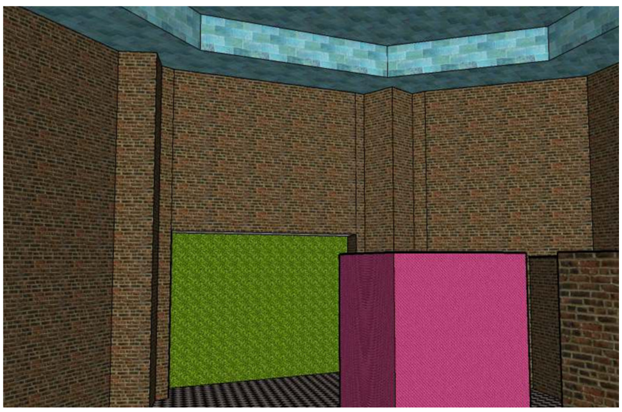


Figure 6:4 TH1 Sketch Up Pro internal view of geometrical model showing the ticket office kiosk (in pink) and a street level entrances (in green).

Overall this space can be considered of low acoustical complexity and representative of a typical medium size and regular shape ticket hall found in the London underground network.

Figure 6.5 shows the simulation of the VA loudspeaker arrangement (sound sources) found at the hall which incorporated 11 column loudspeakers equally spaced at approx 5m intervals fixed on the hall walls at 4m above the floor level. Thirteen simulated receivers were distributed across the hall floor area at a height of 1.5m. Five of the simulated receivers were placed at the receiver locations used for the acoustic measurements.

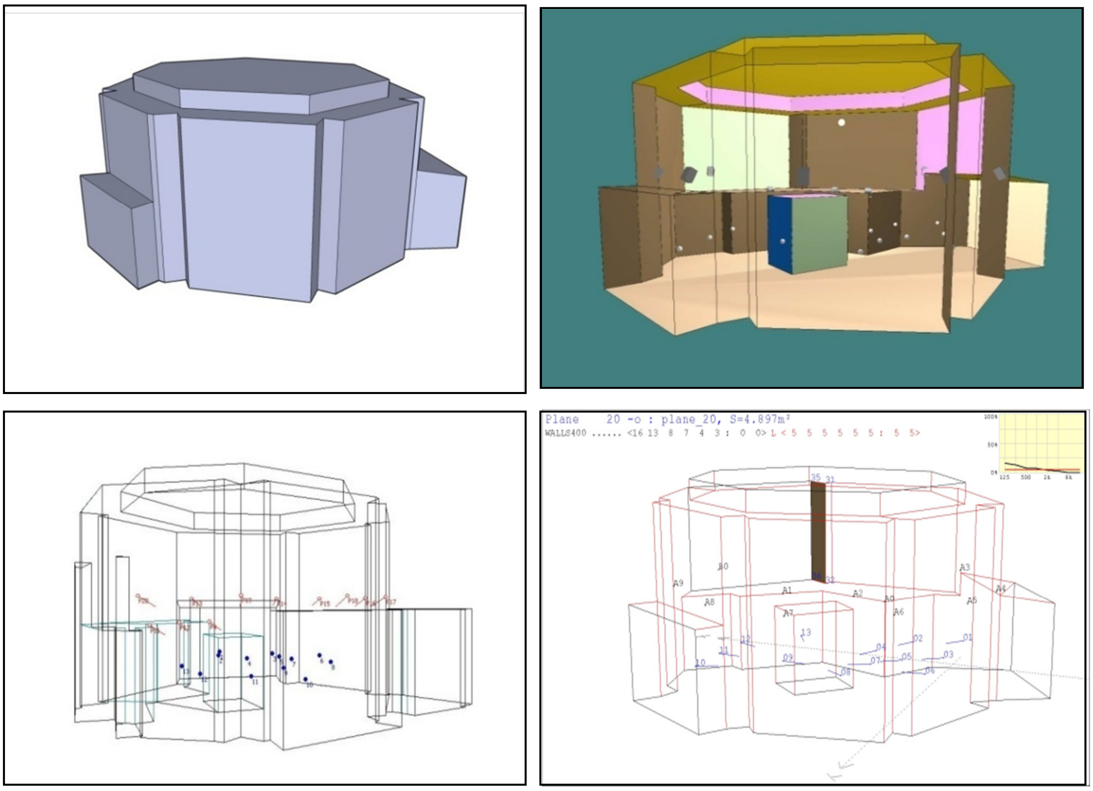


Figure 6:5 Top left to bottom right figures from Sketch Up, CATT 3D viewer, Odeon and CATT wire frame view showing sources and receivers

6.3.2.2 Results

Input Settings	**Odeon**	**CATT**
Calculation algorithm	Hybrid	Hybrid
Number of faces	48	48
Truncation time (mili sec)	2500	2500
Max reflection order	2000	N/A
Number of rays (per source)	(Auto)1828	(Auto)18338
Scattering type	Lambert	Lambert
Active sources (CM20)	11	11
Receivers (Omni)	13	13
Output indicators	**Odeon**	**CATT**
Processing time (hh:mm)	00:04	01:10
Volume (m^3)	1133	1103
Total surface (m^2)	863	862
Mean Free Path (m)	5.7	5.6
Lost Rays (%)	0.2	0

Table 6:3 Input settings, calculation parameters and output indicators for each program simulation.

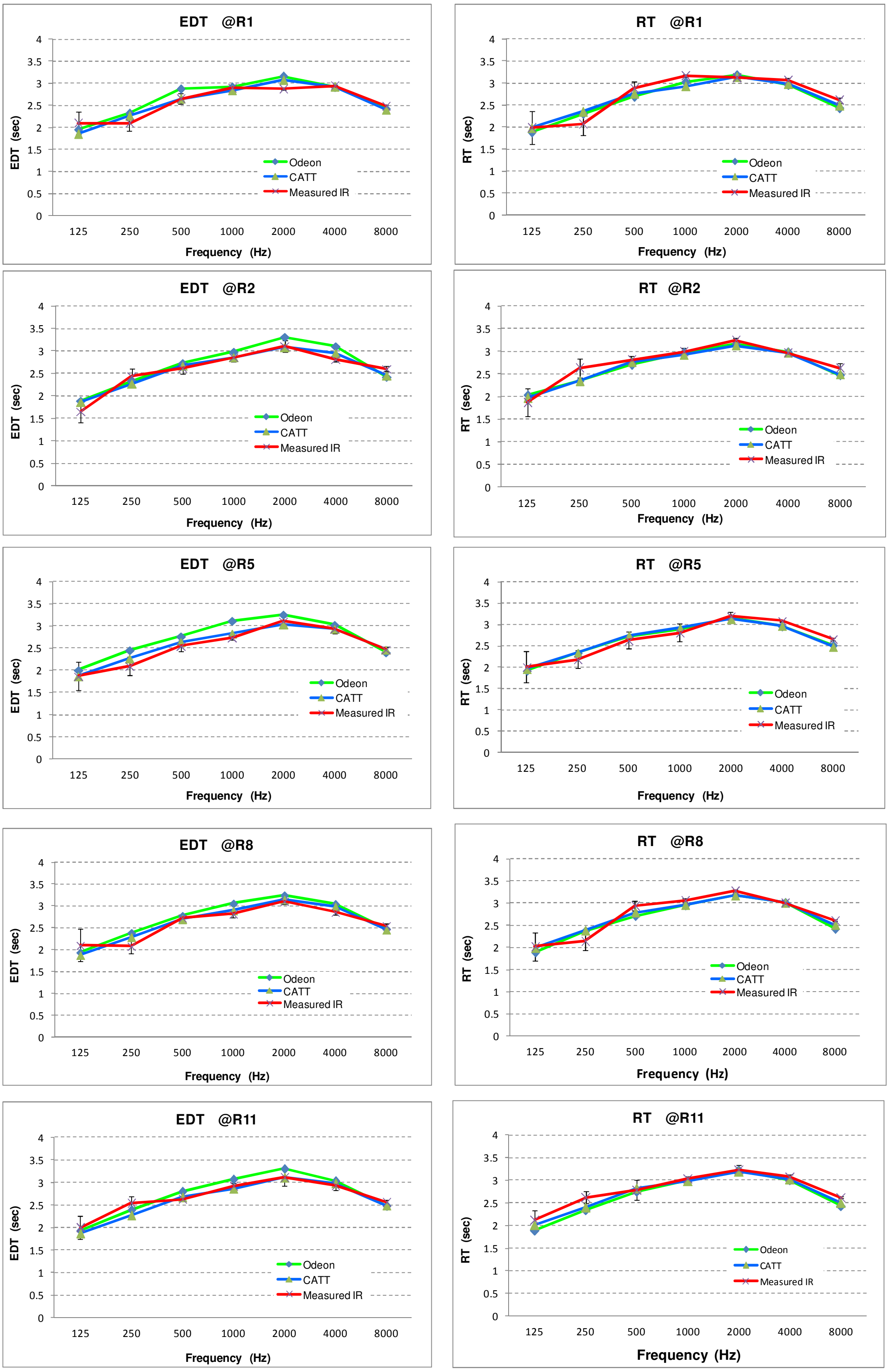


Figure 6:6 EDT and RT values measured and predicted by Odeon and CATT at each receiver position.

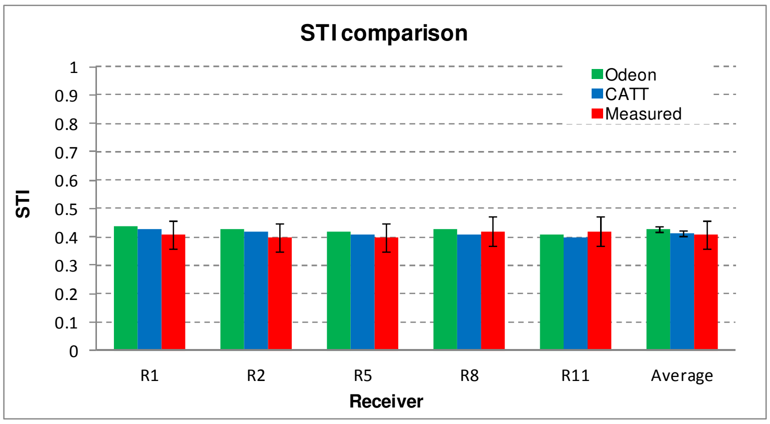


Figure 6:7 STI values measured and predicted by Odeon and CATT.

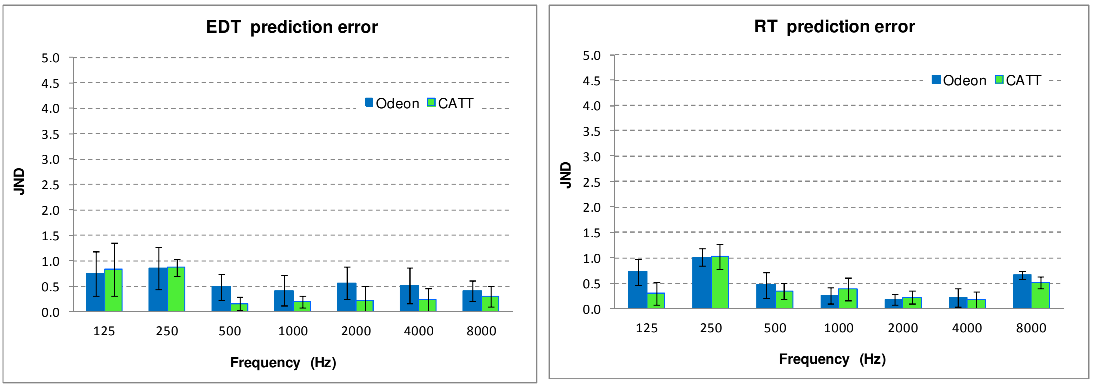


Figure 6:8 EDT and RT relative error averaged over five receiver positions

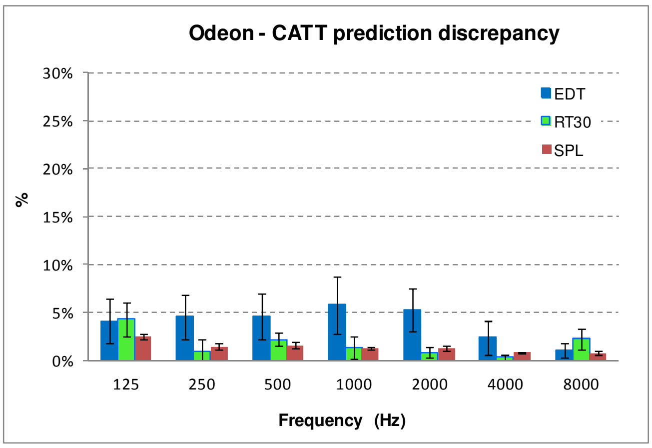


Figure 6:9 EDT, RT and SPL relative prediction discrepancy between the two programs averaged over the five receiver positions.

	Frequency (Hz)							
	125	250	500	1000	2000	4000	8000	dBA
Odeon	85.3	86.8	85.4	80.8	82.7	80.5	67.9	88.8
std	1.1	1.2	0.7	1.0	0.5	0.9	0.8	0.4
CATT	83.5	85.5	85.1	80.2	82.2	80.4	67.3	88.3
std	1.18	1.01	0.35	0.60	0.43	1.01	0.73	0.39
Discrepancy	1.8	1.3	0.3	0.6	0.5	0.1	0.6	0.5
Discrepancy JND	0.9	0.6	0.1	0.3	0.3	0.1	0.3	0.3

Table 6:4 Predicted SPL levels and absolute discrepancy averaged over the five receiver positions

	Receiver positions				
	R1	**R2**	**R5**	**R8**	**R11**
STI Odeon- CATT discrepancy	0.01	0.01	0.01	0.02	0.01

Table 6:5 STI absolute discrepancy between the two programs at each receiver position

6.3.2.3 Analysis and discussion

The prediction accuracy shown by both programs was remarkably high as indicated by figures 6:6 and 6:8 where the prediction error for EDT and RT averaged over the receiver positions was below 1 JND at all frequency bands. This result is supported by closer observation of prediction deviation at individual receivers in figure 6:6. In that figure it can be seen the small EDT and RT prediction deviations from measured values were consistently within 1JND at all receiver positions and at mid and high frequency bands. Deviation at low frequency bands (125Hz and 250Hz) were marginally above 1JND. Prediction accuracy at low frequencies by simulation programs based on hybrid techniques is naturally lower than at higher frequencies due to the inherent limitations of the geometrical acoustics approximation used in those techniques.

The high prediction accuracy by both programs was also shown in terms of the STI parameter for both programs simulations where the average STI predition error over the five receiver positions was below half the JND (figure 6:7 and appendix E.1).

The SPL prediction agreement between the two programs was remarkably close as shown by low discrepancy below 1 JND for all octave bands (figure 6:9, tables 6:4 and appendix E.1). Both program simulations yielded A-weighted SPL predictions at all receivers positions within 2dB (1JND) below the 90dBA target (table 6:4).

Calculations of the volume, internal surface area and MFP by both programs also showed a very close match (table 6:3) which supports the finding of both programs performing very closely in this space.

The high spatial uniformity of measured EDT and RT values as well as the strong correlation between EDT and RT values (EDT/RT ratio) at all frequencies and receiver positions clearly indicates the high diffuse field behaviour of this space correctly represented by both programs (figure 6:6 and appendix E.1).

The overall prediction error in JND units calculated for Odeon and CATT were 0.49 and 0.38 respectively. These figures indicate the remarkably high overall prediction accuracy of both programs and their close prediction agreement.

The processing time taken by CATT to complete the simulation was 17 times longer than Odeon (table 6:3). Odeon makes use of optimised hybrid calculation algorithms which do not produce exponentially growing number of reflections, but rather keeps the reflection density the same in all the calculations in order reduce calculation time. The industrial version of Odeon utilised in this study, is a simplified and optimised version of the complete program for typical industrial applications in which some architectural acoustics parameters, detailed graphical and numerical information present in the full version are not available. This lighter version of the program enables the calculation processes to be simpler resulting in faster calculation times. This is possibly another of the reasons for the important discrepancy in processing time between the two programs since CATT does not offer dedicated versions of the program to suit different types of applications or requirements (see chapter 2).

Another factor for the CATT slower run time is that this program calculates the diffused reflections according to input scattering coefficients assigned for the seven octaves bands whilst Odeon only calculates this once for one input octave band and then extrapolates values for the other octave bands. Moreover CATT projects many more rays per source and octave than Odeon under equivalent input and accuracy settings. It can be seen in table 6:3 CATT calculation algorithm utilized 10 times more rays than Odeon. This significant higher number of rays in CATT is intended to enable the additional prediction of other parameters and capabilities such as auralisation which are not available in the lighter industrial Odeon program.

The significant and consistent agreement observed between predicted and measured values for all parameters by both programs can be interpreted as a successful validation of the two programs in this first simulation case study. This provides assurance in the simulation capabilities of the programs and the suitability of the evaluation procedure for the subsequent simulation investigations of distributed VA systems.

6.3.3 Ticket Hall 2

6.3.3.1 Room Description

Ticket Hall 2 (TH2) is a large enclosed space of approximately 5000m^3 of volume and 6000m^2 of internal surface. It features a disproportionate and irregular shape, in which the ceiling height dimension (2.3m) is much smaller than main average length and width dimensions (30m x 24m respectively). All internal surface materials are similarly hard and acoustically reflective.

The main hall space is connected to three, narrow non-straight long corridors leading to narrow street exits at higher levels. The main space also incorporates large connecting openings to three escalators volumes (figures 6:10 and 6:11). This large hall presents higher complexity than TH1, hence it is reasonable to expect more discrepancies in the prediction calculations due to the large of number of distributed loudspeakers, intricate geometry and potential local coupling effects.

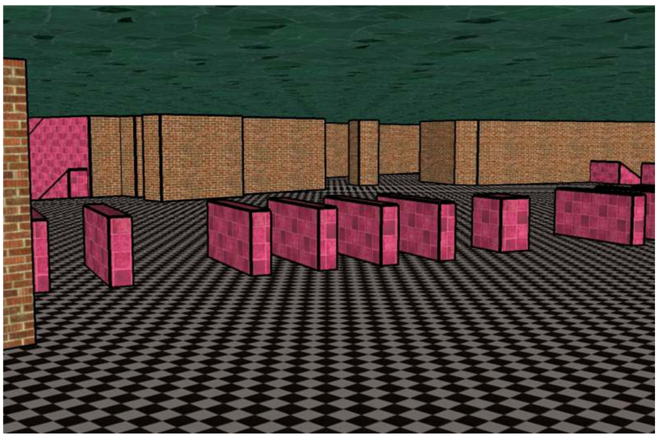


Figure 6:10 TH2 internal view of geometrical model rendered by Sketch Up Pro showing ticket barriers and two escalator entrances (in red)

Figure 6:11 shows the simulated VA loudspeaker arrangement distributed in a grid of 97 equally spaced ceiling loudspeakers (Next 2 CM5, see appendix F.3) at 3m spacing, built into the droped ceiling at height of 2.3m above the floor level. Thirteen simulated point receivers were distributed across the models floor area at a height of 1.5m above the floor. Six of the simulated receivers were placed at the receiver locations used for the acoustic measurements, these are the reported positions.

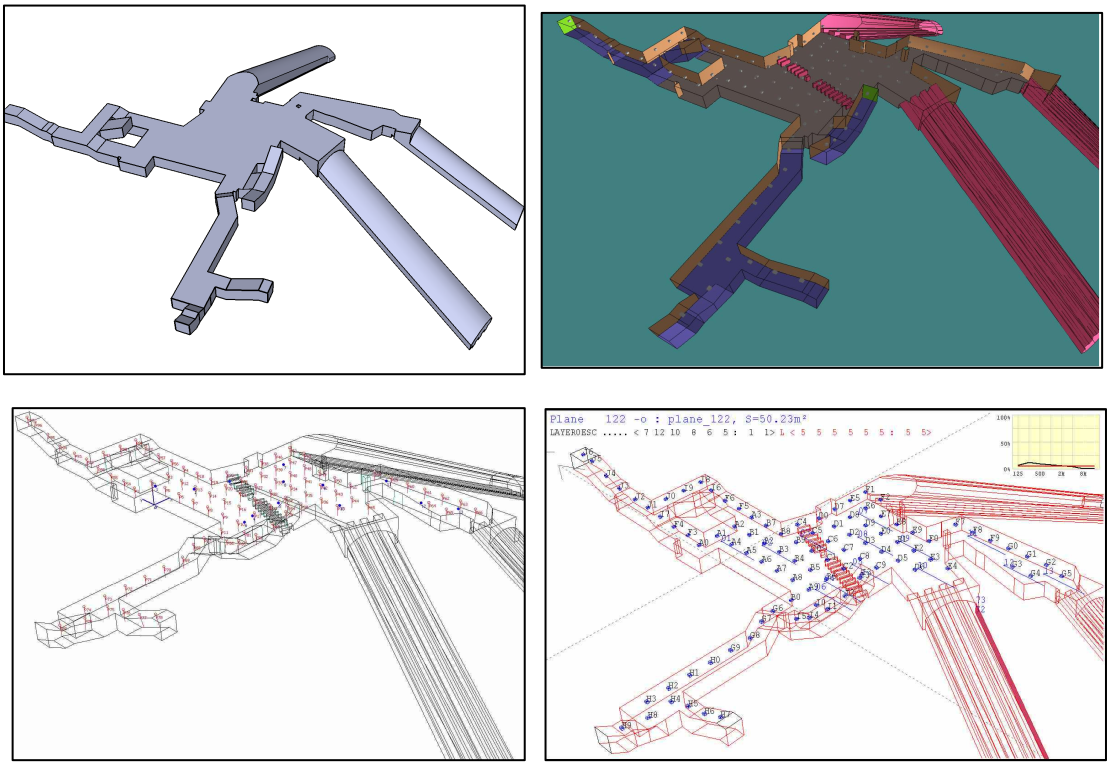
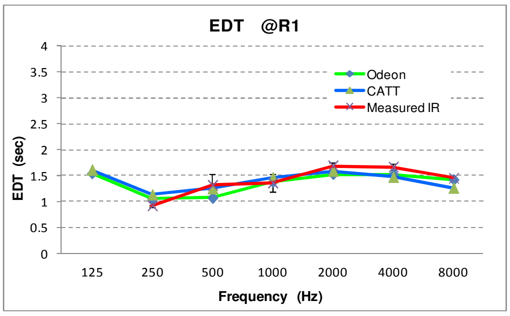


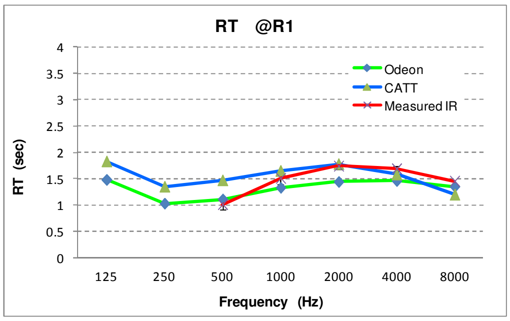
Figure 6:11 TH-2 Top left to bottom right figures from Sketch Up, CATT 3D viewer, Odeon and CATT wire frames view showing sources and receivers

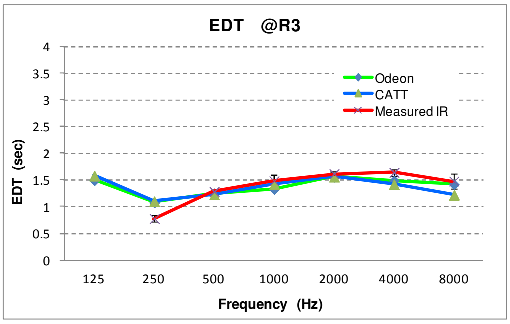
6.3.3.2 Results

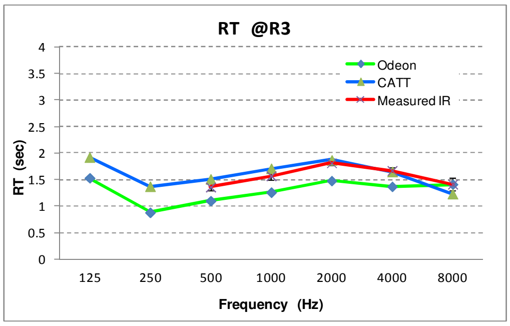
Input Settings	**Odeon**	**CATT**
Calculation algorithm	Hybrid	Hybrid
Number of faces	329	329
Truncation time (mili sec)	2000	2000
Max reflection order	2000	N/A
Number of rays (per source)	(Auto)~39903	(Auto)~75803
Scattering type	Lambert	Lambert
Active sources (CM20)	97	97
Receivers (Omni)	13	13
Output indicators	**Odeon**	**CATT**
Processing time (hh:mm)	01:36	21:10
Volume (m³)	4439	5308
Total surface (m²)	6001.7	5995
Mean Free Path (m)	2.96	3.23
Lost Rays (%)	0.6	2

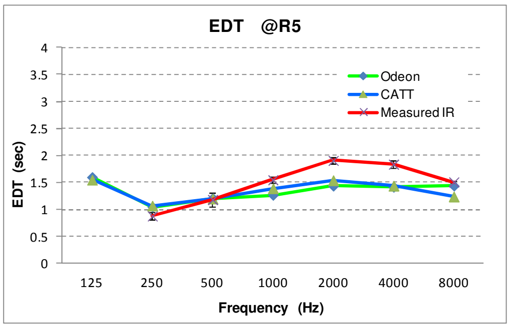
Table 6:6 Input settings, calculation parameters and output indicators for each program simulation

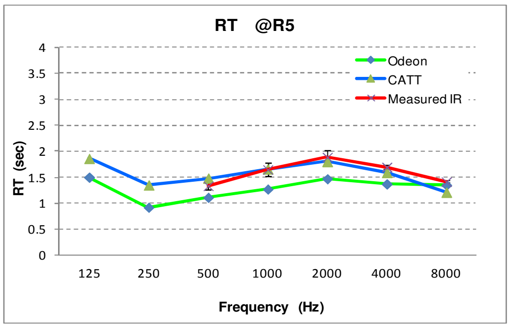

EDT @R1
EDT (sec)
Frequency (Hz)
125 250 500 1000 2000 4000 8000
Odeon
CATT
Measured IR

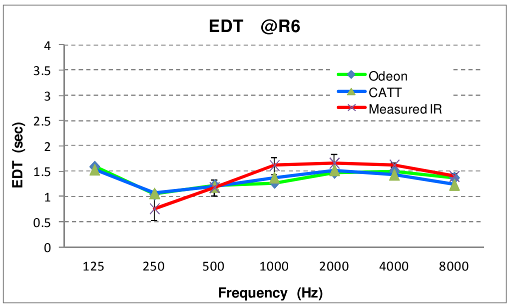

RT @R1
RT (sec)
Frequency (Hz)
125 250 500 1000 2000 4000 8000
Odeon
CATT
Measured IR

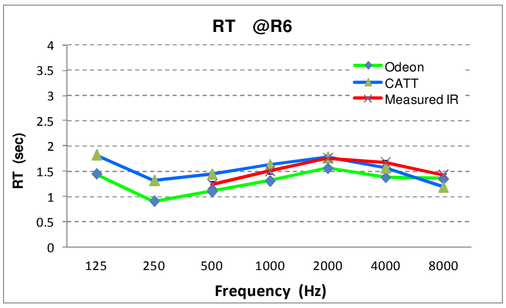

EDT @R3
EDT (sec)
Frequency (Hz)
125 250 500 1000 2000 4000 8000
Odeon
CATT
Measured IR

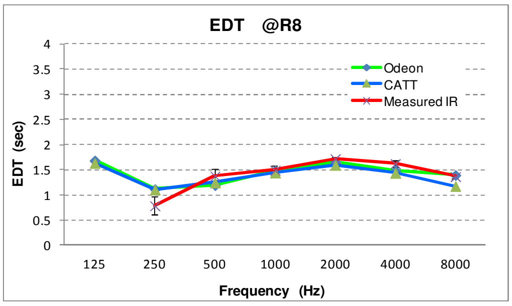

RT @R3
RT (sec)
Frequency (Hz)
125 250 500 1000 2000 4000 8000
Odeon
CATT
Measured IR


EDT @R5
EDT (sec)
Frequency (Hz)
125 250 500 1000 2000 4000 8000
Odeon
CATT
Measured IR


RT @R5
RT (sec)
Frequency (Hz)
125 250 500 1000 2000 4000 8000
Odeon
CATT
Measured IR


EDT @R6
EDT (sec)
Frequency (Hz)
125 250 500 1000 2000 4000 8000
Odeon
CATT
Measured IR


RT @R6
RT (sec)
Frequency (Hz)
125 250 500 1000 2000 4000 8000
Odeon
CATT
Measured IR


EDT @R8
EDT (sec)
Frequency (Hz)
125 250 500 1000 2000 4000 8000
Odeon
CATT
Measured IR

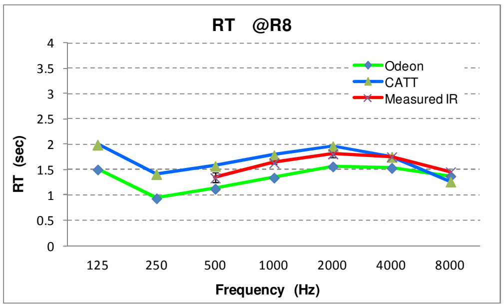

RT @R8
RT (sec)
Frequency (Hz)
125 250 500 1000 2000 4000 8000
Odeon
CATT
Measured IR

Figure 6:12 EDT and RT values measured and predicted by Odeon and CATT at receiver positions, (see receiver 12 in appendix E.2).

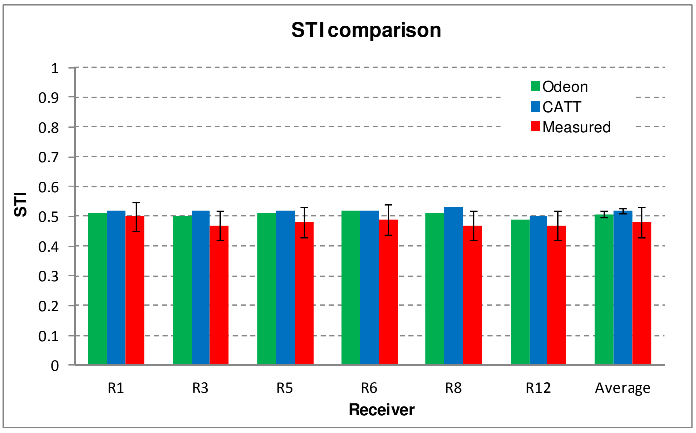


Figure 6:13 STI values measured and predicted by Odeon and CATT

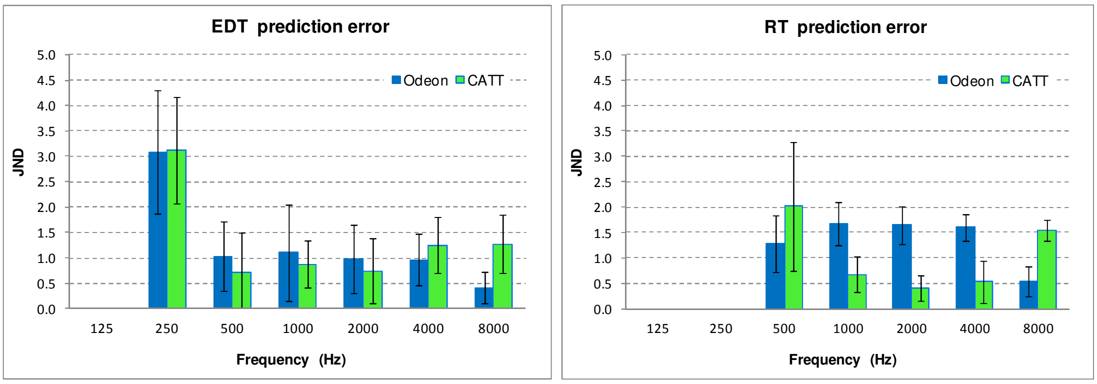


Figure 6:14 EDT and RT relative prediction error averaged over six receiver positions normalised to 1JND.

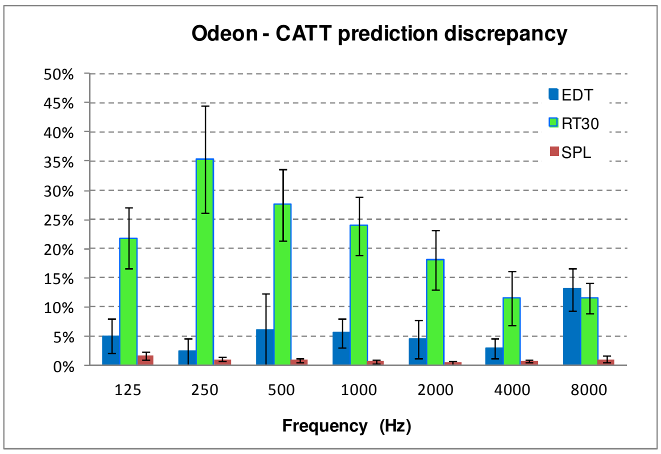


Figure 6:15 EDT, RT and SPL relative prediction discrepancy between the two programs averaged over the six receiver positions.

	Frequency (Hz)							
	125	**250**	**500**	**1000**	**2000**	**4000**	**8000**	**dBA**
Odeon	85.3	87.0	85.4	80.8	82.7	80.7	67.8	88.9
std	1.1	1.0	0.7	1.0	0.5	0.9	0.8	0.4
CATT	83.5	85.5	84.9	80.2	82.2	80.3	67.3	88.3
std	1.18	1.00	0.19	0.60	0.43	1.02	0.73	0.34
Discrepancy	1.8	1.4	0.5	0.6	0.5	0.3	0.5	0.6
Discrepancy JND	0.9	0.7	0.2	0.3	0.3	0.2	0.3	0.3

Table 6:7 Predicted SPL levels (dB) and discrepancy averaged over the five receiver positions

	Receiver positions					
	R1	**R3**	**R5**	**R6**	**R8**	**R12**
STI Odeon-CATT discrepancy	-0.01	-0.02	-0.01	0	-0.02	-0.01

Table 6:8 STI absolute discrepancy between the two programs at each receiver position

6.3.3.3 Analysis and discussion

Measurements of RT at 125Hz and 250Hz and EDT at 125Hz frequency octave bands were invalid or not reliable at the reported receiver positions due to limiting high background noise at the time of the measurement (figure 6:12 and Appendix E.2).

In figure 6:14 it can be seen that EDT predictions averaged over the receiver positions were close to the measured values over five octaves (generally within 1JND) (figures 6:12 and 6:14). A higher prediction error of up to 3.3JND was produced by both programs at 250 Hz. This higher error magnitude is caused by the relative nature of the calculation error (equation 6:1) where small absolute deviations from the measured reference (up to 0.3sec and 0.26sec on average) values appear as large relative error. However considering that measurement uncertainty of about 0.3sec is common at low frequency in reverberant spaces (Bork,2005a,b), (Segura et al, 2011) and the fact that an absolute deviation of 0.3sec is very unlikely to be perceived under a normal situation (Christensen et al, 2008b), it can make the EDT prediction error at 250Hz acceptable for practical purposes.

Both programs consistently predicted very similar values of EDT at all frequency bands and at all receiver positions. EDT discrepancies between the two programs averaged over the six receivers were below 1JND at all frequency bands except at 8kHz where it was marginally higher (1.3JND) (figures 6:12 and 6:15). The overall agreement between program predictions and those with measured values provided significant further evidence of the accuracy of both programs at predicting the early part of the simulated decay curve

RT predictions errors averaged over the receivers were within 1.6JND by Odeon and up to 2JND by CATT when observed across the five octave bands available. Looking in more detail at individual receiver results (figure 6:12) it can be seen that CATT consistently predicted RT values more accurately at all receivers and almost at all frequencies. Nevertheless, the range of RT prediction errors shown by the two programs in this space were similar to those found in other comparative studies of typically 1.5 JND (Vorländer, 1995), (Bork, 2000, 2005b), (San Martin & Arana, 2006), (Shiokawa & Rindel, 2007), (Christensen et al, 2008b),(Astolfi et al, 2008), (Hodgson et al,2008), (Segura et al, 2011).

The RT prediction discrepancy between the two programs was between 3.5 JND and 1.2JND showing a clear diminishing discrepancy trend towards high frequency bands (figure 6:12 and 6:15) where both programs predictions converged.

Unlike RT, late reflections have a low influence in the calculation process of EDT. It is in this EDT calculation where the two programs were in close agreement. This observation leads the author to believe that the discrepancy in RT was caused by the systematic different handling of late reflection in the programs' calculation algorithms. This fact might have been exacerbated by the large dimensions and complex geometry of this ticket hall. Similar orders of prediction discrepancies among different simulation programs have been reported in the literature (Vorländer, 1995), (Bork, 2000, 2005b), (Shiokawa & Rindel, 2007), (Hodgson et al, 2008), (San Martin & Arana, 2006), (Nagy et al, 2010), (Segura et al, 2011).

SPL predictions by the two programs were consistently in close agreement at all receivers and across the frequency range (table 6:7 and appendix E.2). Average discrepancies over the receiver positions were within 1JND at all octave bands except at 125Hz where the discrepancy was of 1.4 JND (appendix E.2). The level of discrepancies reported in this case are similar to values reported in a relevant work (San Martin & Arana, 2006). Both program simulations yielded overall A-weighted SPL predictions at all receivers positions within 2dB (1JND) below the 90dBA target (table 6:7).

Good STI prediction accuracy was achieved by both programs , with Odeon reporting errors within 0.8JND and CATT within 1.2JND (figure 6:13) showing absolute discrepancies of up to 0.02 (table 6:8). These results imply that a moderate degree of RT predicted error and discrepancy (up to 3JND observed above) does not significantly affect the correct prediction of STI values. The observed low sensitivity of STI to late reverberation in this space agrees with finding in previous chapters (see chapter 5 and 7) and other works (Christensen et al ,2008b), (Picard, & Bradley, 2001).

Calculations of the ticket hall internal surface area by both programs showed close agreement. However CATT volume calculation was 16% higher than Odeon calculation (table 6:6). The room volume discrepancy in this case could be attributable to the different ray tracing algorithms calculating the mean free path (MFP) from where the calculated room volume is derived. The extreme geometry as well as the large quantity of loudspeakers could lead to this discrepancy. The MFP value calculated by CATT was 8% higher than Odeon. Additionally the significant different number of rays emitted per source by each program may have been influential in the different volume calculation result. The slight higher volume and MFP calculated by CATT can be linked to the slightly higher predicted RT values.

The processing time taken by CATT for this space was 13 times longer than Odeon. The reasons for this significant difference are believed to be the same as expounded for TH1 simulation, albeit the more complex geometry and higher number of sound sources appears to have exacerbated the difference.

The overall prediction error figure calculated in JND units for Odeon and CATT were 1.05 and 1.04 respectively. These values indicate the high overall accuracy of both programs and their close prediction agreement. These values were slightly higher than the previous TH1case suggesting the higher simulation complexity of the present space.

6.3.4 Platform 1

6.3.4.1 Room Description

The geometrical and architectural characteristics of this platform are representative of the majority of deep London Underground platforms (see chapter 3), (figure 6:16 and 6:17). Platform-1 (Plat-1) is straight, has a volume of approximately 3060m³ and internal surface area of 2590m². It features a wide concave ceiling, 4 cross-passages openings, a passenger standing platform of 117m long by 3.3m wide, and a track pit of 3m wide and 0.7m deep. The height from the edge of the standing platform floor to the platform ceiling highest point is 4.2m. All internal surface materials of this platform are acoustically hard and reflective as is typical of underground spaces. The two tunnel openings and four cross-passage openings were modeled as areas of very high absorption and negligible scattering coefficient.

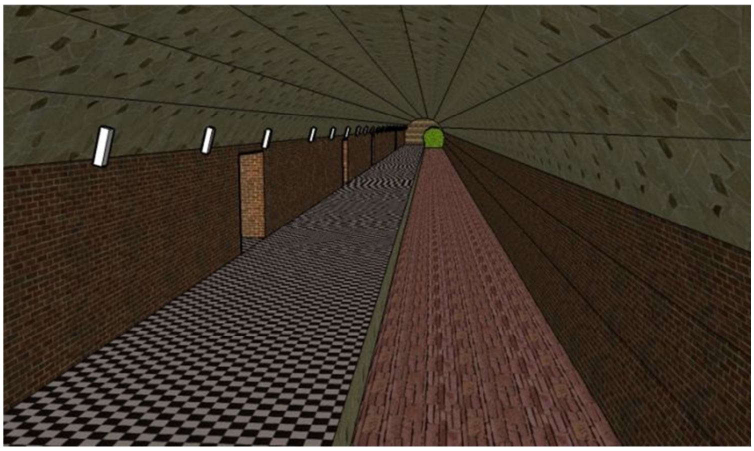


Figure 6:16 Sketch Up Pro Plat- 1internal view of geometrical model showing the surfaces, loudspeaker arrangement , cross passages and a tunnel opening.

Figures 6:16 and 6:17 show the conventional platform VA system configuration used on Plat- 1 featuring a distributed loudspeaker arrangement. Simulated receiver locations are also shown in figure 6:17. The simulated loudspeaker arrangement matching the actual one, comprised twenty three column loudspeakers (Next Two CM20, appendix F.2) equally spaced by 5m interval, fixed to the passenger standing platform wall at a height of 3m above the platform floor. The simulated loudspeakers were aimed at the edge of the standing platform replicating the aim angle in actual conditions for optimal coverage at listener plane (ear height,1.5m).

In the program simulations up to eleven point receivers were placed along the platform length at ear height. Five of the simulated receivers were placed at the five receiving positions used during the acoustic measurements as specified by the relevant performance test procedure (VA system specifications, 2005). Measurements and predictions reported correspond to those five receivers.

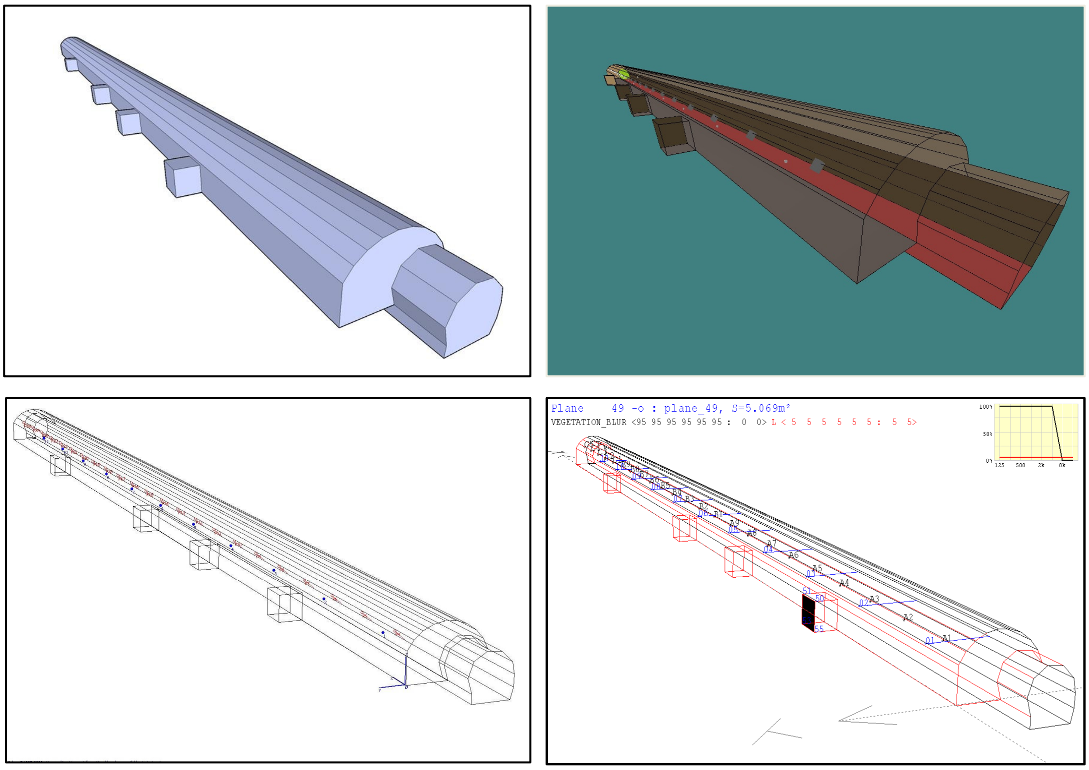


Figure 6:17 Plat- 1 Top left to bottom right: Sketch Up outside view, CATT 3D viewer, Odeon and CATT wire frame view showing sources and receivers.

6.3.4.2 Results

Input Settings	**Odeon**	**CATT**
Calculation algorithm	Hybrid	Hybrid
Number of faces	50	50
Truncation time (mili sec)	2500	2500
Max reflection order	2000	N/A
Number of rays (per source)	(Auto)10068	(Auto) 205760
Scattering type	Lambert	Lambert
Active sources (CM20)	23	23
Receivers (Omni)	11	11
Output indicators	**Odeon**	**CATT**
Processing time (hh:mm)	00:15	06:40
Volume (m^3)	3064	3078
Total surface (m^2)	2590	2586
Mean Free Path (m)	4.55	4.56
Lost Rays (%)	0.5	0.1

Table 6:9 Input settings, calculation parameters and output indicators for each program simulation.

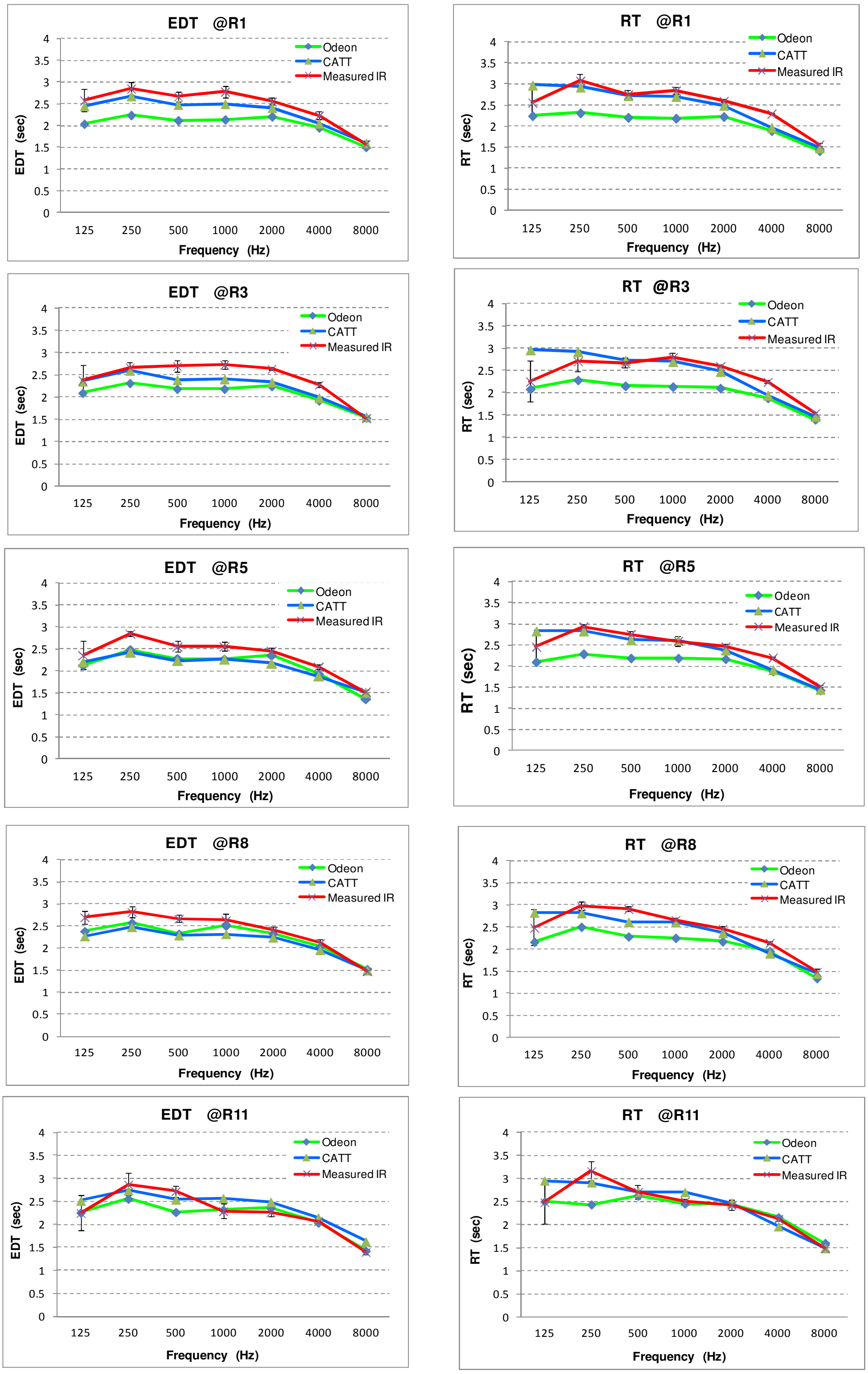


Figure 6:18 EDT and RT measured and predicted by Odeon and CATT at each test position.

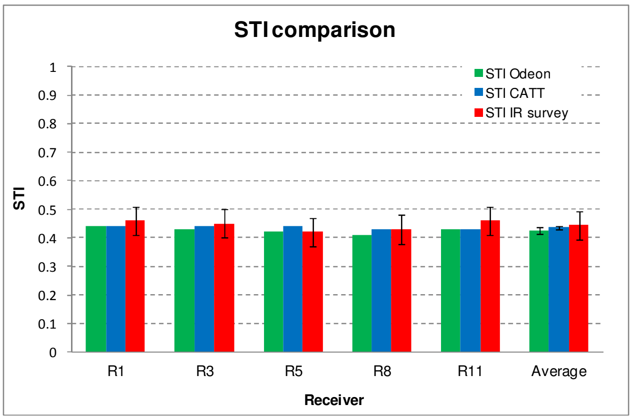
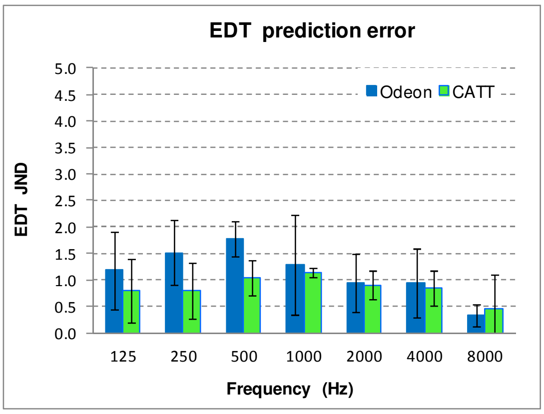


Figure 6:19 STI values measured and predicted by Odeon and CATT.



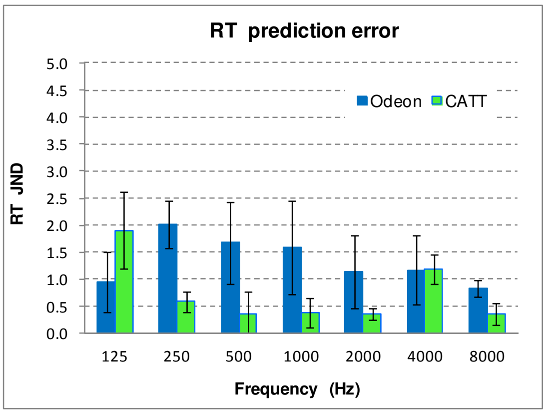


Figure 6:20 EDT and RT relative prediction error averaged over five test positions normalised to 1JND.

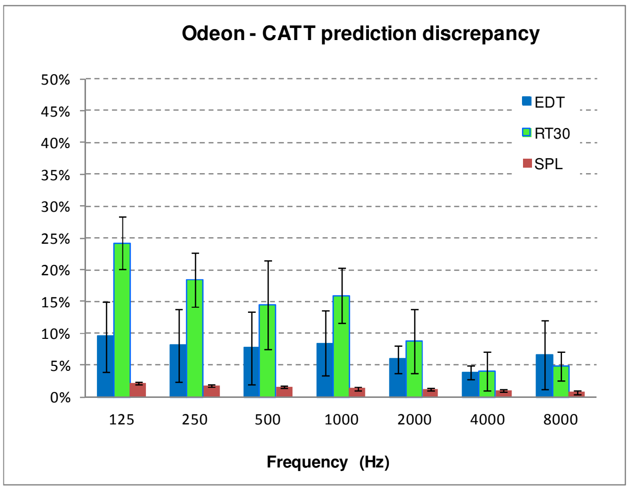


Figure 6:21 EDT, RT and SPL relative prediction discrepancy between the two programs averaged over five test positions.

	Frequency (Hz)							
	125	**250**	**500**	**1000**	**2000**	**4000**	**8000**	**dBA**
Odeon	82.9	89.4	84.7	86.1	79.8	73.2	69.5	89.18
std	0.38	0.45	0.36	0.53	0.42	0.27	0.25	0.4
CATT	81.1	87.9	83.3	85.0	78.9	72.5	69.1	88.03
std	0.36	0.43	0.38	0.49	0.51	0.23	0.50	0.37
Discrepancy	1.78	1.54	1.32	1.1	0.92	0.7	0.4	1.15
Discrepancy JND	0.89	0.77	0.66	0.55	0.46	0.35	0.21	0.58

Table 6:10 Predicted SPL levels (dB) and discrepancy averaged over the five test positions.

	Receiver positions				
	R1	**R3**	**R5**	**R8**	**R11**
CATT discrepancy	0	-0.01	-0.02	-0.02	0

Table 6:11 STI absolute discrepancy between the two programs at each receiver position

6.3.4.3 Analysis and discussion

Odeon EDT prediction error averaged over the test positions was within 1.8JND when observed across octave bands while CATT was within 1.1JND (figures 6:18, 6:20 and Appendix E.3). EDT prediction discrepancy between the programs was consistently low for all five receiver positions at all frequency bands with an overall discrepancy below 1JND (figure 6:21 and appendix E.3).

In terms of RT, CATT appeared to predict more accurately than Odeon showing an averaged prediction error within 1.2JND when observed across the octaves bands, except at 125Hz were the error was 1.9JND (figures 6:18 and 6:20). Odeon gave consistently slightly higher prediction errors at all receiver positions and frequency bands except at 125Hz where errors were lower than CATT. Odeon prediction error averaged over the receiver positions was within 2JND observed across the frequency bands (figure 6:20 and appendix E.3).

EDT and RT predictions for both programs showed general convergence with the measurements as frequency increased. This can attributed to the general increased accuracy in the estimation of the acoustic properties assigned to surface materials in the higher octave bands (figure 6:18).

As before, the shape similarity between measured and predicted EDT and RT curves as well as the notable convergence of results in the higher octave bands, denote that both programs correctly modeled the acoustic behavior of the space (figure 6:18). Prediction errors by both programs in the low frequency bands were generally greater than in the other octaves bands. However the range of prediction errors (within 2 JND) observed on both programs results for those frequency bands were not considered significant appearing comparable with prediction errors in other similar works (Bork, 2000, 2005b), (Hodgson et al, 2008), (San Martin & Arana, 2006), (Segura et al, 2011). To put these prediction errors into perspective it is important to note that speech intelligibility and STI are not sensitive to moderate variations of reverberation particularly at low frequencies bands (see 6.3.2 and chapter 2).

The fact that both programs' EDT predictions were essentially in good accord leads the author to think that the Odeon RT prediction errors might have been caused in the handling of the late reflections.

SPL predictions from both programs agreed closely for all receivers across all frequencies bands. Discrepancies averaged over the five receiver locations were not significant and they reduced with increasing frequency, with the highest discrepancy being of 0.9JND at 125Hz (figure 6:21, table 6:10 and appendix E.3). This range of SPL discrepancy between the programs is consistent with similar deviations (within 1JND) found in a similar comparative study (Hodgson et al, 2008). Both program simulations yielded overall A-weighted SPL predictions at all receivers positions within 2dB (1JND) below the 90dBA target (table 6:10).

Prediction accuracy, in terms of STI, by both programs was remarkably good, with both programs reporting errors within 0.6JND. The prediction discrepancy between the two programs across receiver positions was within 0.4JND corresponding to absolute discrepancies of up to 0.02 (figure 6:19 and table 6:11). These results are consistent with previous observations (see TH2 in 6.3.2) indicating that a moderate degree of RT predicted error at low and mid frequency (Odeon RT error) does not affect the correct prediction of STI value.

Calculations of the volume, internal surface area and MFP by both programs also showed close agreement providing supporting evidence for the models highly comparable calculation performance (table 6:9). CATT employed 26 times the processing time needed by Odeon to complete the simulation mainly due to the much higher number of ray projected from each source.

Measured EDT/RT ratios agreed closely with the predicted ratios produced by both programs. The observed high spatial uniformity of measured EDT and RT values as well as the high EDT/RT correlation ratios (decay curve linearity) at all frequencies and receiver positions (figure 6:19 and appendix E.3), resembles the typical characteristics of a diffuse sound field . This phenomenon will be further discussed in chapter 8.

The overall prediction error value calculated in JND units for Odeon and CATT were 0.95 and 0.64 respectively. These low values indicate the high overall accuracy of both programs and their comparble prediction agreement.

These results provided confidence in the prediction accuracy of the two programs when modeling a straight tubular-like geometry equipped with a distributed sound system. The CATT simulation of this platform 1 will be used as a validated model for investigations in chapters 7 and 8.

6.3.5 Platform 2

6.3.5.1 Room Description

Platform 2 (Plat-2) is a type of deep underground platform exhibiting similar geometry characteristics as for Plat-1, except for three passenger over-bridges situated approximately in the centre section of the platform length. The separation between the central and the two adjacent overbridges are 5.3m and 15.6m. The over bridges dimensions at their base level are 5.97m long, 1.65m to the highest point on the platform ceiling, and 1.9m and 2.56m wide. Their base length run overhead perpendicular to the length of platform at 2.85m high from the standing platform (figure 6:22). The sides of the overbridges are fully covered by heavy metal panels not offering contact with other station space.

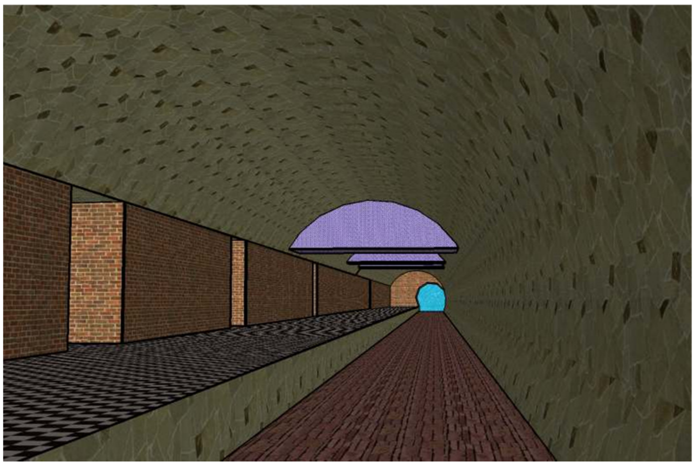


Figure 6:22 Sketch Up Pro Plat-2 internal view of the geometrical model showing the three over bridges (purple), cross passages and one tunnel opening (blue).

It was initially hypothesized that these over-bridges could challenge the programs prediction calculation algorithms with regards to diffraction and diffusion simulation processes. The over bridges appear as large obstacles of comparable or larger size to a range of wave lengths which could obstruct progressive propagation of direct and first reflections sound from distant sources.

Figure 6:23 shows the VA distributed loudspeaker arrangement and receiver locations utilised in the simulations. The actual and simulated loudspeaker arrangement incorporated twenty six column type loudspeakers configured as in Plat-1. Five of the simulated receivers were placed at the specified five receiving positions on the listening plane used during the acoustic measurements and are reported upon below.

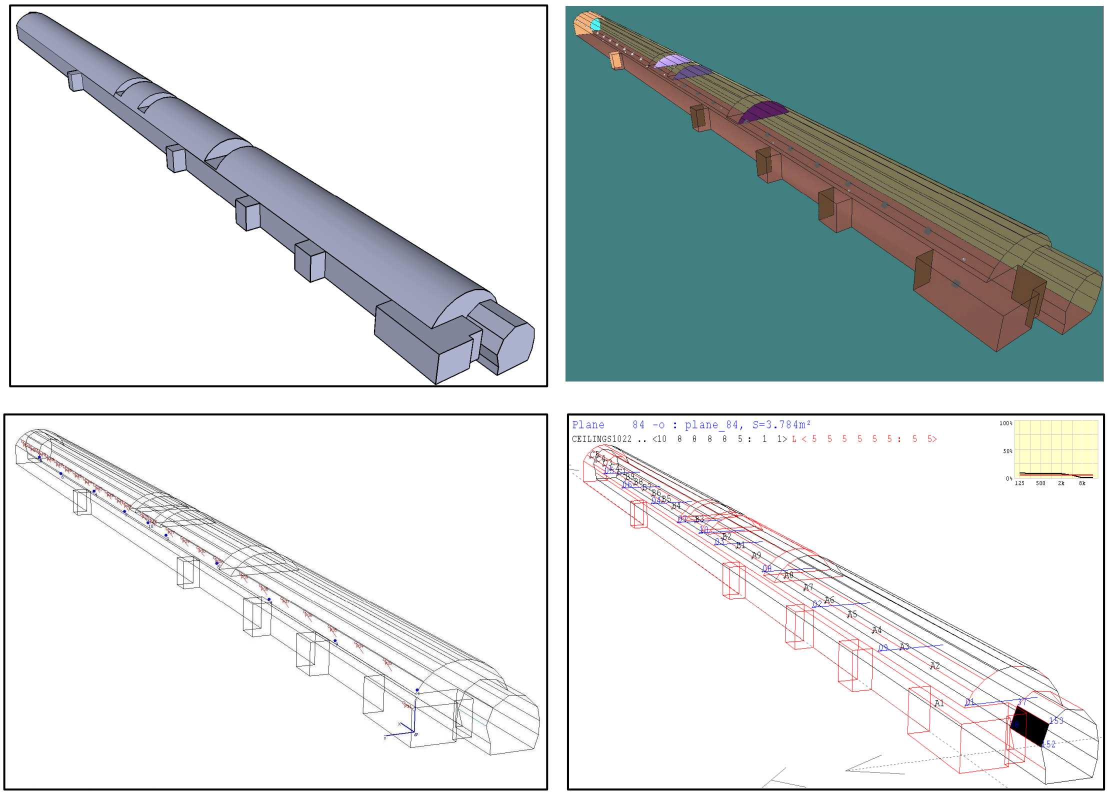


Figure 6:23 Plat- 2 Top left to bottom right figures from Sketch Up outside view, CATT 3D viewer, Odeon and CATT wire frame view showing sources and receivers.

6.3.5.2 Results

Input Settings	Odeon	CATT
Calculation algorithm	Hybrid	Hybrid
Number of faces	87	87
Truncation time (mili sec)	3000	3000
Max reflection order	2000	N/A
Number of rays (per source)	(Auto) 378612	(Auto)~197653
Scattering type	Lambert	Lambert
Active sources (CM20)	26	26
Receivers (Omni)	10	10
Output indicators	**Odeon**	**CATT**
Processing time (hh:mm)	00:55	11:04
Volume (m³)	2814	2872
Total surface (m²)	2402	2373
Mean Free Path (m)	4.46	4.69
Lost Rays (%)	0.5	0

Table 6:12 Input settings (calculation parameters) and output parameters

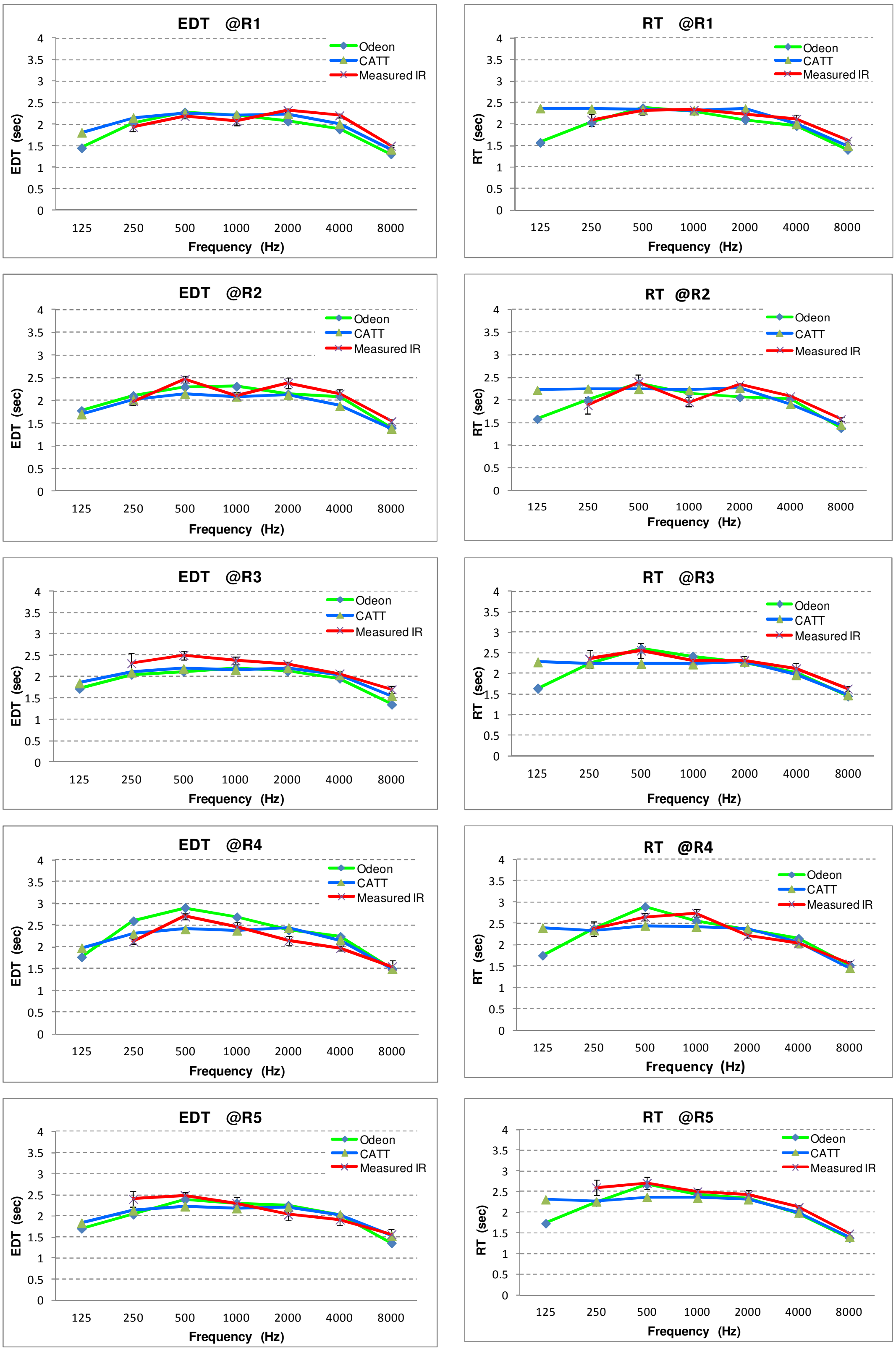


Figure 6:24 EDT and RT measured and predicted by Odeon and CATT at each receiver position.

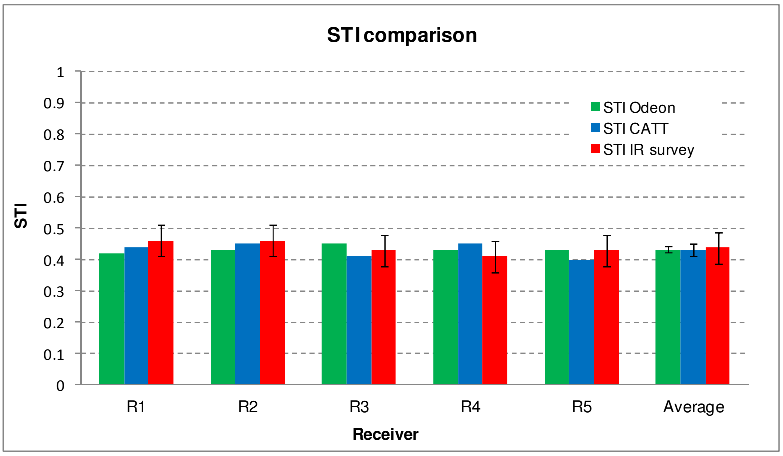


Figure 6:25 STI values measured and predicted by Odeon and CATT at the five receiver positions.

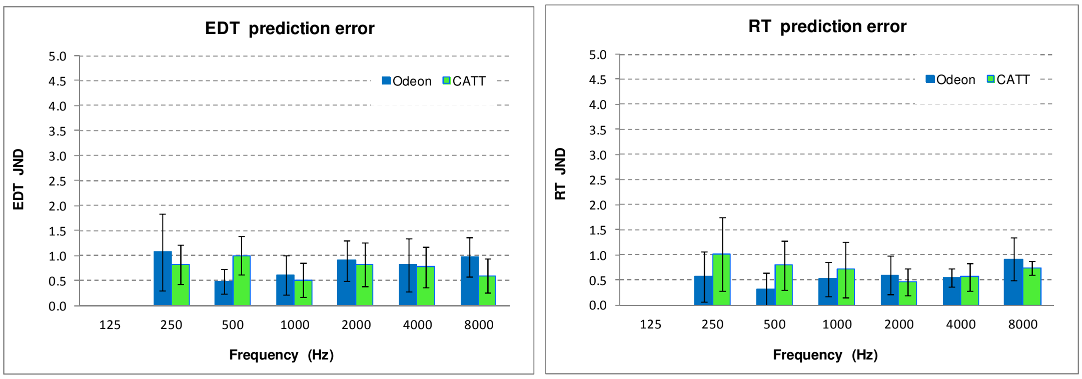


Figure 6:26 EDT and RT absolute prediction error averaged over five receiver positions normalised to 1JND.

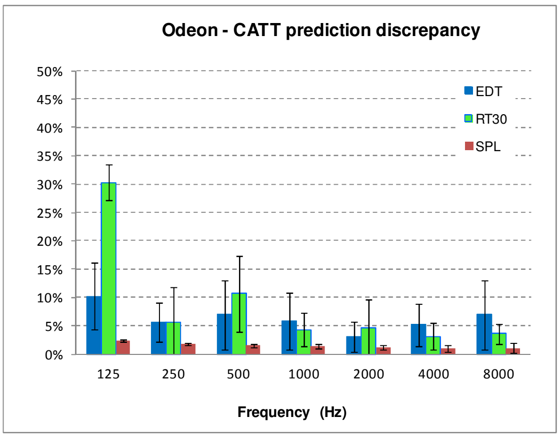


Figure 6:27 EDT, RT and SPL relative prediction discrepancy between the two programs averaged over the five receiver positions.

	Frequency (Hz)							
	125	**250**	**500**	**1000**	**2000**	**4000**	**8000**	**dBA**
Odeon	82.1	88.8	84.6	86.1	80.6	73.5	67.9	89.28
std	0.41	0.86	0.61	0.41	0.85	0.56	4.25	0.3
CATT	80.2	87.3	83.4	84.9	79.7	72.8	67.3	88.12
std	0.36	0.92	0.68	0.21	0.96	0.32	4.71	0.27
Discrepancy	1.92	1.54	1.24	1.18	0.9	0.72	0.7	1.16
Discrepancy JND	0.96	0.77	0.62	0.59	0.45	0.36	0.33	0.58

Table 6:13 Predicted SPL levels (dB) and discrepancy averaged over the five receiver positions

	Receiver positions				
	R1	**R2**	**R3**	**R4**	**R5**
STI Odeon- CATT discrepancy	-0.02	-0.02	0.04	-0.02	0.03

Table 6:14 STI absolute discrepancy between the two programs at the receiver positions

6.3.5.3 Analysis and discussion

As it can be seen in figure 6:24 EDT prediction errors by both programs were consistently low at all receiver positions and for all frequency bands available. Odeon prediction error averaged over the receiver positions was within 1.1JND when observed over the six frequency octave bands while CATT was within 1JND (figures 6:24 and 6:26). Both programs predicted consistently very similar values of EDT at all frequency bands and at all receiver positions showing average discrepancies within 1JND (figure 6:27).

Both programs appeared to correctly predict RT values. Odeon showed prediction errors averaged over the receiver positions within 0.9JND when observed across the octaves bands while CATT errors were within 1JND (figures 6:24 and 6:26).

The RT prediction discrepancy between the two programs was only significant at 125Hz (3JND). This moderate discrepancy was consistent with Plat-1 discrepancy and therefore very likely to be generated by the same causes. An additional contributing factor in the prediction disagreement at 125Hz for this platform was the lack of reference measured data in the model calibration process. Nevertheless the magnitude of that discrepancy is comparable to others reported in other similar comparative works (Dance, & Shield, 1999), (Bork, 2000, 2005b), (Hodgson et al, 2008), (San Martin & Arana, 2006), (Segura et al, 2011).

The similar shape of measured and predicted EDT and RT curves denote that both programs correctly modeled the acoustic behavior of the space (figure 6:24).

SPL predictions from both programs agreed closely across all receivers at all frequencies bands. SPL prediction discrepancies averaged over the five receiver locations were insignificant and again decreased with frequency with the highest discrepancy being of 0.9JND at 125Hz (figure 6:27 and table 6:13). Both program simulations yielded overall A-weighted SPL predictions at all receivers positions within 2.5dB (1.2JND) below the 90dBA target (table 6:13).

Prediction accuracy by the two programs in terms of STI was high, with both programs reporting errors over the receiver positions within 0.8JND, with absolute discrepancies of up to 0.04. This indicated strong agreement between programs in the calculations of STI values (figure 6:25 and table 6:14).

Calculations of the volume, internal surface area and MFP by both programs showed a close agreement which provides supportive indication of their highly comparable overall calculation performance (table 6:12). However, and consistent with previous case studies, CATT took several times longer than Odeon to complete the simulation (a factor of 12).

Considering that Plat-1 and Plat-2 are geometrically equivalent except for the presence of the three over bridges on Plat-2, a relative comparative analysis in terms of processing time gave the following results:

Odeon took 3.7 times longer to calculate Plat-2 simulation than for Plat-1 while CATT was only 1.7 times longer for the same comparison. This was due to an increment of a factor of 37 in number of rays projected by Odeon for Plat-2 with respect to Plat-1. Hence it can be realised that the introduction of the three over bridges had a significant impact on Odeon processing time. This was due to Odeon calculation algorithm being adaptive (automatic selection of number of rays) to calculate time-efficiently depending on the detected geometry complexity. However CATT showed a less adaptive sensitivity to the introduction of geometrical complexity (over bridges) by comparatively no increasing the number of rays projected between Plat-1 and Plat-2 simulations.

An analysis on Plat-2 compared measured values between receiver positions near to the overbridges (R3) and far from them (R1,R2, R5). Results did not show significant differences between relevant positions when observed over the frequency bands (overall within 0.7 JND for EDT and 0.8 for RT and 0.6JND for STI). The same order of differences was found when the same comparative analysis was performed on the same receiver positions although on Plat-1 which has no bridges (overall within 0.6 JND for EDT, 0.5 JND for RT and 0.8JND for STI).

An additional analysis compared measured values on Plat-1 and Plat-2 observed at the same receiver position (near the over bridges for Plat-2). As in the previous analysis no significant difference was observed between values of that receiver position for the two platforms when observed over the octave bands (overall within 0.7 JND for EDT, 0.9 JND for RT and 0.2JND for STI). The insignificant differences observed in the analysis demonstrated that the presence of over bridges does not influence the sound field measured on a deep underground platform equipped with a distributed loudspeaker system. This finding is consistent with the relevant conclusion derived from measured values analysis of chapter 5.

The overall prediction error calculated for Odeon and CATT were 0.61 and 0.65 JND respectively. These figures indicate the high overall accuracy of both programs and their close prediction agreement for this featured geometry. This provides further confidence in the programs in the case of increased geometrical complexity of deep platforms.

6.3.6 Platform 3

6.3.6.1 Room Description

Platform 3 (Plat-3) is a type of underground station platform with similar geometrical and architectural characteristic to Plat-1, volume and internal surface area of 2900m^3 and 2400m^2 respectively. However this platform possesses the distinguishing geometrical feature of a sharp lengthwise concave curvature (figure 6:28). The passenger standing platform is 128m long and has three cross-passages openings, (figure 6:29). The height from the edge of passenger standing platform floor to the highest point on the ceiling is 4.2m.

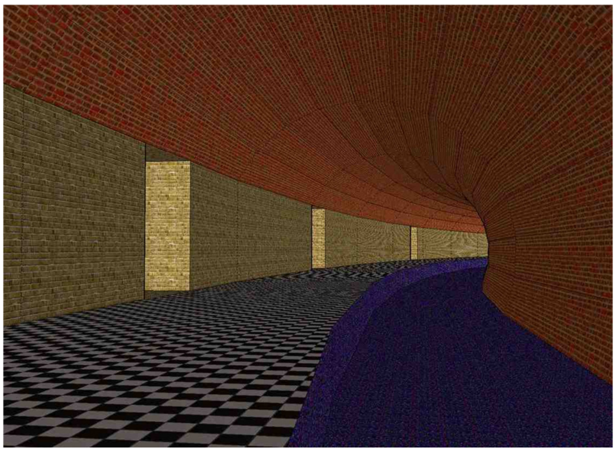


Figure 6:28 Sketch Up Pro Plat- 3 internal view of the geometrical model showing the platform curvature and three cross passages.

Figure 6:29 shows the VA distributed loudspeaker arrangement and receiver locations utilised in the simulations. The simulated loudspeaker configuration replicated the actual one and was the same as described for Plat-1. In the program simulations nine point receivers were used, however only five of the simulated receivers were placed at the receiver positions used during the acoustic measurements, these are reported upon below.

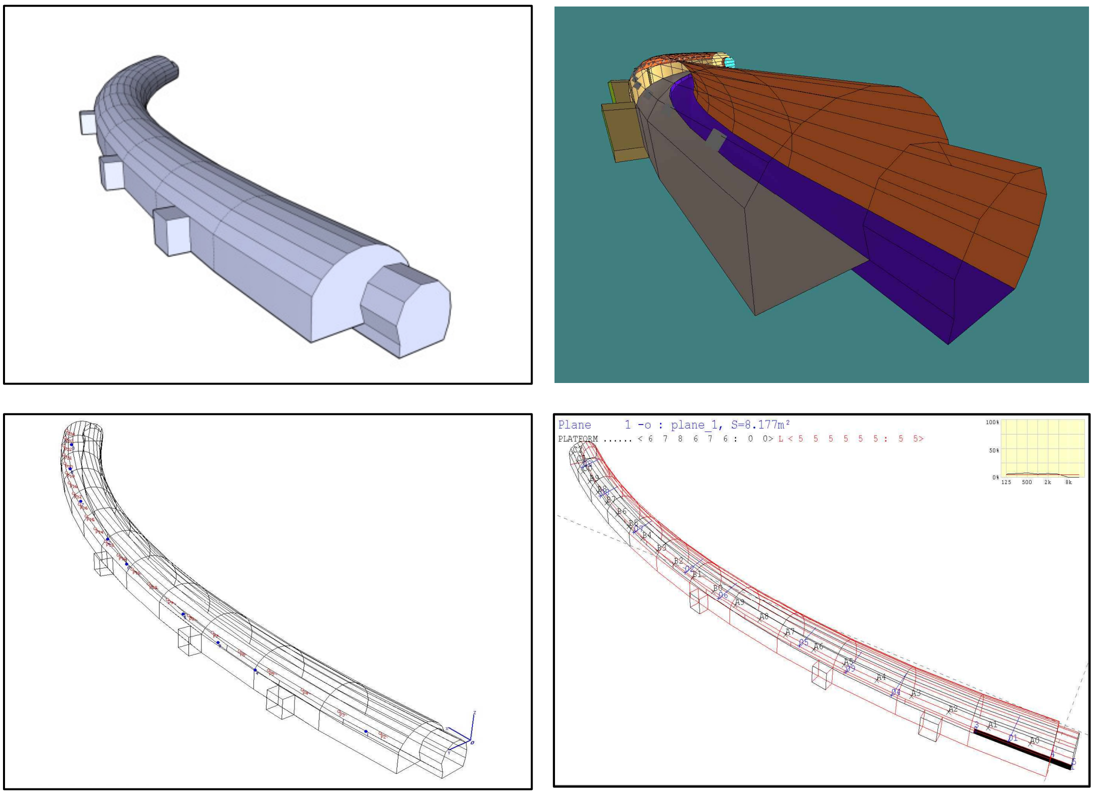


Figure 6:29 Plat- 3, Top left to bottom right figures from Sketch Up outside view, CATT 3D viewer, Odeon and CATT wire frame view showing sources and receivers.

6.3.6.2 Results

Input Settings	**Odeon**	**CATT**
Calculation algorithm	Hybrid	Hybrid
Number of faces	198	198
Truncation time (mili sec)	3500	3500
Max reflection order	2000	N/A
Number of rays (per source)	(Auto)15216	(Auto)198639
Scattering type	Lambert	Lambert
Active sources (CM20)	23	23
Receivers (Omni)	9	9
Output indicators	**Odeon**	**CATT**
Processing time (hh:mm)	00:24	10:40
Volume (m³)	2843	2958
Total surface (m²)	2412	2430
Mean Free Path (m)	4.58	4.9
Lost Rays (%)	0.6	0

Table 6:15 Input settings (calculation parameters) and output parameters.

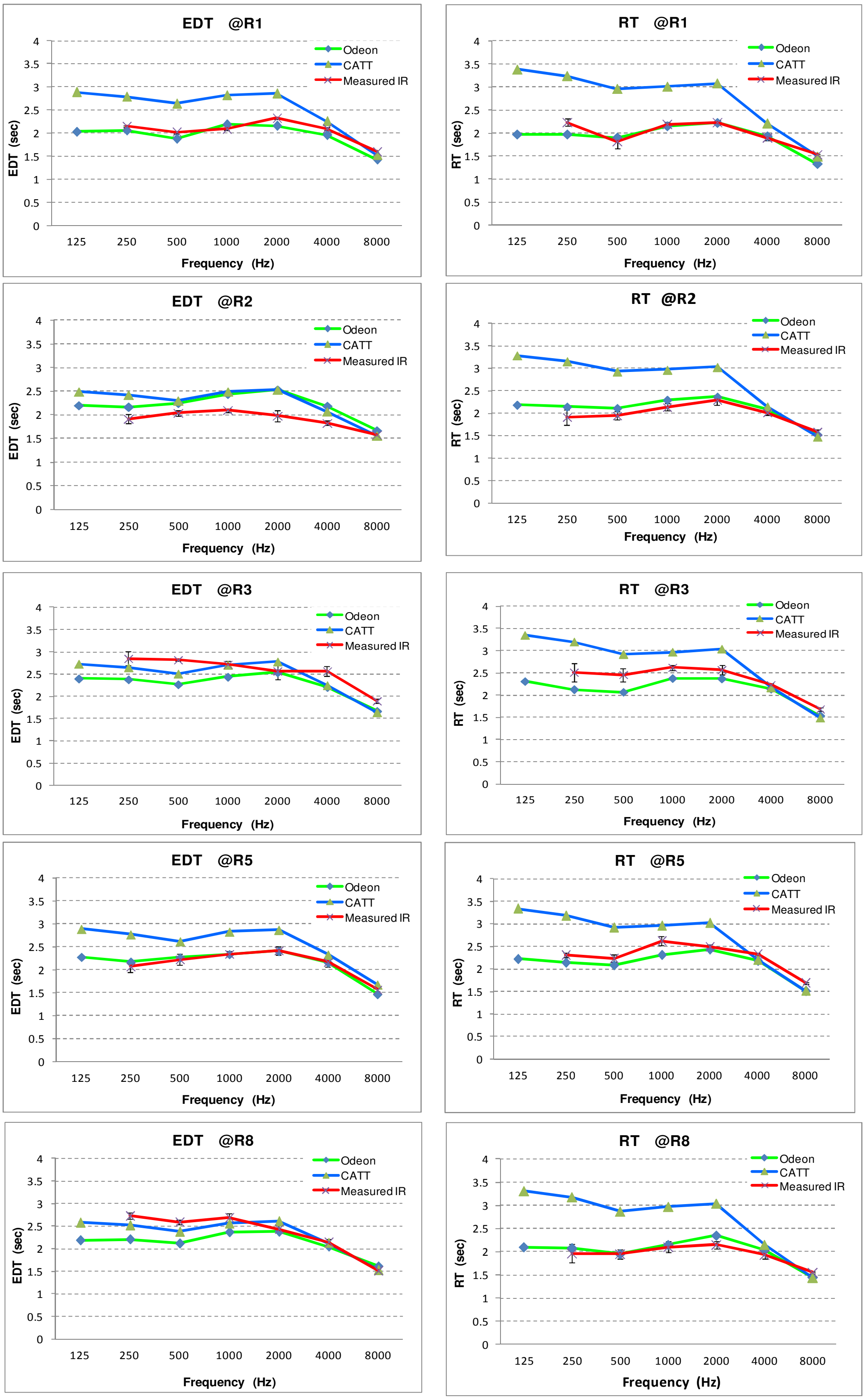


Figure 6:30 EDT and RT values measured and predicted by Odeon and CATT at each receiver position.

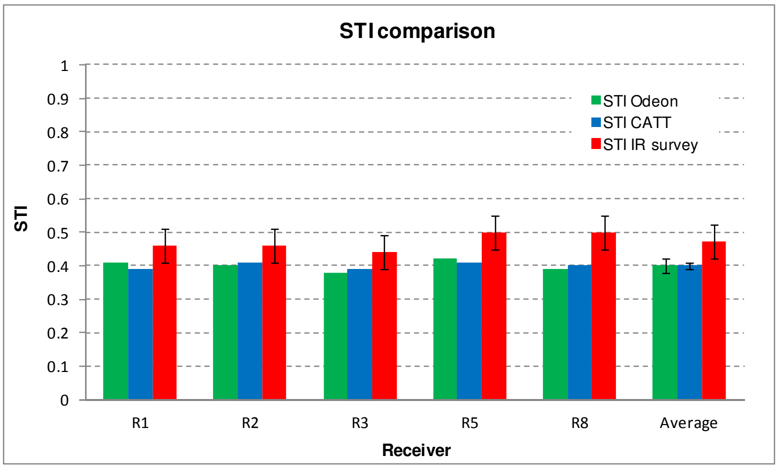


Figure 6:31 STI values measured and predicted by Odeon and CATT at five receiver positions.

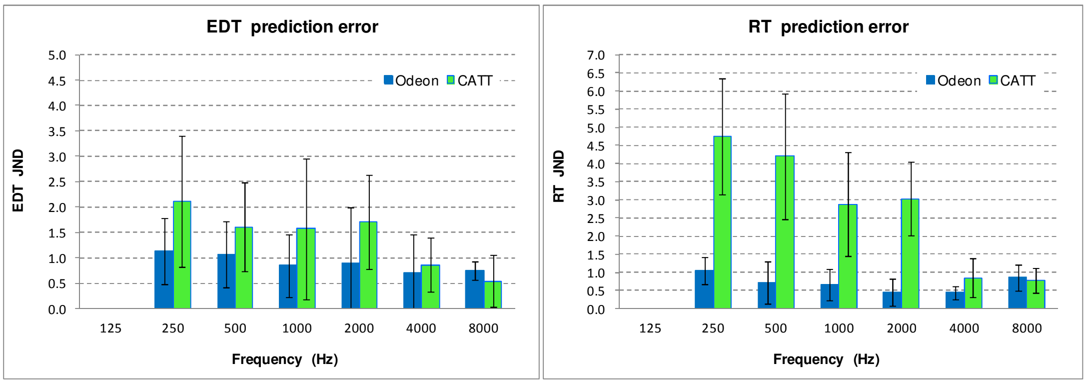


Figure 6:32 EDT and RT relative prediction error averaged over five receiver positions normalised to 1JND.

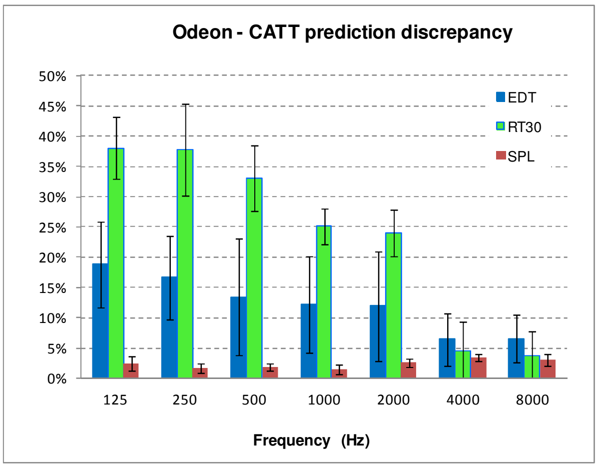


Figure 6:33 EDT, RT and SPL relative prediction discrepancy between the two programs averaged over the five receiver positions.

	Frequency (Hz)							
	125	**250**	**500**	**1000**	**2000**	**4000**	**8000**	**dBA**
Odeon	83.6	87.4	85.8	85.0	81.2	73.6	69.5	89.07
std	0.65	0.81	0.50	0.47	1.63	0.52	0.30	0.2
CATT	81.6	85.8	84.3	83.7	79.8	72.4	68.2	87.64
std	0.68	0.51	0.75	0.30	1.68	0.57	0.50	0.41
Discrepancy	1.96	1.62	1.56	1.32	1.4	1.18	1.2	1.43
Discrepancy JND	0.98	0.81	0.78	0.66	0.7	0.59	0.62	0.71

Table 6:16 Predicted SPL levels and absolute discrepancy averaged over the five receiver positions

	Receiver positions				
	R1	**R2**	**R3**	**R5**	**R8**
STI Odeon- CATT discrepancy	0.02	-0.01	-0.01	0.01	-0.01

Table 6:17 STI absolute discrepancy between the two programs at each receiver position

6.3.6.3 Analysis and discussion

Odeon EDT predictions on this platform were accurate with relative errors averaged over the receiver positions within 1.1JND when observed across the frequency octave bands (figures 6:30 and 6:32). Looking at specific receiver location results, it can be observed that receiver 2 was less accurately predicted than the other receivers with error of up to 2.8JND at mid frequencies (figure 6:30). Receivers 3 and 8 gave errors at 250Hz and 500Hz octave bands slightly higher than the overall average of up to 1.9JND.

CATT EDT predictions were less accurate than Odeon as seen in figures 6:30 and 6:32 which shows CATT generally overestimating and giving errors averaged over the receiver positions of up to 2.1JND when observed across the octave bands. Receiver 1 and 5 produced the highest overestimations of up to 3JND between 250Hz and 2000Hz octave bands (figure 6:30 and appendix E.5). EDT prediction discrepancy between the two programs averaged over the test positions again decreased with increasing frequency, the highest was 1.9JND at 125Hz octave band (figure 6:33 and appendix E.5). The high standard deviation of this average values resulted from the significant discrepancies at receivers 1 and 5 (figure 6:30 and appendix E.5).

Overall, EDT prediction errors produced by both programs were higher than compared to errors produced in previous case studies. However the range of EDT prediction errors obtained in this space are not deemed significantly high considering that 1JND is optimal accuracy and comparable errors are reported in similar studies (Bork, 2000, 2005b), (San Martin & Arana, 2006), (Bradley & Wang, 2007)[27], (Hodgson et al,2008), (Segura et al, 2011).

RT prediction error analysis for both programs followed similar pattern as EDT analysis, although CATT errors were comparatively higher. Both programs EDT and RT prediction curves matched the shape of those measured which indicates an overall ability of the programs to represent the sound field (figure 6:30).

In figure 6:30 it can be seen Odeon predicting accurately RT at all receivers and across the frequency bands showing spatially averaged errors within 1JND observed across frequency bands (figure 6:32). However, CATT significantly and consistently over predicted RT for all receiver positions and at frequency bands between 125Hz and 2000Hz. In contrast, RT predictions by both programs at 4KHz and 8kHz were found to agree closely (discrepancy below 1JND) showing each program errors below 1JND when averaged over the receiver positions (figures 6:30, 6:32 and 6:33). CATT RT prediction errors were up to 6.5JND and up to 4.7JND when averaged over all the receiver positions observed between 125Hz and 2000Hz frequency bands. As a consequence of the significant CATT RT errors, predictions discrepancies between programs were consistently high over the receiver positions for the 125Hz to 2000Hz octave bands (up to 3.8JND averaged over the receivers positions) (figure 6:33).

The EDT and especially RT prediction errors produced by CATT at low and mid frequencies were not consistent with results obtained from the two previous cases of straight platforms. The CATT error pattern was of the similar order of magnitude and consistent for all the receiver positions. These indications suggest a systematic error attributed to inaccurate tracing and calculation processing of late reflections caused by the sharp platform curvature.

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[27] In this work the authors considered RT_{15} prediction error of up to 2 JND as “relatively good agreement” based on JND=0.15sec

A pattern showing higher prediction error as frequency decreases is naturally to be expected in a simulation by geometrical acoustics of curved surface approximated by a finite number of planes. However, the significant prediction error produced by CATT suggests that its ray tracing (related to late reflections) and calculation algorithms could not handle accurately the geometrical approximation of the platform curvature[28]. This explanation is further suggested by the much higher prediction errors produced for RT (late reflections dependent) than in EDT (late reflections independent) (figure 6:30 and 6:32).The pronounced curvature of the platform also prevented receivers from receiving line of sight and therefore direct sound from a number of distant loudspeakers[29].

The following observations further support the above argument of CATT prediction inaccuracy due to platform curvature.

CATT energy decay curves determining EDT and RT values, appeared straight at high frequencies (4 kHz and 8 kHz) where prediction error and discrepancy with Odeon was insignificant. However at mid and low frequencies, where discrepancy and prediction errors were significant, CATT showed decay curves of uneven shape (figure 6:34).

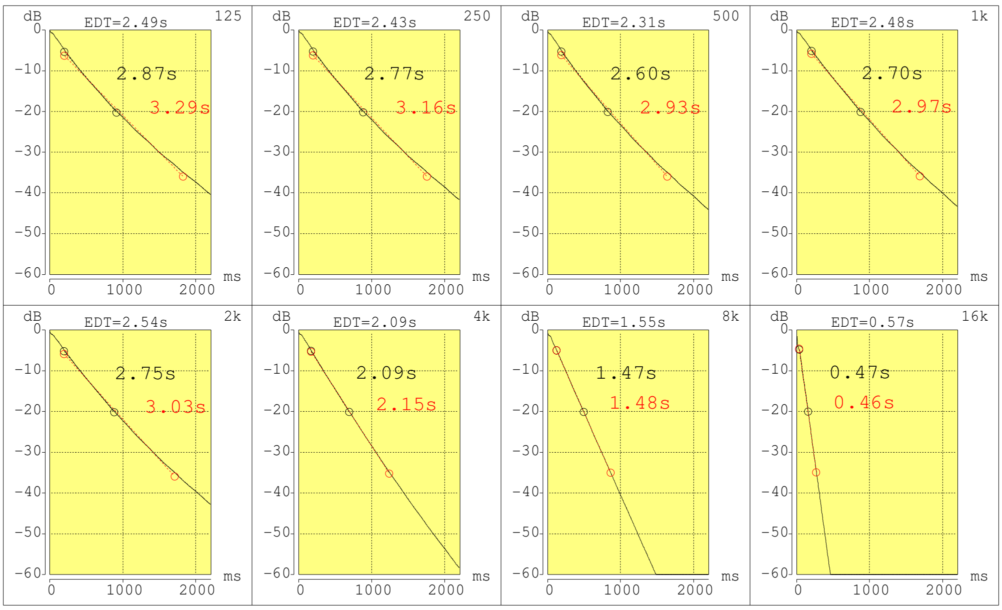

Figure 6:34 CATT decay curves at receiver position 2 with all sources active

[28] Platform curvature approximated by 12 sections of 13 planes (figure 6:29)
[29] Direct sound received from distant sources effectively acts as a late arrival similar to a late reflection.

The effect of curvature on CATT predicted EDT and RT values can be seen more in detail by observing the decay curves of two individual source-receiver pairs differing in the grade of curvature due to their separation (figures 6:34, 6:35 and 6:36).

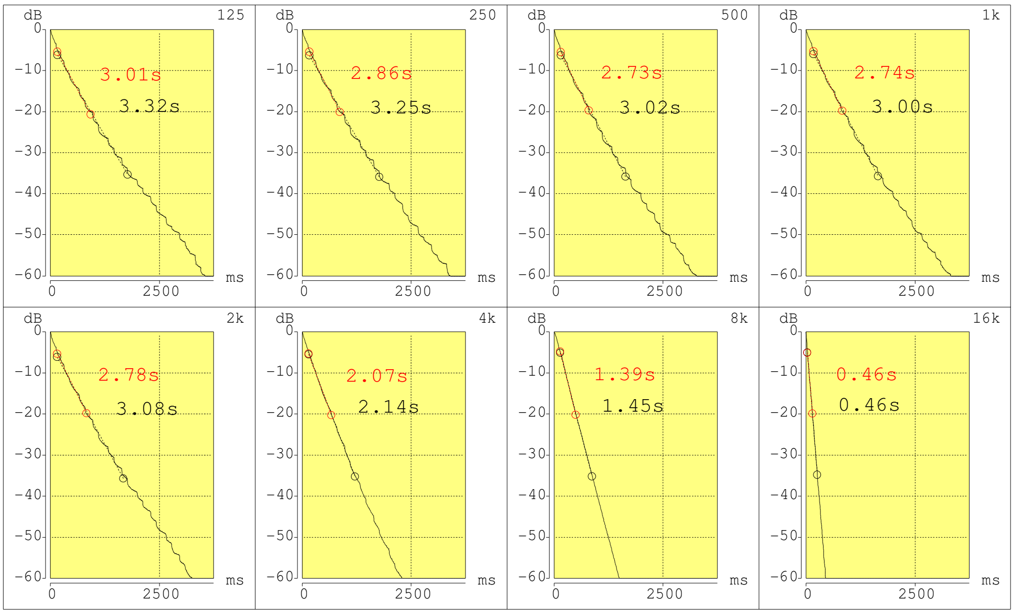

Figure 6:35 CATT predicted decay curves from source A4 and receiver 2 pair separated approx by 35m (low grade of curvature between source-receiver)

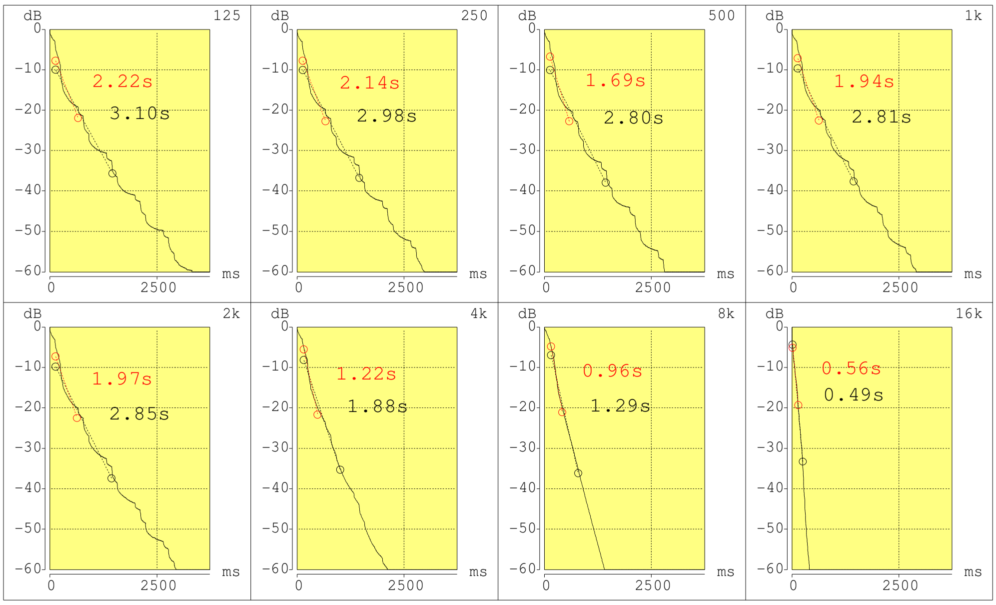

Figure 6:36 CATT predicted decay curves from source A4 and receiver 8 pair separated by approx 70m (high grade of curvature between source and receiver)

Figure 6:37 illustrates the linearity dissimilarity in the decay curves between the two programs for the same high grade curvature source receiver pair. Odeon decay curves appeared straight at all frequencies while CATT curves were straight only at high frequencies.

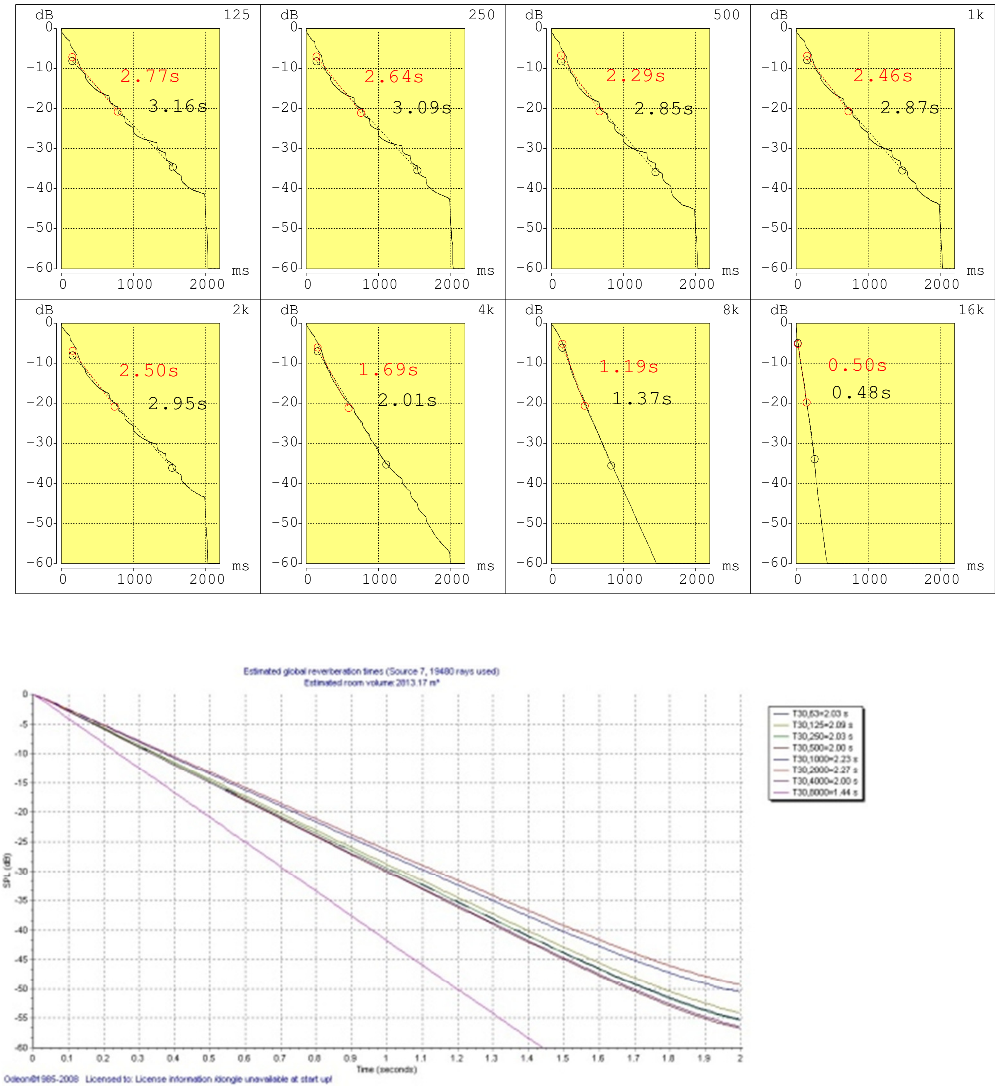

Figure 6:37 CATT (above) and Odeon (below) energy decay curves at receiver 8 (only source A4 (7) active)

Again, SPL discrepancies averaged over the five receiver locations typically decreased with frequency with the highest discrepancy being of 1JND at 125Hz (figure 6:33 and table 6:16). Odeon yielded A-weighted SPL predictions at all receivers' positions consistently within 2dB (1JND) below the 90dBA target while CATT produced also consistent A-weighted levels although within 3dB (1.5JND) below the target (table 6:16 and Appendix E.5). The good SPL accord obtained between programs can be expected despite the platform curvature, since the SPL calculations are largely reliant on direct sound and early reflections from the nearest sources for which both programs use mostly image-source techniques.

Prediction accuracy by the two programs in terms of STI was slightly lower than for the previous platform cases. Odeon under predicted at the five receivers showing errors between 1 and 2.2 JND. Despite higher RT prediction error, CATT gave a similar STI error range of 1 and 2JND (figure 6:31 and table 6:17). This low sensitivity of the calculated STI to variations in RT has been noticed in cases previously analysed and is acknowledged in the literature. However larger STI prediction errors and discrepancies would have been observed if EDT predictions had been more erroneous since STI is more closely correlated to EDT (see chapter 2). The STI prediction errors shown by both program for this complex space are not deemed excessively high considering that error up to 1JND is considered excellent accuracy. Moreover the STI range of errors obtained in this simulation case is comparable to STI prediction errors obtained in other similar works (Field & Shimada, 2005), (Christensen et al, 2008b), (Segura et al, 2011).

Calculations of the volume, MFP and total internal surface area by both programs showed close agreement (table 6:15). The processing time taken by CATT to run the simulation was 27 times longer than Odeon. CATT projected approximately 13 times more rays than Odeon. Odeon efficiently detected the more complex geometry of Plat-3 Odeon and projected 1.5 times more rays than for Plat-1 simulation case, while CATT emitted approximately the same number of rays in both cases.

The overall prediction error calculated in JND units for Odeon and CATT were 1.01 and 1.44 respectively. These values are slightly higher than previous case studies. The curved geometry of this platform had a degree of negative impact in the prediction algorithms accuracy for both programs. However, overall and excepting for CATT RT predictions, both programs were able to predict with acceptable accuracy the evaluating parameters in this challenging platform geometry.

6.3.7 Conclusions

The evaluation performed in this chapter has provided evidence of the validity of both acoustic simulation programs for the prediction of acoustic and intelligibility parameters in key spaces of underground stations equipped with VA systems. The overall prediction error performance shown by both programs in terms of the overall prediction error was generally within one unit of just noticeable difference and this result was very similar between programs (figure 6:38).

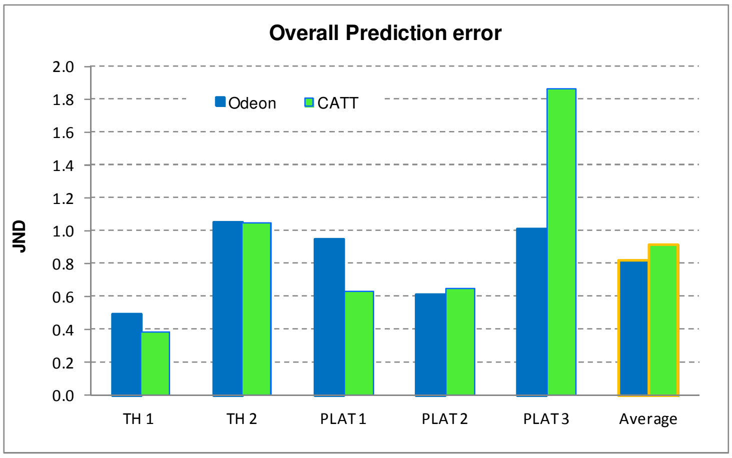

Figure 6:38 overall prediction error calculated for each case study and average.

Odeon appeared more sensitive and efficient in the adaptive assignment of number of rays to detected geometrical complexity. This quality was the main reason for significant shorter processing times of Odeon with respect CATT for the five cases (figure 6:39).

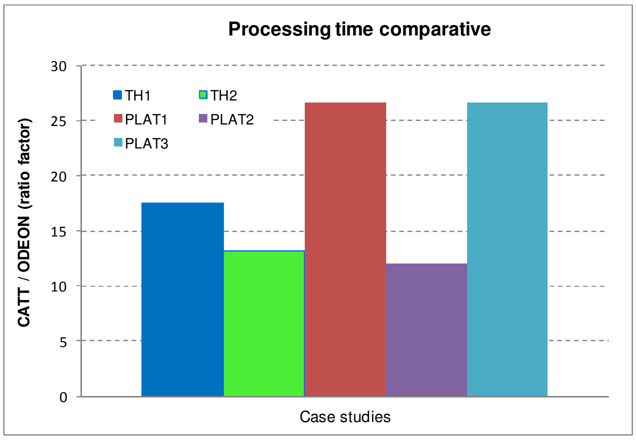

Figure 6:39 Processing time ratio comparison for the five cases

Prediction errors observed on EDT, RT and STI parameters shown by both programs were within or around the target reference of 1JND for all case studies, except for the most challenging geometry (Plat-3) where higher errors were seen for STI and RT parameters (figure 6:38). The sharp curvature of Plat-3 had an impact in the programs prediction quality being more notable in CATT.

Overall results revealed that the two modeling programs in the five spaces under test generally predicted very similar results for EDT, SPL and STI as well as for volume, total surface area and MFP calculations. Prediction discrepancies between the two programs were generally around the target reference of 1 JND. Higher discrepancies were observed generally for RT in the most extreme case geometries. Both programs provided very similar performance in the calculation of number of lost rays, detection of modeling bugs and warped surfaces.

In terms of usability, it was clearly perceived by the author that the Odeon program offered a more intuitive interface and better graphical presentation which led to higher modeling efficiency with respect to CATT. In particular it was observed that Odeon possessed a more satisfactory accessibility to inputting data, adjusting operational settings, and retrieving results than CATT. Other subjective qualities such as, flexibility and versatility of use, interface ergonomics and ease of learning were subjectively compared giving Odeon a higher score. Other features such as user error warnings, de-bugging processes, additional tools, manual and online support, were perceived of similar standard. Overall, Odeon showed superior usability qualities, which can lead to less human errors and improved and efficient delivery of the work in hand.

In summary, both programs demonstrated their ability to predict relevant parameters used in VA electro acoustic design of underground stations with similar satisfactory accuracy.

Chapter 7 Evaluation of VA design variables and their impact on performance

7.1 Introduction

The acoustic transmission between the loudspeakers and the room has a vital contribution to the final acoustic performance of VA systems on platforms (see chapter 3). Close attention must be paid to these two elements at the design stage where adjustments and an informed selection process should be applied to attain a desired performance.

Due to cost, operational and practical considerations the loudspeaker is normally the only element of the electro acoustic transmission section available as controllable design variable (see chapter 3). However knowledge of other non-controllable influencing factors can assist in attaining the highest possible performance within the limiting constraints

This chapter identifies, evaluates and quantifies the impact on the acoustic and predicted speech intelligibility of the most significant variables and design factors in the last section of those VA systems. The main focus is given to aspects and parameters related to speech transmission quality.

Chapters 4, 5 and 6 have shown that commercial computer simulation programs are the most suitable and reliable prediction tool for the design of deep platform VA systems under real world conditions. Although these programs are used by the industry concerned, no research has been found on the acoustic characterization or design guidelines for VA system on underground platforms based on or derived from commercial simulation programs.

This chapter attempts to fill this knowledge gap by producing reliable description of the platform sound field and providing validated electro-acoustic design guidance utilizing CATT Acoustics program for real world platform VA system implementations.

Iterative investigations were carried out on a validated representative platform simulation to establish and evaluate influential design variables on the acoustic performance of platforms VA systems. Some of these findings will be utilised in the development of the empirical model, see chapter 5.

The knowledge produced in this chapter is expected to enable the design of renovated or new platform VA systems to achieve or surpass the challenging speech transmission quality performance target under some of the usual practical constraints .

Below is a set of variables or factors investigated considered to be the most influential in the acoustic transmission section. The variables and their effect on performance will be investigated in a prioritized order as shown below and illustrated in figure 7:1.

- Loudspeakers layout
- Air temperature and relative humidity
- Scattering coefficient
- Acoustic absorption

A novel and efficient acoustic treatment design concept is presented.

7.2 Methodology

This section describes the general methodology employed. The specific methodologies for each variable investigation are presented separately in their corresponding subsection.

7.2.1 Investigation strategy

Most of the investigations of this chapter are undertaken following a cumulative, validating and optimization order. In this process findings from previous investigations are applied as optimum and then used in the following investigations. This strategy was followed to focus down on an optimized design where all the variables and factors influencing the performance have taken progressively taken into consideration.

The overall chapter strategy is illustrated in figure 7:1 below.

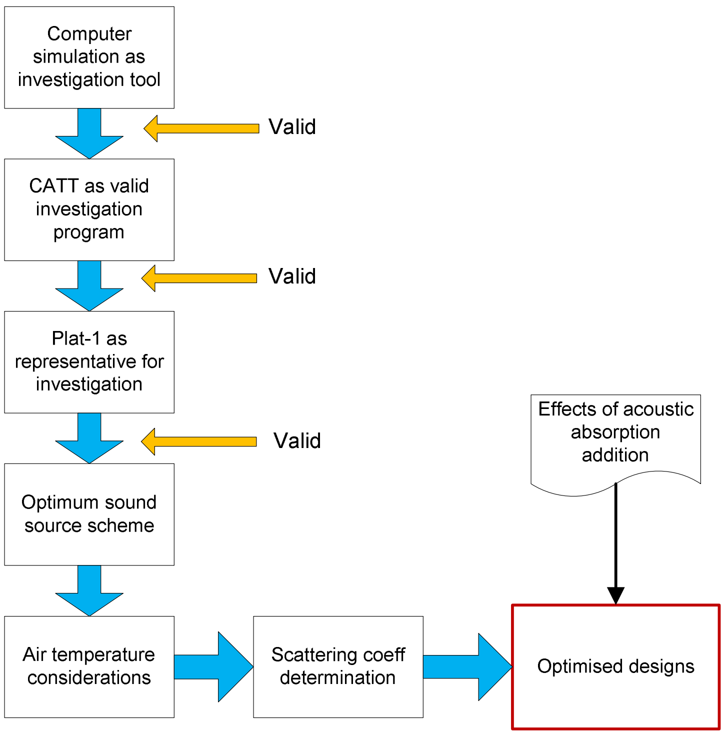

Figure 7:1 Flow chart of chapter structure

7.2.2 Use of reference platform suitability and validity

The representative platform 1 prediction model, as presented and validated in chapter 6, was chosen as the working reference platform model on which most of the investigations of this chapter were conducted and will be referred hereafter as Plat-1 original (figure 7:2). Simulation results of Plat-1 original are referred in graphs and tables as “original” and utilized as reference to assess the impact of design factors and variables on performance. Findings from the investigations in this chapter are contrasted and crossed validated with findings drawn from previous chapters where measured data was utilized, using the same methodology as given in Chapter 6 with the addition of D50 analysis.

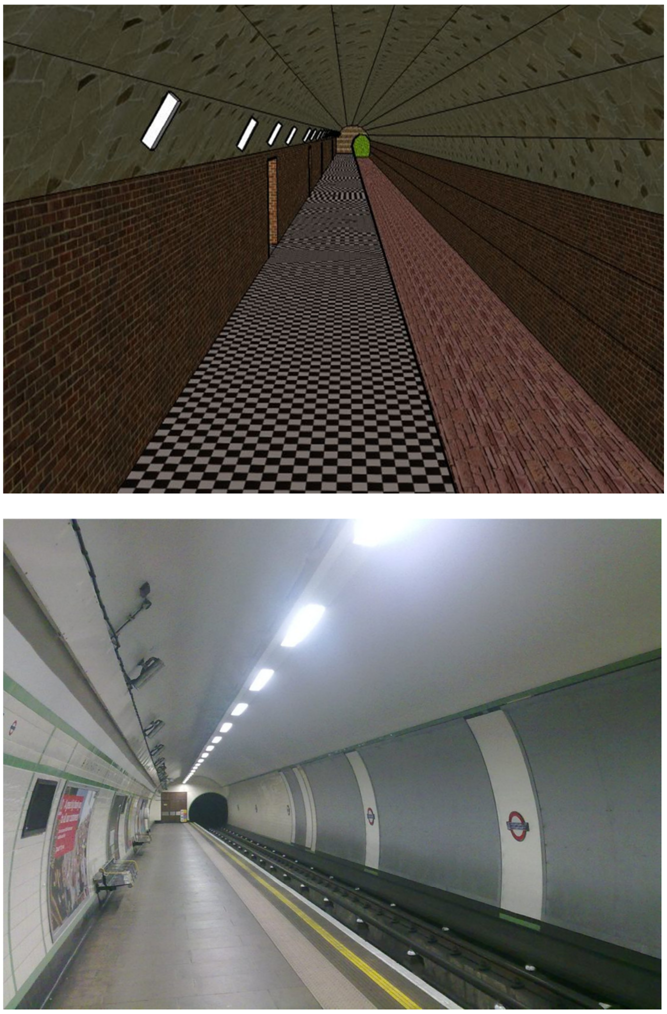

Figure 7:2 Sketch-Up and photo views of Plat-1 original

7.3 Design variables investigation

This section presents the investigations on different variables and factors influencing acoustical and speech related performance. The effects and influence on performance of each of the variables listed is examined separately in the following sub-sections.

7.3.1 Effects of different loudspeaker configurations

The effect of different loudspeaker array configurations on the evaluating parameters is investigated in this section.

7.3.1.1 Introduction

The loudspeaker configuration is the most modifiable element of the acoustic transmission section and the only possible variable when other influencing factors are fixed. Most of the significant factors related to this critical element in the VA system chain are examined here, including loudspeaker type, spacing, positioning and aim. The impact of factors iterations on several evaluating parameters is also investigated to establish the best performing loudspeaker configurations in terms of predicted STI.

7.3.1.2 Methodology

The approach taken for the consecutive investigations was to vary sequentially the loudspeaker characteristics and configurations as modeled on the Plat-1 original simulation while maintaining the rest of the simulation reference settings and input data.

The first factor considered for these investigations was the spacing between contiguous loudspeakers in the array. Three spacing intervals (providing three different loudspeaker densities) were investigated: 3 meters, 5 meters and 7 meters. The interval ranges selected above are in line with spacing recommended or used as by other authors for underground platforms (London Underground, 2004), (Harrison,2001), (Yang et al, 1996), (Orlowski, 1994). The next factor studied was the influence of type the of loudspeaker in terms of its directionality.

The two commercially available loudspeakers types used in this investigation are the column and the projector (aka “canister projector”) (figure 7:3). The short column consists of two identical mid range drivers arranged at both ends of the metallic enclosure with a high frequency tweeter placed in between (figure 7.3 and appendix F.1 and F.2). The projector loudspeaker consists of a single driver mounted in a cylindrical metallic enclosure. These two loudspeaker types are the most widely used on London underground subsurface platforms.

Figure 7:3 Column and canister projector type of loudspeakers widely employed on London underground subsurface platforms.

A third loudspeaker type (not normally installed on platforms), was employed in a particular simulation to show the effect of highly directional loudspeakers (Figure 7:4 and appendix F.4).

Figure 7:4 Highly directional projector type of loudspeaker.

Once the optimum spacing and type of loudspeaker was established, the aim and positioning of the loudspeakers was the third factor investigated. The selection of loudspeaker position on underground platforms is mainly constrained by practical requirements such as maintenance accessibility and vandal proof minimum safe height (2.3m) (*London Underground,2004*). Taking into consideration these two main constraints , three potential positional configurations were investigated. The first one consisted of placing the array of loudspeakers on the passenger platform wall (platform wall) at 3m above the platform floor and aiming them at the edge of the platform to obtain maximum coverage on the listeners' area (figures 7:5 and 7:6). This configuration is used by Plat-1 original and is the most commonly found on underground platforms. This is due to the convenient access for installation and maintenance of loudspeakers and because the close proximity of loudspeakers to the cable management system (CMS) route (figure 7:6). This configuration will be referred hereafter as "Project on wall", "Column on wall" and "HighProject on wall" (higher directivity loudspeaker type)" depending on the type of loudspeaker used.

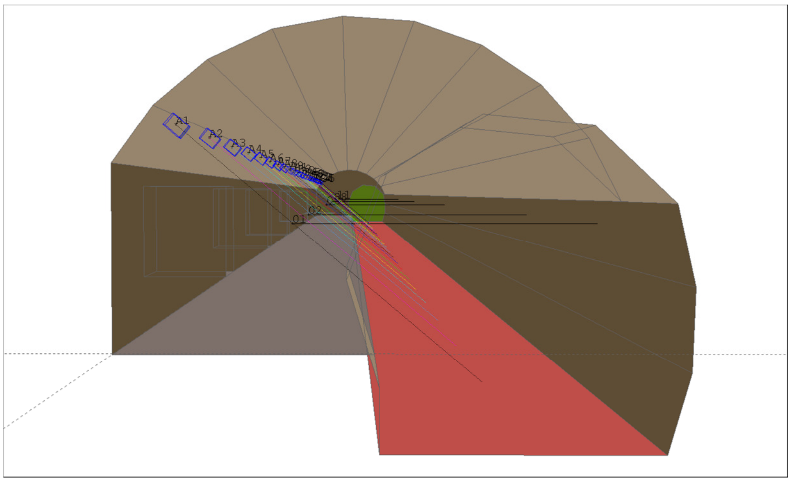

Figure 7:5 CATT view of project on wall configuration. Loudspeakers (in blue) show aim lines. Receivers (in black) above platform floor (receiver aim lines are void).

Figure 7:6 Platform during maintenance works showing CMS and projector on wall configuration.

The second positional configuration option consisted of placing the array of loudspeakers on the ceiling at 3.9m above the platform floor aiming perpendicularly downwards onto the listener plane. This configuration used only the projector loudspeaker and will be referred as “Project on ceiling” (figures 7:7 and 7:8, table 7:1).

The third positional configuration was equivalent to the “projector on ceiling” except for the height of loudspeakers which was set closer to the listening plane at distance of 2.5m from the platform floor. This configuration is referred as “Project low down” (or “Project low”), (figures 7:7 and 7:8, table 7:1).

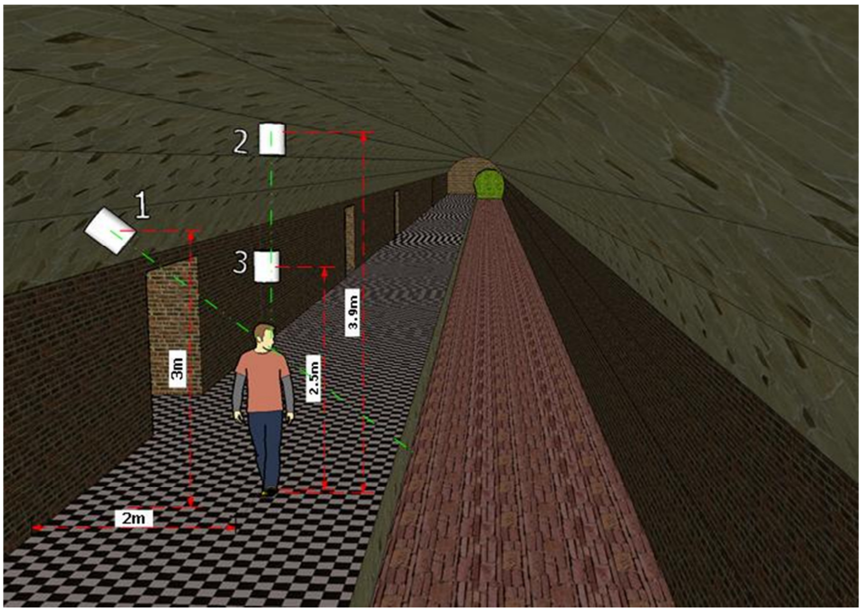

Figure 7:7 Plat-1 An array loudspeaker showing three different positional configurations and aims: “on wall” (1), “on ceiling” (2) and “low” (3).

The seven loudspeaker configuration arrangements studied in this section are summarized in table 7:1. Height is measured from the passenger platform floor.

Description	Type	Location	Height (m)	Aim	Spacing (m)	Speaker quantity	Notes
Original	Column CM20	On wall	3	Edge platform	5	23	As modelled in Plat-1 Chapter 6
Project on wall	Projector PAC15	On wall	3	Edge platform	5	23	Position as Fig7:7 (1)
HighProject on wall	Projector MHS20	On wall	3	Edge platform	5	23	Position as Fig7:7 (1)
Project on ceiling	Projector PAC15	On ceiling	3.9	0º down	5	23	Position as Fig7:7 (2)
Project low	Projector PAC15	Suspended from ceiling	2.5	0º down	5	23	Position as Fig7:7 (3)
Column at 3m	Column CM20	On wall	3	Edge platform	3	38	Position as Fig7:7 (1)
Column at 7m	Column CM20	On wall	3	Edge platform	7	17	Position as Fig7:7 (1)

Table 7:1 Summary of details of seven loudspeaker array configurations

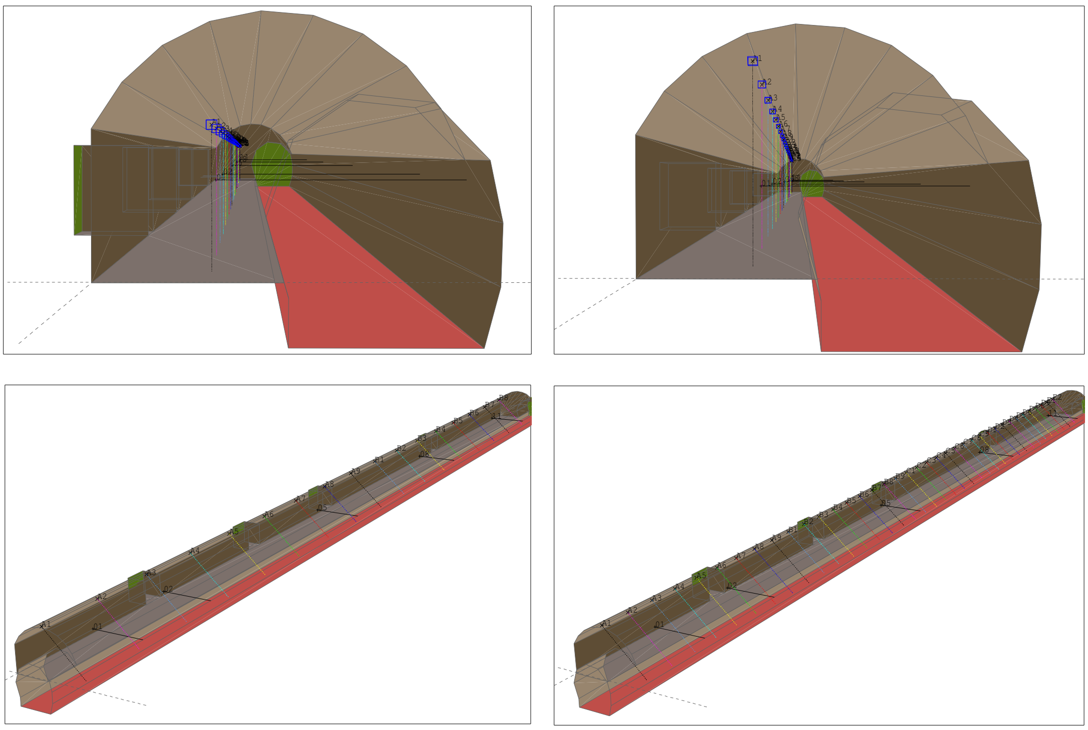

Figure 7:8 From top left to bottom right, CATT views of project low, project on ceiling, and column at 7m and 3m spacing. Aim lines in colours. Receivers (in black) show void aim lines.

7.3.1.3 Results

The prediction results for the evaluating parameters averaged over the receiver positions for the seven loudspeaker configuration simulations are presented in graphical form below. Tabulated data and standard deviation of parameters are presented in appendix F.5.

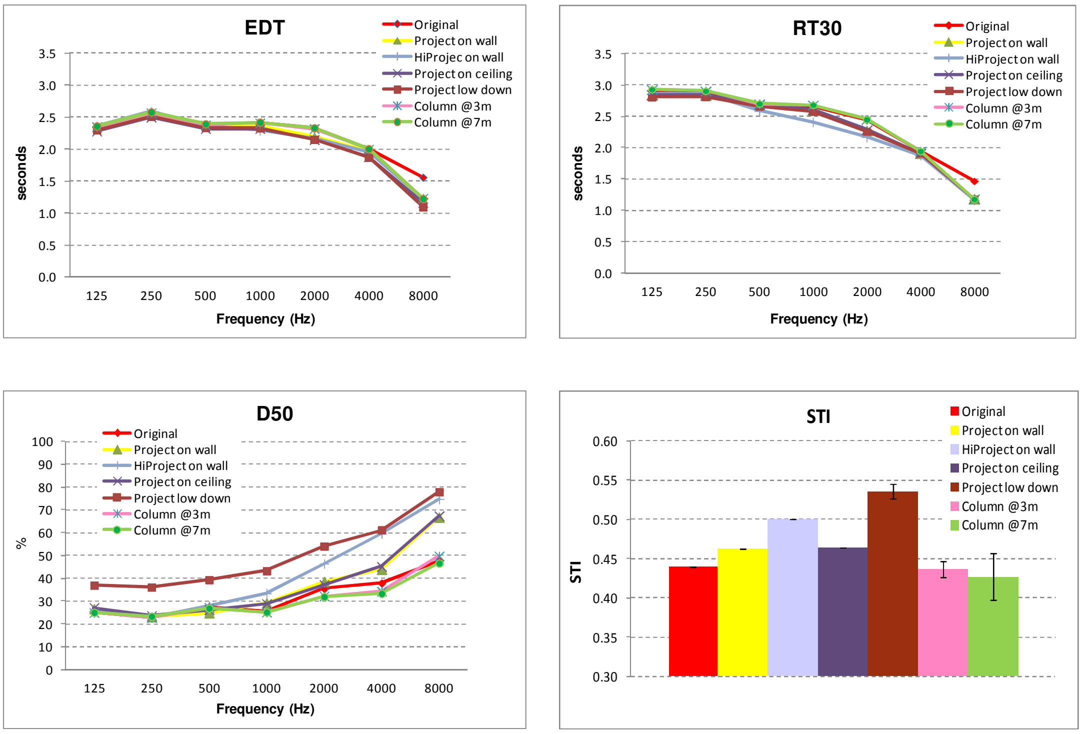

Figure 7:9 Seven different loudspeaker configurations prediction results. Black bars in STI graph are standard deviation

7.3.1.4 Analysis and discussion

The spatial variation of all predicted parameters for all distributed array configurations is remarkably low at all frequency bands as indicated by the standard deviation (σ) (generally below 0.5JND) in appendix F:5. This spatial homogeneity of predicted results along the platform length was also observed in actual measurements (chapter 5) and is consistent with results from relevant works found in the literature (Barnett,1994b), (Orlowski,1994), (Kang, 1996d,1998), (Shield & Yang,1999b), (Yang, 1997).

From the EDT and RT results (figure 7:9) it becomes evident that different configurations of a distributed loudspeaker system involving different loudspeaker position, aim and density do not affect the reverberation in the platform space. This finding is consistent with the results obtained in a similar investigation in a large live room (James, 2003) where it was concluded that the measured RT in large live room was not affected by the type or directivity of the loudspeaker.

The sound field's low sensitivity to type and directivity of sound source is a characteristic indicating the presence of diffuse field conditions. The low EDT and RT spatial variability (σ) as well as the EDT/RT ration close to 1 at mid and high frequency bands (table 7:2) for all configurations is additional evidence that indicates that the sound field of Plat-1 excited by a evenly distributed loudspeaker system exhibit diffuse sound field characteristics.

		f (Hz)						
		125	250	500	1000	2000	4000	8000
Max	EDT std	0.13	0.13	0.13	0.13	0.13	0.10	0.06
	RT std	0.06	0.15	0.14	0.10	0.05	0.04	0.05
	EDT/RT	0.83	0.91	0.91	0.95	1.00	1.04	1.05
Min	EDT std	0.11	0.12	0.11	0.12	0.09	0.06	0.02
	RT std	0.06	0.05	0.06	0.05	0.05	0.03	0.02
	EDT/RT	0.80	0.88	0.86	0.89	0.94	0.98	0.93

Table 7:2 Seven configurations' maximum and minimum values of EDT and RT std values and EDT/RT ratio from five receivers' results.

These observed diffuse sound field indications are in agreement with other authors' measurements and indications obtained in similar conditions (Barnett,1994b), (Orlowski,1994), (Kang, 1996d), (Shield & Yang,1999b), (Yang, 1997). This important diffuse field characteristic observed on platforms is further examined in detailed in chapter 8.

D50 results showed that the type of loudspeaker and specially their position relative to the receiver have major effects on the D50 parameter. The "Project low down" (also "Project Low") and "High project on wall" (also "Hi Project") configurations provided the highest D50 values at all frequency bands when compared to any of the other configuration (figure 7:9). The use of the Projector and High-projector types showed a D50 increase for all octaves bands above 1000Hz when compared to similar loudspeaker-receiver distance configurations using the column type (figures 7:9 and 7:10). This improvement in D50 appeared more significantly towards the high frequencies and it was due to higher loudspeaker directivity characteristics at those frequencies. This high directivity concentrates sound energy onto the listening area spilling less energy into the reverberant field consequently increasing the speech related parameters D50 and STI.

Figure 7:10 shows examples of relative D50 differences in terms of JND between configurations. It can be seen that “Project low down” is the most effective configuration at achieving higher D50 values at all frequency bands when compared to conventional configuration (“Original”) and “Projector on ceiling”. “High Project on wall” also achieved higher D50 values respect to the same conventional configurations although only at mid and high frequencies.

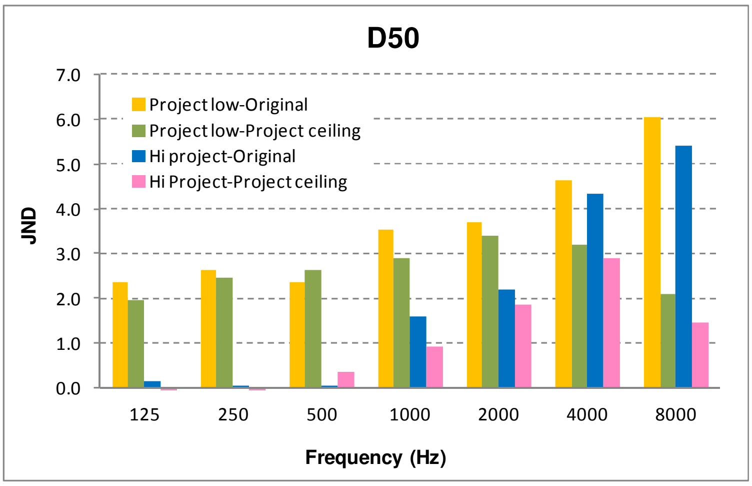

Figure 7:10 Examples of D50 relative differential values between effective and conventional configurations (“Hi project” refers to “High Project on wall”)

The highest D50 values were obtained for the closer source-receiver configuration (1.4m closer than the other five configurations) as a result of the increase of the direct and early reflections energy at the receivers and reduced spill into the reverberation field (figure 7:10).

A very important side benefit of highly directional loudspeakers is that they possess high electro-acoustical sensitivities which results in a reduced electrical power requirement from the associated amplifier, a major practical benefit. In the prediction simulations it was observed that replacing the column loudspeakers by the high directional projectors, would lead to reduce the amplifier gain (*Ampl gain*) approximately by 6dB to achieve the same 90dBA SPL of direct sound at the receivers. This gain saving translates in that four times less amplifier power is required (equation 7:1).

$$Ampl\ Gain = 10log_{10}\frac{W_2}{W_1}$$

$$-6dB = 10log_{10}\frac{x * W_1}{W_1}\ ; \qquad x = 0.25 = 1/4$$

Equation 7:1 Required amplifier power reduction by using highly directional projectors instead of column loudspeakers

Where W1 is the initial amplifier power

This type of highly directional loudspeaker cannot reproduce low frequency sounds, in this case below the 500Hz octave band, (appendix F.4). Due to the low contribution of these frequencies to speech intelligibility (see chapter 2) this frequency limitation has relatively little impact on the expected speech intelligibility (STI). The reverberation time on deep platforms is the longest at low frequencies (see chapter 5) and the most difficult to control. Limiting the low frequency reproduction on platforms has the beneficial effect of avoiding upward masking of vital higher frequencies in speech intelligibility. This was partially reflected on the higher predicted STI values from the “HighProject” configuration than values from other configurations using wide frequency response excepting for “Project low” configuration (figure 7:10). “HighProject” configuration achieved the STI performance target showing the potential of utilizing loudspeaker of high directionality or narrow dispersion.

A disadvantage of the limited low frequency reproduction characteristic will involve the perceived degradation in the naturalness and quality of the reproduced speech. Moreover, the narrower dispersion of highly directional loudspeakers compared with broadband frequency types will normally require more sources in the design to attain a given coverage uniformity. Therefore the choice for this type of loudspeakers requires a careful balanced judgment on the benefits and disadvantages in attempting to achieve all the performance specifications.

From the simulation results (figure 7:9) it was observed that increasing or decreasing the loudspeaker density (within the practical range examined) had an insignificant effect on D50 and STI. This finding agrees closely with similar work by Shield & Yang (1999b) where they found that increasing the number of sources on an underground platform array has little effect on speech intelligibility related parameters when background noise level is significantly low. This result implies that the reverberant field still dominates over the direct field even when the loudspeaker density is increased. This in turn suggests that equivalent approaches such as increasing the power from the sources will not increase the expected intelligibility.

It was also noticed that the configurations providing higher D50 values also scored higher STI values, (figure 7:9). This observation closely agrees with the high correlation between STI and D50[30] in noiseless conditions reported by other researchers (Bradley,1998),(Mapp,2002a).

“Project low” configuration scored an average of 2JND of STI and 3.6JND of D50 higher than “Original”. “Project low” configuration demonstrated that significantly reducing the loudspeaker-receiver distance is the most effective approach to achieve or exceed the challenging 0.5 STI performance target without using reverberation control (acoustic absorption) or any other critical architectural modifications (figure 7:9). This ideal configuration however, may encounter constraints in certain practical implementations regarding maintenance, aesthetics and vandalism prevention.

A very important side benefit of the “Project low” configuration approach, is the reduced output power needed from the amplifier to achieve 90dBA of direct sound at a receiver. Since the distance loudspeaker - receivers of this configuration is 1.5m - 1.4m shorter than conventional configurations “project on-wall” or “project on-ceiling” (figures 7:7 and 7:11), the same amplifier needs to deliver 6.3 times less power to achieve the required 90dBA at the receiver. Therefore the “Projector low” configuration would provide a significant saving in amplifier power with respect to conventional loudspeaker configurations fixed to the platform wall. This translates to high cost savings in the operation and provision of the VA system equipment.

[30] (Bradley,1998) and (Mapp,2002a) reported STI high correlations with C50 parameter; however D50 is an equivalent parameter expressing the same concept (Dalenback,2007) which can be simply converted from C50 (see chapter 2)

Following is a brief numerical demonstration illustrating the estimated power saving of the “projectors low” configuration.

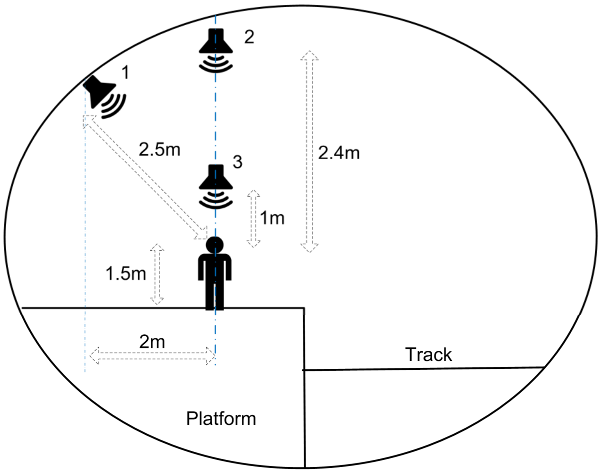

Figure 7:11 Three loudspeaker- receiver distance configurations (1,2,3) as per figure 7:7.

The following calculation is based on the following initial assumptions: Spherical and free field propagation, single loudspeaker source sounding over one on axis receiver. If the loudspeaker output power tapping is maintained but the loudspeaker-receiver distance was reduced from 2.5m to 1m as a result of changing the conventional “projector on wall”, (configuration-1) to “projector low”, (configuration - 3); then a 8dB SPL increase would be approximately expected at the receiver resulting a total 90 + 8 = 98dB at the receiver (equation 7:2).

$$SPL_{d1} - SPL_{d2} = 20log_{10}\frac{d_2}{d_1}$$

$$SPL_{2.5m} - SPL_{1m} = 90 - x = 20log_{10}\frac{1}{2.5} \Rightarrow SPL_{1m} = 98dB$$

Equation 7:2 Expected SPL increase at the receiver by using highly directional projectors instead of column type on the wall

Where d_1= conventional configuration source-receiver distance in meters; d_2= project low configuration source-receiver distance in meters.

In order to attenuate this SPL excess of 8dB and achieve the target SPL at the receiver of 90dBA, the power fed to the "Projectors low" loudspeakers must be reduced by 6.3 times (equation 7:3).

$$SPL_1 - SPL_2 = 10log_{10}\frac{W_2}{W_1}$$

$$90 - 98 = 10log_{10}\frac{x * W_1}{W_1} \; ; \qquad x = 0.158 = 1/6.3$$

Equation 7:3 Amplifier electrical output power reduction by using projector low configuration instead of project conventional configurations

The high reduction in the required output power by using the "Projector low" configuration would translate in the specification of much less powerful and inexpensive amplifiers and cabling to achieve the same target SPL.

"High Project on wall" scored an average of 0.06 of STI (1.2 JND) higher than "Original" showing the important contribution of loudspeaker directivity to achieve higher speech intelligibility predictions in very reverberant spaces.

"Projector on the wall" and "projector on the ceiling" scored an average of 0.02 STI higher than "Original". This difference corresponded to the higher D50 values of those two configurations when compared with the original configurations incorporating columns loudspeakers. Although this STI difference is a marginal improvement (<1JND), it provided a further indication of the trend of the projector configurations providing higher values of intelligibility-related parameters than the columns configurations under the same or less favorable conditions.

STI spatial variability was negligible for all configurations with the maximum observed for "Column at 7m" at σ=0.03 (0.6 JND).

In the subsequent investigations, the "Original" configuration will be utilized as the standard working loudspeaker configuration. This will keep consistency with the design configuration employed on most of current and renovated London underground platforms and with the configuration presented in previous chapters.

7.3.2 Effect of air humidity and temperature variations

This section investigates the influence of air temperature and relative air humidity (RH) on room acoustic parameters.

7.3.2.1 Introduction

The sound absorption exhibited by air over relatively long distances is an important environmental factor which should be taken into account in the acoustic design of any large enclosure in which speech sounds are reproduced.

In the relevant literature most of researchers mention the air absorption consideration when developing or using mathematical models and computer simulations in large enclosures (Kang,1996e) ,(Yang & Shield, 2000), (Yang & Shield, 2001), (Kang & Orlowski, 2001), (Christensen, 2008a), (Dalenback, 2007), (San Martin & Arana,2006), (Wijngaarden & Verhave, 2006), (Sü & Çalışkan, 2007), (Segura et al, 2011). Only Kang (1996c) reported effects of air absorption on acoustic parameters (RT and EDT). However those results were obtained in a long enclosure by means of a computer simulation using only one sound source.

No study has been found in the literature on the effects of air absorption variation on VA acoustic performance on underground platforms as a function of temperature and humidity fluctuations.

The air sound attenuation processes vary strongly and irregularly depending on climate conditions (ISO 9613-2, 1996), (Wenmaekers et al, 2008). This makes the air attenuation a factor difficult to take into account in the design of VA on underground platforms. Most room acoustic simulation programs that predict acoustical parameters can take into account the air sound absorption as a function of temperature and RH data which is input by the user. This data is usually neglected in practical simulation designs of large spaces. On many occasions this data is loosely assumed to be constant throughout the months or year and/or to be same as the temperature and humidity program's default values. However actual platform climate variations from those assumed might significantly impact the accuracy of the simulation prediction.

Air absorption occurs when a proportion of the energy of a sound wave propagated through the air is converted to heat by means of shear viscosity and molecular relaxation losses. The resulting air attenuation becoming more significant as the frequency increases, these frequencies being the most relevant for speech intelligibility (range comprinsing centre octave bands of 2, 4 and 8 kHz). Hence for relatively long propagation distances, the air appears as a natural low-pass filter for sound propagation.

Sound absorption provided by the air, depends upon frequency, temperature, relative humidity, and atmospheric pressure. The latter has much lesser influence on air sound absorption fluctuations that the other three environmental parameters (ISO 9613-2:1996). Hence atmospheric pressure is assumed constant and of negligible influence on underground platforms acoustical predictions.

The estimated absorption area (A_{air}) provided by the air is calculated from the attenuation coefficient in air, *m*, in m^{-1} and the volume of air in the space, V in m^3, using equation 7:4 (Cox and D'Antonio, 2009).

$$A_{air} = 4mV$$

Equation 7:4 Estimated air absorption area in an enclosure.

Another parameter name associated to the same phenomenon as noted by Wenmaekers (2008) is the air attenuation per meter, *α,* which has been shown to be:

$$\alpha = k * f^2 + \alpha_2$$

Equation 7:5 Air attenuation per metre (α) where $k = 14.24 * 10^{-11}$, f = frequency in Hz, α_2 is humidity dependent (Smith,1996).

In equation 7:5 and figure 7:12 it can be realised the strong weight that sound frequency has in the air absorption process.

The coefficient *α* expressed in dB/m can be calculated and found in ISO 9613-1: 1993 (equation A:1).

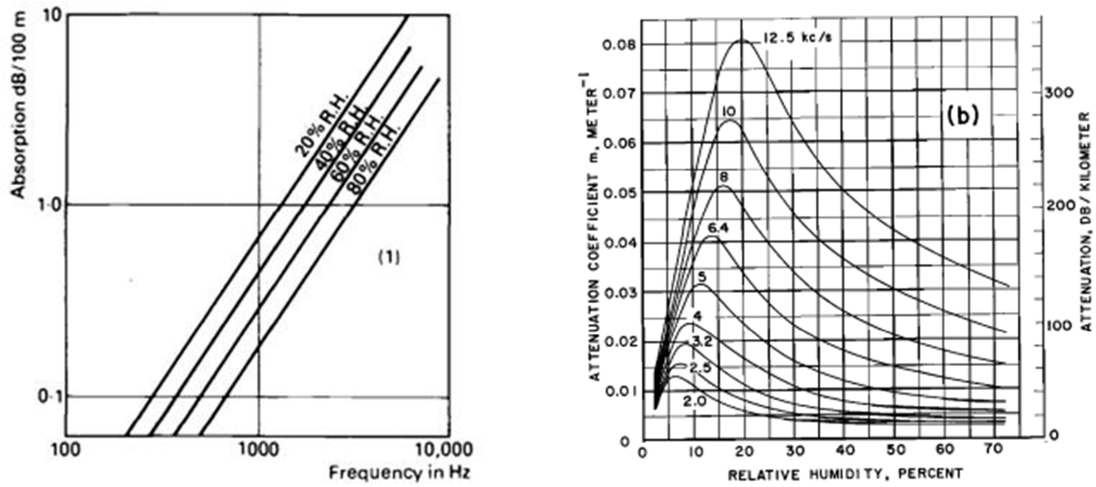

Figure 7:12 Air absorption or attenuation at 20ºC as a function of frequency (Hz or Kc/s)[31] and relative humidity (Smith,1996),(Harris,1963).

The speed of sound (c_o) at constant humidity is directly proportional to the square root of the temperature *(Cº) and* is calculated according to the equation 7:6 (Dalenback, 2007):

$$c_o = 331.4\sqrt{1+\frac{\vartheta}{273}}$$ where ϑ is the temperature in ºC

Equation 7:6 Speed of air (m/s) at constant humidity

One important effect of the sound absorption provided by the air in large enclosures is the consequential reduction of the reverberation time and the sound pressure level of the reverberant field. However this reduction becomes significant only at frequencies above 2 kHz (Howard, 2009).

The effect of air sound absorption is also determined by the distance the sound has travelled. Hence the air attenuation A in dB during propagation through a distance d in meters can also be calculated by the following equation 7:7, where α is the air atmospheric attenuation coefficient in dB/km for each octave band (ISO 9613-2 :1996)

[31] Kc/s refers to kilo cycles per second. This term is equivalent to kilo Hertz, abbreviated as kHz.

$$A = \alpha * d/1000$$

Equation 7:7 Air sound attenuation observed over distance

ISO 9613-1:1993 provides an extensive tabular database of outdoor air attenuation coefficient as a function of temperature and relative humidity. In figure 7:13 part of that database has been compiled and presented in curves for the octaves centre frequencies bands where attenuation can be significant and for relevant temperatures and relative humidity ranges.

The air sound attenuation in a large enclosed space is directly proportional to the reverberation time (RT) since the effect of air absorption is greater with a longer reflection distance. In this way the distance that sound travels in a large enclosure until it becomes imperceptible (60dB drop in SPL) can be expressed by the product of RT by Speed of sound (ANSI/ASA S1.26, 1995).

Neglecting the absorption of the air in design calculations can lead to the over-specification of surface materials absorption properties and the inaccurate prediction of acoustic and speech intelligibility related parameters.

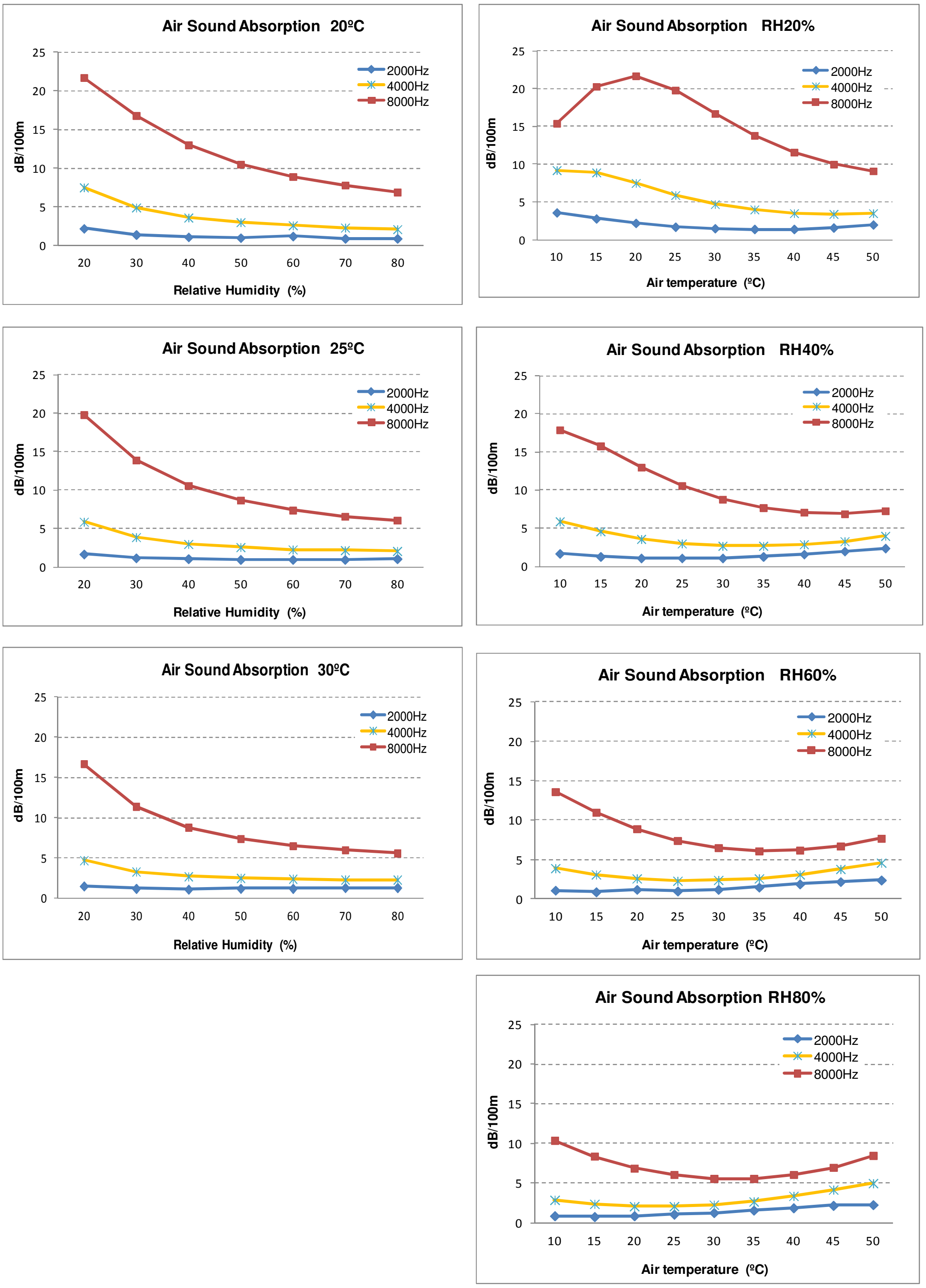

Figure 7:13 Air absorption coefficient at three octave band frequencies as function of air temperature (right column) and RH (left column) and, from ISO 9613-1:1993

CATT-Acoustics allows the user to choose among three options for the determination of the air sound absorption which can be used in the calculation of acoustical parameters. The three options are: a) influence of the air ignored; b) estimation of the air absorption from air humidity and temperature according to ISO 9613-1:1993; c) user defined by specifying the energy dissipation coefficient.

The user can enter the air specific parameters' data comprising temperature, humidity and density which are used for the estimations mentioned above. The program's default air parameter values are temperature 20ºC, relative humidity 50% and air density 1.2kg/m³.

The large volume and long length of London underground platforms (typically of 3000m³ and 120m respectively), makes this type of space susceptible to exhibit significant air sound absorption. Figures 7:14 and 7:15 show an example of outside temperature and RH observed at London (Heathrow) during one year.

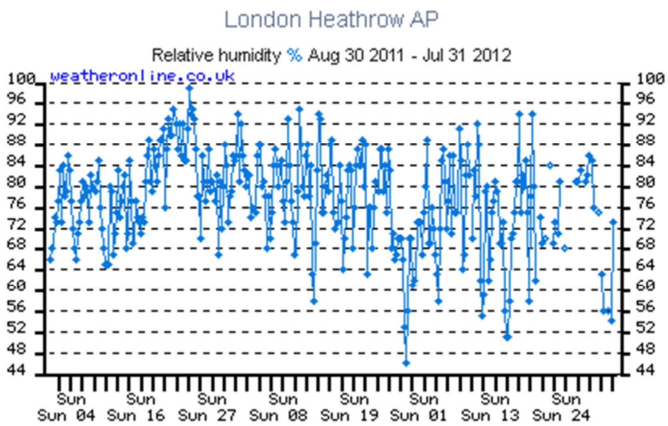

Figure 7:14 Relative humidity variation at London (Heathrow) during 2011-2012

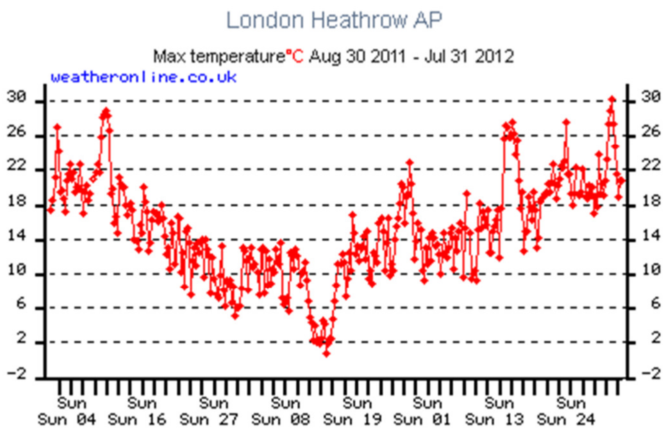

Figure 7:15 Maximum temperature (°C) variation at London (Heathrow) 2011-2012.

Air temperature and air relative humidity observed on London underground platforms throughout a typical year ranged from 20ºC up to 31ºC (figure 7:16) and from 25% to 50% respectively (Ampofo et al, 2011) while the humidity ratio tracks closely the of outside surface conditions, (Gilbey, 2009).

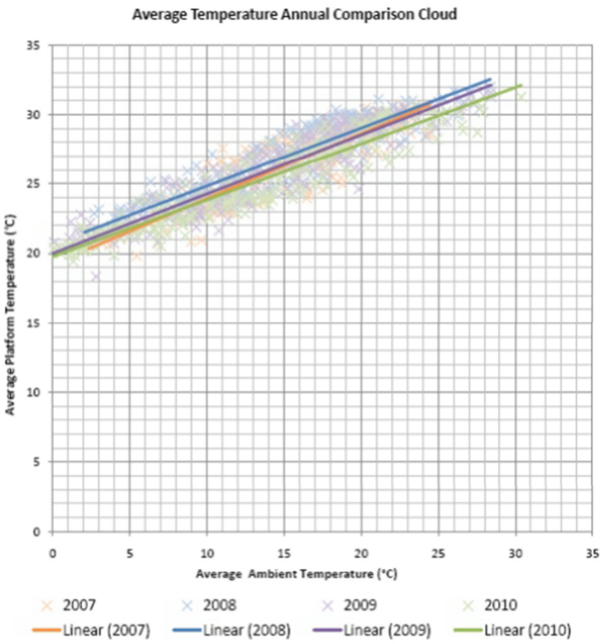

Figure 7:16 Correlation between air temperature on the surface and on underground platforms (Gilbey,2009).

7.3.2.2 Methodology

Since both temperature and humidity interact and effect the air sound absorption, a range of representative and realistic combinations of both air climatic parameters will be used in the predictions of acoustic and speech related parameters on platforms.

The selected temperature range was from 15ºC to 30ºC (in steps of 5ºC) while the relative humidity (RH) range was taken from 20% to 80% (in steps of 20%). The air density setting of the program was maintained constant for all simulations at the value measured on London underground platforms at 1.2Kg/m³ (Ampofo et al, 2004) which is also the simulation program's (CATT-Acoustics) default value.

For the results analysis, two representative case groups from a pool of possible combinations were selected to illustrate the effects of air temperature and humidity on the acoustical parameters of interest. These two case groups consisted of the following combinations:

Case 1 : constant 50% RH and various temperatures within the range above

Case 2 : constant temperature of 20ºC and various air humidity levels within the selected range.

The significance of the parameters' changes due to air and humidity variations was quantified by calculating parameters deviations in terms of JND from reference values according to equation 6:1, 6:2 and table 6:2. In order to show the maximum impact possible of air temperature and humidity conditions on evaluating acoustic parameters, results at 15ºC and RH 20% are taken as the references values. The spatial variability observed across the receiver positions for all predicted evaluating parameters was low, as indicated by the standard deviation (σ) and presented in appendices F.6 and F.7. This high spatial uniformity allows the presentation of results and analysis below to concentrate on averaged results. Standard deviation (σ) values of spatially averaged STI results, are shown in figure 7:17 and 7:18 as vertical black bars. A best fit trend line is shown on those graphs.

7.3.2.3 Results

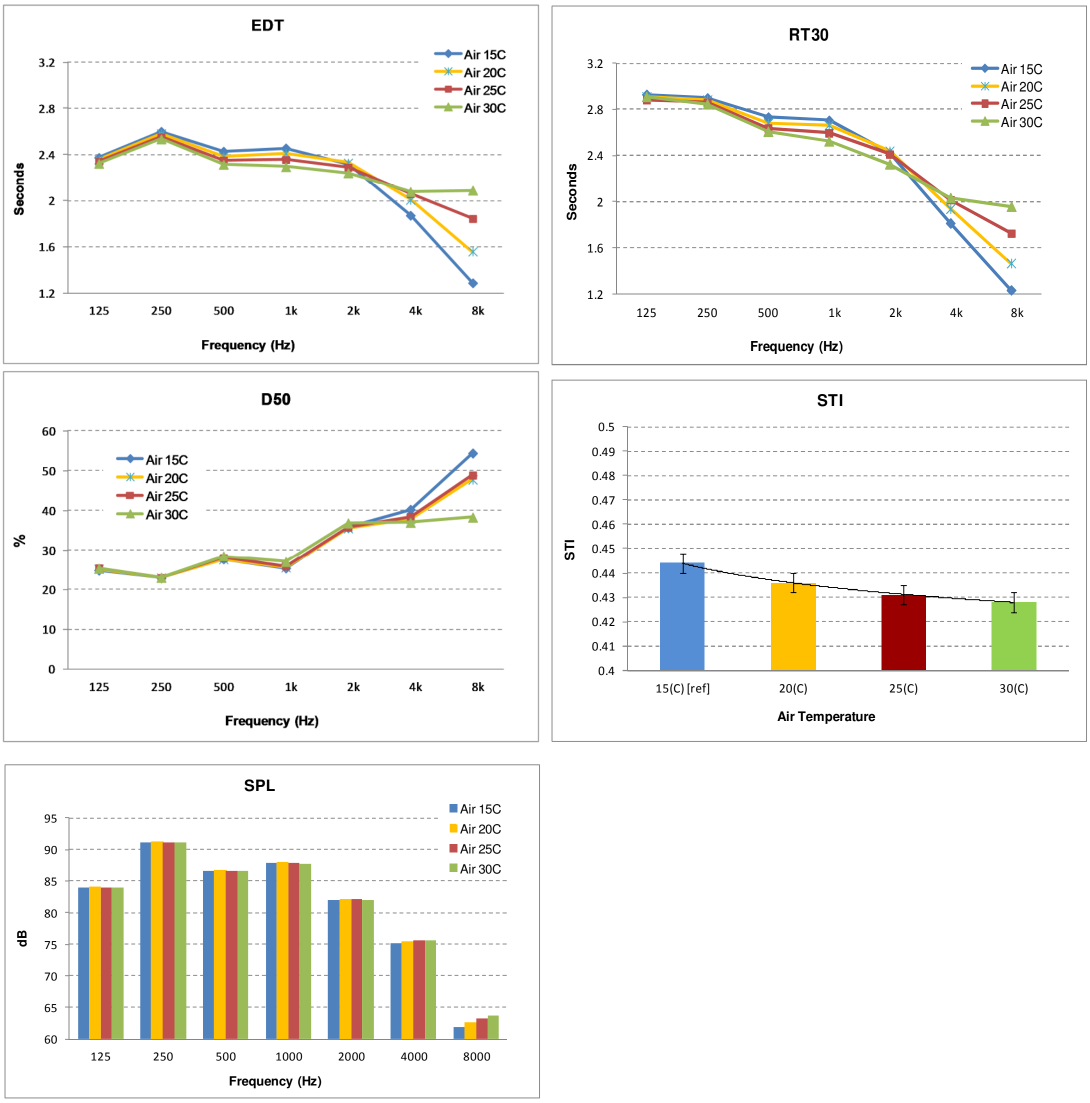

Figure 7:17 Case 1:Acoustic parameters predicted in octave frequency bands at 50% RH for various air temperatures.

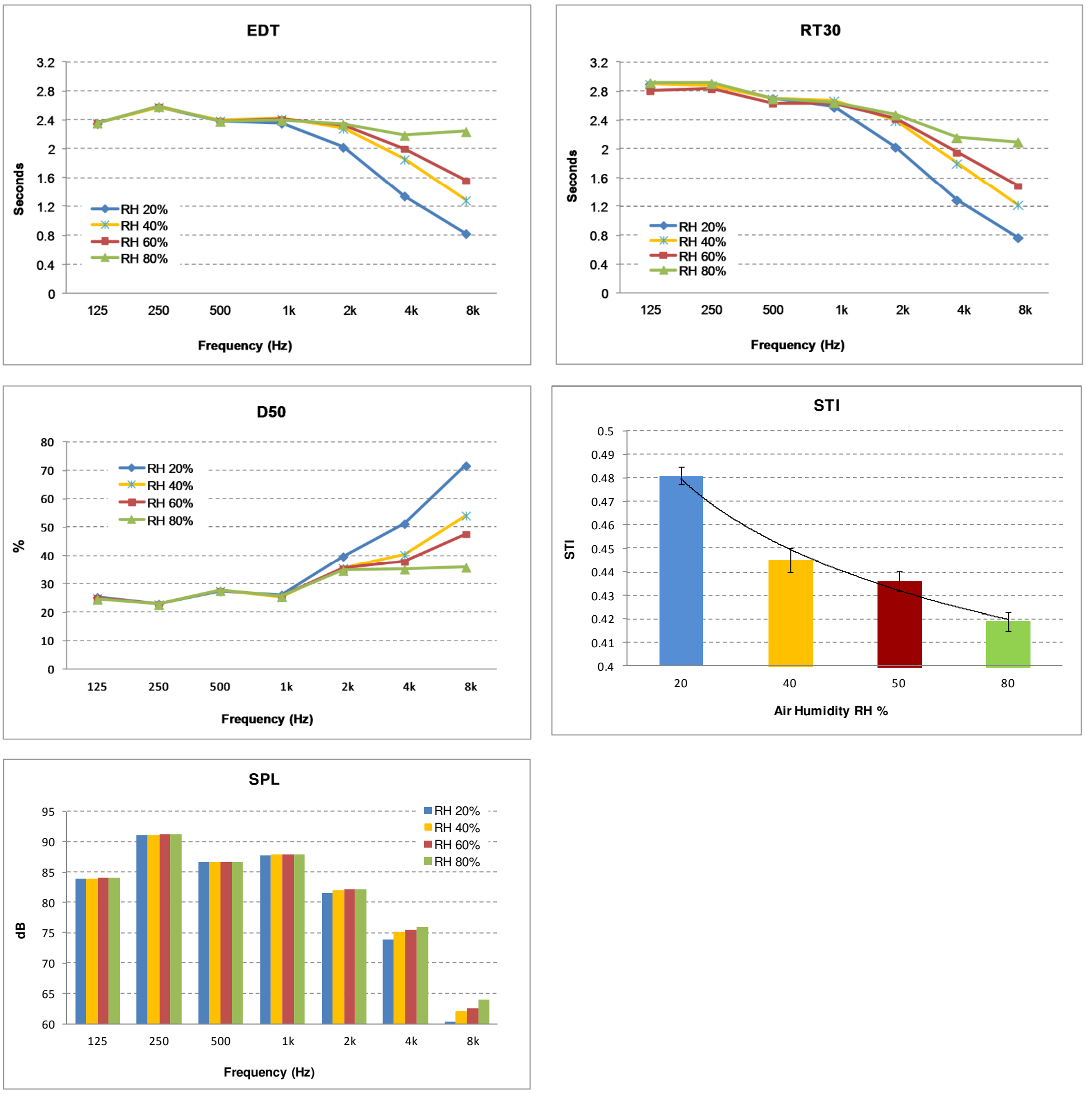

Figure 7:18 Case 2: Platform acoustic parameters predicted in octave frequency bands at 20ºC for various air humidity values.

7.3.2.4 Analysis & discussion

<u>Generic air sound absorption characteristics</u>

From the combined observation of the experimental data presented by Harris (1963) (figure 7:12) and the calculated variability curves based on ISO 9613-1:1993 (figure 7:13), it can be seen that the air exhibits a prominent maximum in sound absorption coefficient at 8kHz octave band when measured in climatic conditions of RH 20% and 20ºC. This maximum value decreases rapidly for lower frequency bands, higher temperatures and higher humidity values. In this decreasing wide region, a less significant absorption variation is observed for considerable changes of any of the three parameters: audio frequency, air temperature and air humidity (figure 7:12).

Case 1, Acoustic effects with air temperature variation at constant humidity

From the results graphs (figures 7:17 and 7:18) and analytical table below (table 7:3), it can be noticed that EDT and RT parameters showed almost identical gradual increase at 4Khz and 8Khz octave bands as the temperature rose. This increase in these reverberation parameters begun to be significant at 4Khz showing much greater increases of up to 6.2 JND at 8KHz.

Table 7:3 shows that the increments in reverberation with temperature associated with the decrease in air absorption are especially significant at 8Khz octave frequency band.

	15 ºC [ref]	20 ºC	25 ºC	30 ºC
EDT(4KHz)	1.87	2.01	2.06	2.08
Increment from [ref]	0	0.134	0.188	0.206
JND increment from [ref]	0	0.7	1	1.1
EDT(8KHz)	1.29	1.56	1.84	2.09
Increment from [ref]	0	0.272	0.556	0.802
JND increment from [ref]	0	2.1	4.3	6.2
RT30(4KHz)	1.82	1.94	2.02	2.04
Increment from [ref]	0	0.124	0.202	0.22
JND increment from [ref]	0	0.7	1.1	1.2
RT30(8KHz)	1.24	1.47	1.73	1.96
Increment from [ref]	0	0.234	0.492	0.724
JND increment from [ref]	0	1.9	4	5.9

Table 7:3 EDT and RT30 increments in seconds and in JND relative to [ref] as a function of frequency and temperature at constant RH 50%.

	15 ºC [ref]	20 ºC	25 ºC	30 ºC
D50 (4KHz)	40.3	37.9	38.5	37.0
JND decrement from [ref]	0	0.5	0.35	0.64
D50 (8KHz)	54.4	47.8	48.9	38.3
JND decrement from [ref]	0	1.3	1.1	3.2
STI	0.44	0.44	0.43	0.43
JND decrement from [ref]	0	0.16	0.26	0.32

Table 7:4 D50 and STI decrements in JND relative to [ref] as function of frequency and temperature at constant RH 50%.

Increases in air temperature from reference values resulted in slight decrements of D50 although these were below the JND (table 7:4). The reductions in STI values due to the increase in air temperature were not notable, being below the JND, but did show a clear trend (figure 7:17 and table 7:4). An empirical model was derived from the best fit trend line obtained from the correlation analysis between predicted STI results relative to air temperature *T* variations on the platform (equation 7:8). This model is able to provide estimated changes in STI with variations of air temperature T (ºC) for fixed RH 50%. The operational input range is limited to maximum and minimum temperatures typically registered on underground platforms. The model is applicable for predicting STI values at other air temperatures expected on similar underground platforms equipped with conventional VA system.

$$STI = -0.012 * \ln(T) + 0.444 \qquad \text{(R=0.99)}$$

Equation 7:8 Platform prediction model providing estimated STI for air temperature T (ºC) input for fixed RH 50%.

It was observed that the influence of air temperature on predicted total SPL was negligible at all octave frequency bands except at 8kHz in which the increase in level was considerable although marginally below 1JND. In this octave band a trend was observed by which increasing air temperature showed total SPL increases due to decrements in air absorption observed before (figure 7:17 and table 7:5).

	15 ºC [ref]	**20 ºC**	**25 ºC**	**30 ºC**
SPL (8KHz)	71.94	72.62	73.26	73.76
Increment from Ref	0	0.68	1.32	1.82
JND increment from [ref]	0	0.34	0.66	0.91

Table 7:5 Total SPL relative increment at 8kHz as function of temperature

Case 2: Acoustic effects with air humidity variation at constant air temperature

From the relevant results graphs (figure 7:18) and analytical table 7:6 below it can be seen that EDT and RT30 parameters showed almost identical gradual increase at 4Khz and 8Khz octave bands as the humidity rose. This increase in EDT and RT30 became most significant at 8kHz with increases of up to 17 and 17.2 JND respectively from reference value (table 7:6).

	RH %			
	20 [ref]	**40**	**50**	**80**
EDT(4KHz)	1.34	1.85	2	2.19
Increment from [ref]	0	0.51	0.66	0.85
JND increment from [ref]	0	3.8	4.9	6.3
EDT(8KHz)	0.83	1.28	1.56	2.24
Increment from [ref]	0	0.45	0.73	1.41
JND increment from [ref]	0	5.4	8.8	17.0
RT30(4KHz)	1.29	1.8	1.95	2.16
Increment from [ref]	0	0.51	0.66	0.87
JND increment from [ref]	0	4.0	5.1	6.7
RT30(8KHz)	0.77	1.22	1.49	2.1
Increment from [ref]	0	0.45	0.72	1.33
JND increment from [ref]	0	5.8	9.3	17.2

Table 7:6 EDT and RT30 increments in seconds and in JND relative to [ref] as a function of frequency and humidity at a constant air temperature of 20°C.

The influence of air humidity variation on reverberation is clearly reflected on speech related parameters D50 and STI values (figure 7:18 and table 7:7). Increases in air humidity from reference values resulted in noticeable decrements of D50 of up to 7.2 JND in the 8kHz frequency band. Similarly reductions in STI due to the increases in humidity were notable up to 1.2JND. Both D50 and STI reductions were comparatively higher than the observed in the temperature variation analysis (table 7:4).

	RH %			
	20 [ref]	**40**	**50**	**80**
D50 (4KHz)	51.38	40.38	37.9	35.28
JND decrement from [ref]	0	2.2	2.7	3.2
D50 (8KHz)	71.7	54.2	47.7	35.9
JND decrement from [ref]	0	3.5	4.8	7.2
STI	0.481	0.445	0.44	0.419
JND decrement from [ref]	0	0.72	0.9	1.24

Table 7:7 D50 and STI decrements in JND relative to [ref] as function of frequency and humidity at constant temperature of 20°C.

An empirical model function derived from the best fit trend line obtained from correlation analysis between STI results relative to air humidity RH variations on the platform is provided below as equation 7:9. This model is able to provide estimated changes in STI with variations in air humidity for a constant temperature of 20ºC. The operational input range is limited to maximum and minimum RH typically registered on underground platforms. It could be used for predicting STI values at other humidity levels expected on similar underground platforms equipped with conventional VA system.

$$STI = -0.043 * \ln(RH) + 0.4795 \qquad \text{(R=0.99)}$$

Equation 7:9 Platform prediction model providing estimated changes in STI for air RH(%) input and fixed temperature at 20°C.

The influence of air humidity on total SPL was negligible at all octave bands except at 4kHz and 8kHz, where the effect of increasing air humidity showed corresponding increases in total SPL due to gradual decrements in sound air absorption. These increases relative to the reference value were particularly significant at 8kHz octave frequency band (1.9 JND) (figure 7:18 and table 7:8) and comparably higher to maximum increases obtained with temperature in previous analyses for the same octave band (figure 7:17 and table 7:5)

	RH %			
	20 [ref]	**40**	**50**	**80**
SPL(4kHz)	83.7	85.1	85.4	85.86
Increment from [ref]	0	1	1.74	2.16
JND increment from [ref]	0	0.5	0.9	1.1
SPL(8kHz)	70.2	71.94	72.6	74
Increment from [ref]	0	1.74	2.4	3.8
JND increment from [ref]	0	0.9	1.2	1.9

Table 7:8 Total SPL relative increments in dB and JND at 4kHz and 8kHz as function of air relative humidity.

7.3.3 Effects of scattering coefficient variation

This section examines the influence of surface scattering coefficients on predicted acoustic and speech intelligibility related parameters.

7.3.3.1 Introduction

Over the past two decades it has been demonstrated that room acoustics prediction programs based on geometrical acoustics must include scattering functionalities to attain reliable predictions of the frequency-dependent diffuse reflection occurring on actual surfaces (Hodgson, 1991), (Vorländer,1995), (Dalenbäck,1995), (Zeng et al, 2006), (Cox and D'antonio,2004), (Christensen, 2008b) . These researchers and programs developers have highlighted the importance of the correct assignment of scattering coefficients to surface materials to obtain an accurate and reliable acoustic simulation and derived parameters prediction of enclosed spaces. In spaces where diffusion is not intrinsically promoted due to irregular geometry and uneven absorption distribution, the role of diffuse reflection in prediction accuracy becomes even more relevant. Reflection based diffusion is not promoted on underground platforms due to the extreme disproportionate geometry (Kang,1996a). Other studies (Kang,1997a), (Kang, 2002), (Li & Lam, 2005) and (Yang & Shield, 2001) have studied and accounted theoretically for the effects of diffuse reflection caused by surface scattering properties in acoustic simulation of long enclosures. However, most of the relevant research in the literature considering the importance of diffuse reflection in acoustic prediction involved the use of a single source.

Still today there is not sufficient documented data of scattering coefficients of common materials (Zeng et al, 2006). This is due mainly to the fact that scattering coefficient of different materials is complex and time consuming to determine and results are difficult to harmonise. Developers of all the advanced commercial room simulation programs provide a rough guideline to aid the user to estimate the room surfaces scattering coefficients. This guideline is expected to be combined with a fair knowledge and experience in acoustic modeling.

The objective of this section is to determine the sensitivity to surface scattering coefficients of predicted acoustic and speech intelligibility related parameters. The most accurate scattering coefficient of platforms surfaces is also determined. This knowledge will serve as useful design guidance in practical applications to compensate for the intrinsic uncertainty of the scattering coefficient estimation process.

7.3.3.2 Methodology

Plat-1 original prediction model was utilised to run simulations with different surface scattering coefficients assignments. In an iterative process only the scattering coefficients were modified in turn for each simulation while all other input data including geometric and operational settings were maintained as in Plat-1 original.

“Surface” option was selected in “general settings” of the CATT-Acoustics program interface which allows the calculation of surface diffusion according to the scattering coefficients assigned by the user for each surface of the geometry. Scattering coefficient assignment for all surfaces of the geometrical model were introduced and adjusted manually by the user in the surface definition file.

For consistency with simulations performed in previous sections edge diffusion was not considered as station platforms features mostly very large and flat surfaces with few edges.

The range of scattering coefficients investigated in this chapter was based on the program's manual guidelines (Dalenback, 2007), recommendations given in the literature and the author’s previous modeling experience utilising the ray tracing and hybrid techniques on underground platforms. The different scattering coefficient assignments were within the range of realistic values representative of actual conditions and were applied to the platform hard surfaces.

Four different coefficient assignment cases were estimated for the purposes of this investigation and are described as follows:

- “0%” case: All surfaces assigned 0% scattering at all frequency bands (extreme case of total specular reflection).
- “5%” case: All hard surfaces assigned 5% scattering at all frequency bands.
- “10%” case: All hard surfaces assigned 10% scattering at all frequency bands.
- “High” case: Different coefficients assigned to different octave bands to different surfaces according to their geometry and size and estimated surface roughness (table 7:9).

		Frequency octave bands(Hz)						
		125	250	500	1000	2000	4000	8000
Surface area	Walls	10	15	18	20	23	25	25
	Platform floor	10	15	18	20	23	25	25
	Track pit	30	35	40	45	50	60	70
	Rest of hard surfaces	5	5	5	5	5	5	5

Table 7:9 Scattering coefficients in % estimated for the "High" assignment case

7.3.3.3 Results

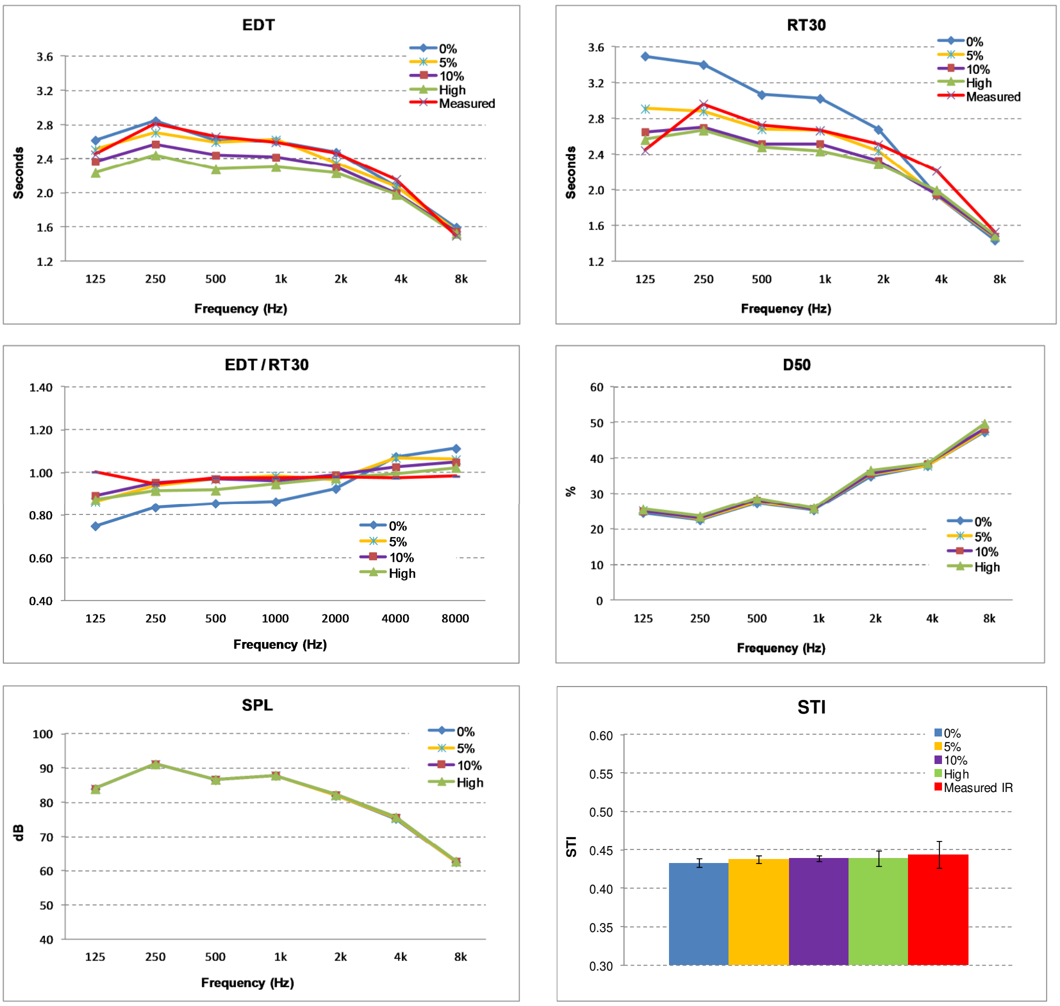

Figure 7:19 Predicted acoustic parameters with various scattering coefficient assignments

7.3.3.4 Analysis and discussion

Both reverberation time parameters EDT and RT30 appeared sensitive to changes of scattering coefficient. Table 7:10 provides EDT and RT30 prediction deviations from the measured reference in terms of JND as a function of different scattering coefficient assignments. Positive values correspond to values higher than the reference.

EDT (JND)		Frequency (Hz)						
		125	250	500	1000	2000	4000	8000
Scattering coefficient	0%	0.69	0.13	-0.15	0.05	0.06	-0.36	0.64
	5%	0.23	-0.37	-0.24	0.10	-0.45	-0.39	0.39
	10%	-0.34	-0.84	-0.82	-0.68	-0.65	-0.74	0.32
	High	-0.87	-1.30	-1.43	-1.11	-0.93	-0.82	0.16

RT30 (JND)		Frequency (Hz)						
		125	250	500	1000	2000	4000	8000
Scattering coefficient	0%	4.32	1.48	1.24	1.34	0.67	-1.26	-0.60
	5%	1.91	-0.28	-0.17	0.00	-0.28	-1.24	-0.38
	10%	0.85	-0.90	-0.77	-0.56	-0.72	-1.21	-0.33
	High	0.50	-0.99	-0.89	-0.86	-0.87	-0.99	-0.25

Table 7:10 EDT and RT30 prediction deviations from measured reference in JND for different scattering coefficient assignments.

In figure 7:19 and table 7:10 it can be seen that for the correct prediction of RT30 the scattering coefficients becomes more critical at low and mid frequencies while at high frequency bands this effect is less notable showing a convergence of results at all scattering assignments.

EDT showed relatively less sensitivity than RT30 to correct values of scattering although both parameters presented similar convergence at high frequencies. These facts suggest that the scattering coefficient estimation for underground platform surfaces becomes less critical in the high frequency bands where reverberation parameters are dominated by absorptive processes. The sensitivity of the space to scattering coefficient as indicated by EDT and RT30 in terms of perceptual changes (JND) is low except for the extreme case of specular reflection where prediction deviations are significant by being above 1JND (table 7:10). A very important indication from the results above is the clear decrease in reverberation time as seen in EDT and RT predictions, with increases of scattering coefficient (figure 7:19). This effect can be taken into account and utilised as an alternative and additional means to control reverberation at the design stage.

EDT was less affected in the extreme case of total specular reflection (0% scattering). This is because the EDT prediction strongly accounts for the energy of first two reflections handled as specular reflection geometry (image-source mode of the hybrid program's operation).

The calculation algorithm of the program produced significant errors calculating the late reverberation part of the energy decay (RT30) if diffuse reflection was disallowed (specular reflection) (figure 7:19). It appears that although the surfaces of the platform are extremely large, smooth and reflective, a minimum and realistic degree of scattering needs to be assigned to the surfaces for the ray tracing calculation algorithm represent correctly the complete sound decay.

Taking the ratio EDT/RT30 =1 as an indication of total diffuse sound field, it can be observed in figure 7:19 that quasi-diffuse field condition observed in previous investigation and in chapter 6 and 8, are exhibited similarly for all scattering coefficient assignments except for the case of specular reflection.

D50 and STI showed insensitivity to variations of scattering coefficient with predicted values remaining unchanged even in the extreme case of total specular reflection (0%) assignment. The changes on reverberation time caused by variation of scattering coefficient were not sufficiently high to influence significantly the speech related parameters which have relatively low sensitivity to moderate reverberation time variations (see 7.3.1 and chapters 5 and 6).

Nevertheless, a marginal trend in increasing values of STI with higher values of scattering coefficient was observed (figure 7:19 and appendix F.8). Although this trend is not significant in terms of JND, it suggests the potential of increasing STI scores resulting from reverberation reduction provided by significantly higher surface scattering. This observation agrees with the conclusions presented in a similar investigation by Kang (1998).

The lack of sensitivity to significant changes in scattering coefficient of speech related and total SPL parameters, can be interpreted as a high tolerance in the scattering estimation. This confirmation becomes very useful knowledge specially when accurate scattering coefficients of common surface materials data remains unavailable.

The 5% (0.05) scattering coefficient assignment case produced the most accurate prediction results overall for all parameter predictions. The above retrospectively validates the utilization of this assignment for chapter 6 simulations. This coefficient assignment will be also consistently utilised for the later investigations in this chapter.

Significant surface scattering on an underground platform has the potential of reducing long distance propagation of late energy and locally promote early reflection energy beneficial in speech intelligibility. In order to substantially increase the surface scattering and thus contribute in the increase of the STI score, some practical approaches are proposed for introduction in the early design or renovation stages.

- Rough textured decorative walls (e.g. modular and removable cover layers)
- Multipurpose acoustic ceiling cladding (see 7:3:5)
- Use of rough/textured concrete if concrete wall surface are to be exposed
- Introduction of functional and decorative furniture/art work (e.g. CMS boxes, luminaries, benches, sculptures) with deliberate protruding three dimensional shapes into the platform volume.

7.3.4 Effect of variations in sound absorption

This section examines the predicted sensitivity of acoustic and speech related parameters to variations in sound absorption.

7.3.4.1 Introduction

Acoustic absorption is one of the most important characteristics determining the sound field of an enclosed space. It therefore plays a major role in the achievement of satisfactory VA performance parameters. Absorptive acoustic treatment is used on underground train stations around the world as an effective solution to control ambient noise and reduce the inherent long reverberation resultant from the large volume and hard surface finishes characterising those spaces (Westerberg,1986), (Sü & Çalışkan,2007), (see chapter 3).

The incorporation of acoustic treatment in the renovation and upgrade of subsurface London underground stations has been traditionally discouraged by its engineering management primarily on grounds of cost.

Consistent with other research works, this thesis has shown the severe difficulty in achieving the required speech intelligibility prediction performance target of 0.5 STI (Orlowski, 1994), (Yang, 1997), (Harrison, 2001) when the loudspeaker array is the sole design variable of the acoustic transmission channel allowed to be altered. In the investigations above it has been shown that compromises on loudspeaker configurations are necessary to achieve modest improvement in STI. It has been found that the maximum attainable STI appears capped by the acoustics of the space, namely reverberation and that limit is well below the minimum performance target.

Few relevant research studies have considered the application of absorbent material treatment in different underground station platforms (Westerberg, G. 1986), (Orlowski, 1994), (Kang, 1996e,1998), (Sü, & Çalışkan, 2007). However these studies did not represent the conditions of deep London underground platforms and the materials employed were unrealistic or not suitable for real implementations during the renovation or upgrade of underground station areas.

This chapter determines the sensitivity to sound absorption and the impact of different amounts of absorption on the sound field as predicted by acoustic and speech intelligibility related parameters. The minimum amount of a realistic acoustic treatment to achieve the minimum STI performance target is determined. This is followed by a brief cost-benefit analysis supporting the benefits on VA system performance by the addition of suitable acoustic treatment.

7.3.4.2 Methodology

Gradual variations of sound absorption on the platform model were simulated in turn (table 7:11). For comparative purposes, only the amount of absorption was varied in the simulations while all other input data including geometric and operational settings were maintained as in Plat-1 original.

Minimal variations of acoustic absorption were simulated to determine the sensitivity of the platform to acoustic absorption. The optimal amount and location of sound absorption to achieve the predicted speech intelligibility target STI ≥ 0.5 was then determined by a gradual incremental approach in the amount of absorption applied and variations of its strategic placement. The strategic placement only considered practical and real-world implementable locations and configurations (i.e. accessible for maintenance and low cost installation).

Table 7:11 summarizes the details of the acoustic absorption arrangement cases considered for this study.

Arrangement Case	Description	Surface area (m²)	Plat 1 Surface area covered (%)
-	Plat 1 Original total internal surface	2586	100.0
A	Four crosspassages removed	20	-1.0
REF	Plat 1 Original	ref	ref
B	50mm Firesound on both End walls	30	1.2
C	30mm Firesound strip (1) centered on ceiling	104	4.0
D	30mm Firesound strip (1) by loudspeakers	104	4.0
E	30mm Firesound 3 strips continous centered on ceiling	312	12.1
F	30mm Firesound 3 strips spaced on ceiling	312	12.1

Table 7:11 Absorption arrangement cases location and coverage surface area

Four cross passages removed refer to the simulated removal of those four open areas modelled as high absorption. A strip refers to the platform long surface sections forming the internal curved ceiling of the model, while the end wall refer to the platform walls situated at each end of the passenger platform. The location of the strip/s can be implemented by the loudspeaker array, centered on the ceiling or spaced apart on the ceiling (figure 7:20).

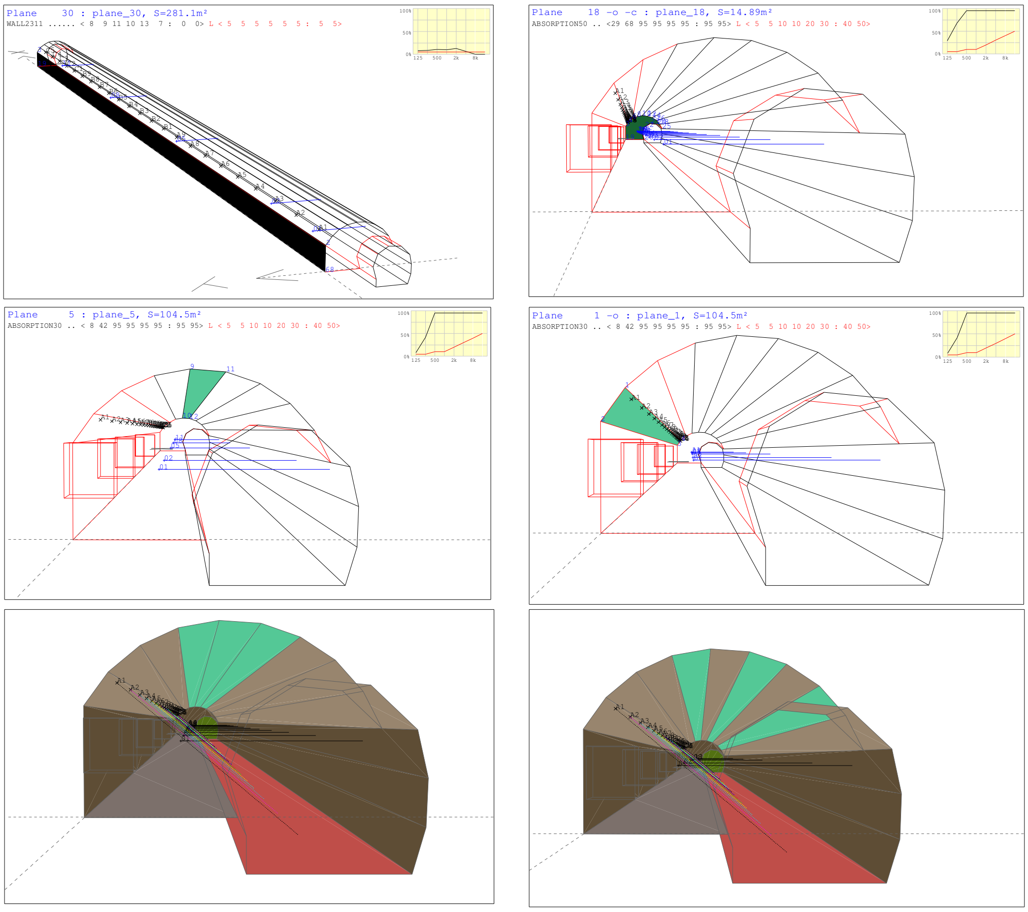

Figure 7:20 From top left to bottom right CATT model view of removal of cross-passages and absorption treatment (green strips) arrangement cases B, C, D, E, F

The acoustic treatment material employed in the simulations was “Firesound” which is a commercially available sound absorbent panel approved[32] by London Underground for utilization on subsurface underground platforms. This product consists of a panel arrangement of packs of mineral wool infill of a certain thickness behind a perforated metallic tray acoustically transparent which provides containment, protection and visual interface. The absorption coefficients assigned to the simulated treatment material on the prediction model were based on the data provided by the manufacturer (appendix F.14). Manufacture’s absorption coefficients

[32] FireSound 30mm and 50mm thick panels are designed to meet the installation, acoustical, mechanical, smoke and fire requirements of London Underground Standard for stations areas and tunnels.

between 1kHz and 8kHz were slightly downgraded from 1 to 0.95 in the acoustic simulation to avoid the extreme and unrealistic 100% absorption situation which would lead the CATT calculation algorithm to erroneous predictions. Table 7:12 lists the absorption coefficient assigned to the Firesound simulated panels. Complete Firesound manufacturer's technical and performance specifications are presented in appendix F.14.

To minimize cost and platform volume taken by the acoustic treatment, the 30mm thick Firesound panel option was used for the large scale application on walls and ceilings. For the specific case of treatment applied only onto the end walls, the 50mm thick panel option was selected for maximum performance effect.

Estimated scattering coefficients assigned to surfaces covered by the acoustic treatment were based on the relevant CATT program guidelines (Dalenback, 2007) and values from relevant literature (see chapter 6). These estimated values are detailed in table 7:12. The absorption and scattering coefficient assigned for the cross-passages and tunnel openings were the same as modeled for Plat-1 original simulations.

			Frequency octave bands (Hz)						
			125	**250**	**500**	**1k**	**2k**	**4k**	**8k**
Firesound	**Sound absortption coefficient**	**30mm**	0.08	0.42	0.95	0.95	0.95	0.95	0.95
		50mm	0.29	0.68	0.95	0.95	0.95	0.95	0.95
	Scattering coefficient	**30mm**	0.05	0.05	0.1	0.1	0.2	0.3	0.4
		50mm	0.05	0.05	0.1	0.1	0.2	0.3	0.4

Table 7:12 Firesound acoustic treatment absorption coefficients based on manufacturer data and corresponding estimated scattering coefficients

7.3.4.3 Results

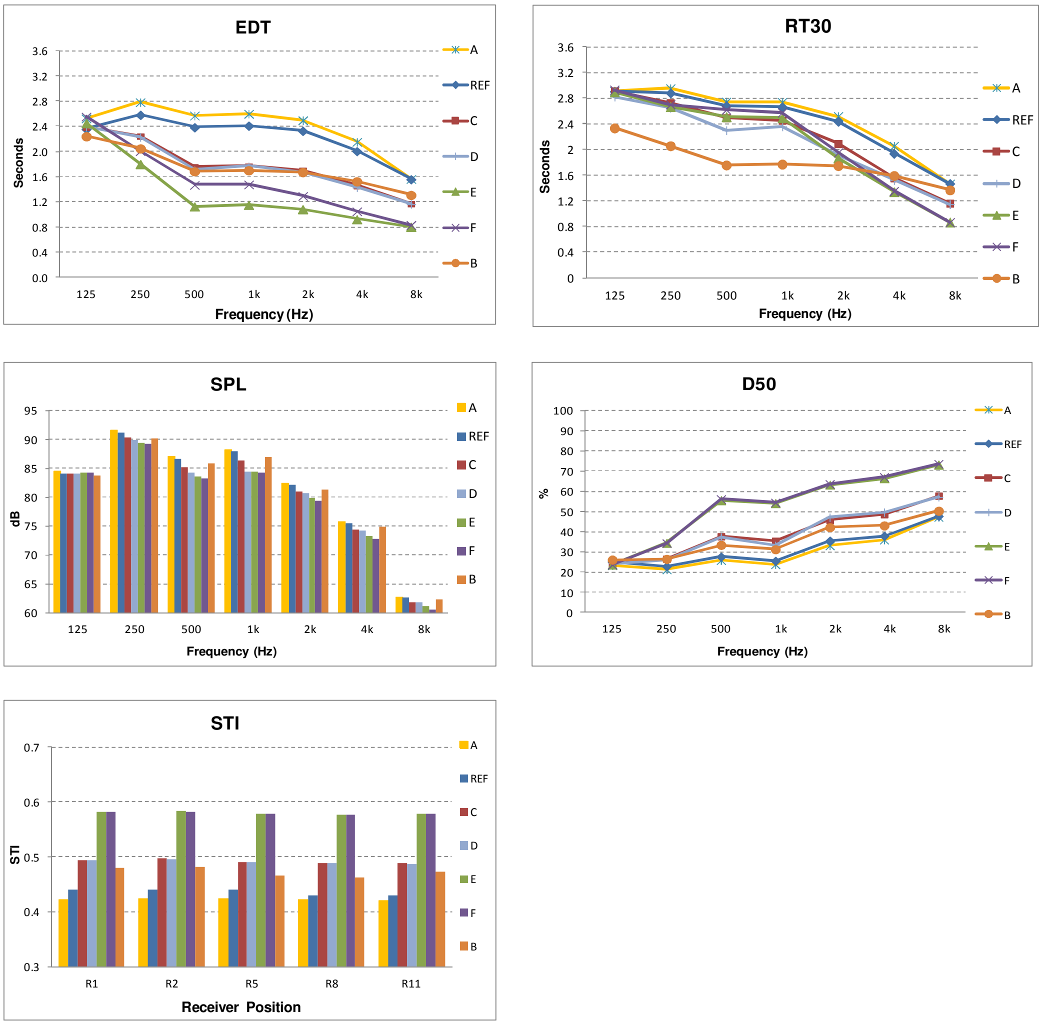

Figure 7:21 Parameters' prediction results averaged over the receiver positions for different absorption arrangement cases

Absorption arrangement case	dBA	Difference with REF (dB)	Difference with REF (JND)
A Crosspassage blocked (-1%)	91.4	0.35	0.18
REF Crosspassage open (0%)	91.1	0	0.00
C One absorbing plane centered (+4%)	89.8	-1.3	-0.65
D One absorbing plane by speaker (+4%)	88.6	-2.4	-1.21
E Threee continuos Absorbing planes (+12%)	88.2	-2.9	-1.44
F Three Spaced Absorbing planes (+12%)	88.0	-3.1	-1.54
B End Walls (+1.2%)	90.2	-0.9	-0.43

Table 7:13 Effect of different absorption arrangements on overall SPL as averaged over the receiver positions

Arrangement case	STI AVERAGE	std	Difference from REF (JND)
A Crosspassages blocked (- 1%)	0.422	0.001	-0.3
REF Crosspassages open (0%)	0.436	0.005	0.0
C One Absorbing plane centered (+ 4%)	0.492	0.003	1.1
D One Absorbing plane by loudspeakers (+ 4%)	0.491	0.004	1.1
E Three continuos Absorbing planes (+12%)	0.579	0.003	2.9
F Three spaced Absorbing planes (+12%)	0.579	0.002	2.9
B Both end Walls (+ 1.2%)	0.473	0.009	0.7

Table 7:14 Effect of different absorption arrangements on STI as averaged over the receiver positions

7.3.4.4 Analysis and discussion

The removal of cross-passages corresponding to a decrease of 1% of absorbent area respect the reference (case A) did not produce significant effect on the evaluating parameters (see figure 7:21 and appendix F.9-13). This result agrees with observations from measured values in chapter 5

However, a coverage platform area of 4% (Cases C and D) produced a clear effect on all evaluating parameters. This modest amount of acoustic treatment reduced the EDT and RT30 values up to 2.1 JND with respect to REF values when observed over the octave bands (see figure 7:21 and appendix F.9-10). The reverberation reduction was more pronounced on EDT than on RT30. This reduction in reverberation created in turn an increase in D50 on average of 1.2JND and 1.1JND for STI respect REF values which resulted in the average platform STI score reaching almost the required target value by scoring 0.49 from the reference value 0.44 (figure 7:21, table 7:14 and appendices F.12-13). The effect of absorption changes in the above analyses was more significant on early reflections related reverberation (EDT) than in overall reverberation (RT30). The evaluating parameters showed no sensitivity to the location of the acoustic treatment strip for the two cases

C and D. All parameters consistently showed a low spatial variability as indicated by a low standard deviation of results observed over individually receive positions (σ typically well below 1JND) (see appendices F.9-13). The introduction of 4% surface area of acoustic treatment reduced the overall average SPL by 0.6JND and 1.2JND in the C and D cases respectively (see figure 7:21 table 7:13). While this decrease is not important, it indicates the potential undesired attenuating effect if large amounts of absorbency is applied to the platform.

An acoustic treatment coverage of 12% of the surface area (cases E and F) produced a substantial reduction of reverberation and subsequent increase of speech related parameters (figure 7:21). As seen in cases C and D, the reduction in reverberation parameters provided by a coverage of 12% was not significant at low frequencies (125Hz and 250Hz) due to the modest absorption coefficient provided by the 30mm treatment material at those frequencies (see figure 7:21 and appendices F.9-10). The selection of 50mm thick panel or the separation of 30mm panels from the applying surface would increase the low frequency absorption performance. However this increase in low frequency absorbency is deemed unnecessary since reverberation at those octave bands has a much less detrimental effect on speech intelligibility and speech intelligibility predicting parameter. Also in cases E and F, it was noticed a large difference in reverberation reduction between EDT and RT30 in which EDT was between 0.4 and 1.38sec lower than corresponding RT30 values. This higher EDT sensitivity to sound absorbency was also seen in cases C and D (figure 7:21 and appendices F.9-10). It was also observed that when the 3 strip panels of case E were arranged together on the centre of the ceiling their effectiveness in the EDT mid frequency range was significantly higher than case F when they were spaced apart. However this location sensitivity was not observed on RT30 values (figure 7:21 and appendix F.10). The reduction on overall SPL produced by the amount of acoustic absorption of cases E and F can be seen in figure 7:21 and table 7:13.The reduction respect the reference was of 2.9dBA and 3.1dBA for cases E and F respectively which corresponds to significant reductions of 1.4 and 1.5JND. The significant reduction of EDT produced by cases E and F was clearly reflected on the substantial increase of speech intelligibility related parameters. For the cases E and F the D50 parameter was equally increased respect the reference case by 3.9 JND on average across the octave bands excepting for 125Hz band were no improvement was appreciable (see figure 7:21 and appendix F.12).

Cases E and F exceeded the required minimum STI performance target by both yielding a STI value of 0.58 which represents an increase from the reference case of 2.9JND (table 7:14).

The strategic placement of the small amount of acoustic treatment (1.2% coverage) on the two end walls of case B produced the most efficient absorbent effect of all arrangement cases. This arrangement reduced the EDT and RT in the mid and high frequencies up to 2.9JND and 3.4JND respectively when compared to the reference case (figure 7:21 and appendices F.9-10). Its effect on D50 and STI was also highly significant considering the small amount of treatment employed. D50 was up to 1.1 JND higher than reference values when observed over the octave bands while the STI score reached 0.47 corresponding to an increase on average of 0.7JND respect the reference value (table 7:14). However the overall SPL attenuation caused by this arrangement with respect to the reference was negligible (0.43JND), (figure 7:21 and table 7:13).

The significant effect of this arrangement was rather uniform across the receiver positions despite its localized position with RT30 (appendix F.10) reporting standard deviations similar to other cases and STI standard deviation being negligible at 0.01. Slightly larger spatial variability was observed in location sensitive EDT and D50 parameters although they were not significant (appendices F.9 and F.12). The high absorption applied on the end walls together with the natural total absorption provided by the large tunnel opening, make the platform geometry in case B as a virtual open ended tube in which direct sound, early and late reflections from distant sources travelling down the tube are efficiently collected and absorbed at that virtual open end (figure 7:22).

Figure 7:22 Deep platform end walls; marked in red is the virtual open end if high sound absorbency is applied to the end wall

These results demonstrated how critical the hard end walls of underground platforms are to late reflections energy and therefore to the degradation of speech intelligibility.

It was observed that as absorption was gradually increased its beneficial effect on reverberation and STI diminish (table 7:14). Acoustic treatment was more efficient when the platform was most sensitive to it and this occurred for the cases of minimal acoustic absorbency and highest reverberation.

An acoustic treatment arrangement consisting of the combination of cases B and C would be recommended as a well balanced option to ensure that STI target is achieved and surpassed at moderate cost and without incurring in significant SPL attenuation. This recommended arrangement is named as case G in the cost analysis below.

7.3.4.5 Cost analysis

The following is a brief and approximated cost analysis illustrating the main cost factors on the practical implementation of the absorption treatment cases studied above. Other smaller and non quantifiable factors are not included in the analysis (i.e. cost overheads, training, transport, equipment)

A cost[33] analysis of the arrangement cases investigated is summarized in table 7:15.

Firesound panels	30mm	50mm
Cost (£) per m² (incl 2012 VAT)	174	183
Manpower cost per hour (£)	80	80

Table 7:15 Firesound current prices for 30mm and 50mm thick panels and installation manpower cost per hour.

Arrangement Case	Description	Surface area (m²)	Plat 1 Surface area covered (%)	Material Cost (£)	Installation costs(£) (estimated)	Total cost (£)	STI	SPL (dBA) attenuation
REF	Plat 1 Original (Ref)	2586	100.0	0	0	0	0.436	0
B	50mm Firesound on both End walls	30	1.2	5508	1200	6708	0.473	0.4
C	30mm Firesound one plane strip centered on ceiling	104	4.0	18096	3200	21296.0	0.492	0.6
D	30mm Firesound on one plane strip by loudspeakers	104	4.0	18096	1920	20016.0	0.491	1.2
E	30mm Firesound on three planes strips continous centered on ceiling	312	12.1	54288	8000	62288.0	0.579	1.4
F	30mm Firesound on three planes strips spaced on ceiling	312	12.1	54288	8000	62288	0.579	1.5
G	50mm on both end walls and one plane strip centred on ceiling	135	5.2	23604	4400	28004	≥ 0.5 (estimated)	1 (estimated)

Table 7:16 Cost analysis of the acoustic treatment arrangement cases considered in this chapter

From the analysis of table 7:16 it can be seen that case D would be the most inexpensive option to almost achieve the STI performance target of 0.5 at a moderate SPL attenuation. If contractual specifications would not allow for the marginal difference in the predicted STI score of case D, then estimated option G would be the next most suitable option balancing satisfactory STI performance, cost and SPL attenuation.

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[33] Acoustic treatment costs including VAT at 20%, were based on Firesound panel price per m² as provided by supplier on 25 July 2012. Installation costs based on an installer average rate of £80 man/hour (typical rates in 2007).

The suitable addition of acoustic treatment to a highly reverberant underground platform can efficiently render an ineffective VA system into an effective and intelligible system fit for purpose. Other subjective benefits derived from the installation of acoustic absorption on reverberant platforms include the enhancement on perception of aural quality, safety and comfort.

The total cost of installing acoustic treatment necessary to ensure an adequate VA system speech transmission quality, will always overweight the costs indirectly generated by an ineffective VA system in case of an emergency situation (e.g. Kings Cross fire disaster 1987, or London 7/7 bombings).

7.3.5 Novel acoustic treatment design

In recent deep platform renovations the exposed concrete ceilings have been clad with plain metal panels for aesthetic reasons. Occasionally the metal cladding conceals cables running along the ceiling length underneath the cladding. In chapter 5 it was found that this metallic ceiling absorbed a range of low frequencies producing modest benefits in STI (see figure 7:23 and chapter 5).

Figure 7:23 Current platform ceiling metal cladding

This section presents a set of novel architectural treatment conceptual designs for the improvement of the acoustic conditions on underground platforms. The implementation of these developments will lead to VA systems to achieve and surpass the minimum specified STI target by greatly reducing the prevalent long reverberation.

The specifications for the proposed designs are as follows:

- Panel designs should provide scattering and sound absorption by employing several absorption mechanisms for maximum efficiency
- The absorbent and scattering performance frequency range should be adjustable and able to cover wide frequency ranges
- The panels should be modular for easy installation, access and maintenance.
- They should be able to conceal or integrate cabling as per current configurations
- They should be aesthetically pleasing or artistically attractive
- Their performance should maximize its effect for minimal coverage
- The design should take into account the smoke, fire and mechanical resistance London Underground specifications
- The material, construction and installation cost should be lower or comparable to current ceiling clad scheme (figure 7:24)

Figure 7:24 depicts the proposed development designs. The figure shows the four different cladding panel configurations for different acoustic performance requirements.

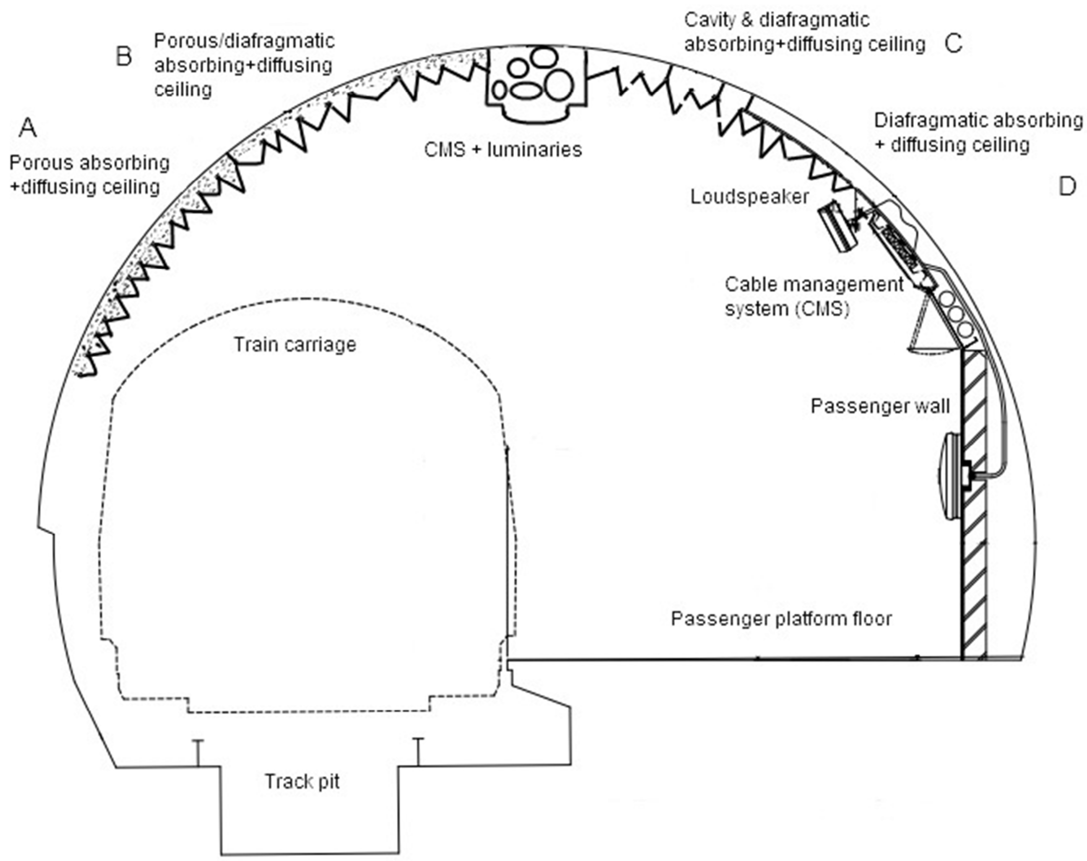

Figure 7:24 Four ceiling cladding panel configurations

Configuration A) consists of a layer (e.g. 20-30mm thick) of porous/fibrous absorber and a micro-perforated metallic cover featuring wedges of a range heights, widths and orientations to promote even diffusive reflection for a wide frequency range. The porous/fibrous absorber layer will provide efficient absorption at mid and high frequencies. The micro-perforations will make the tray acoustically transparent to mid and high frequencies allowing the absorbent layer to perform. The wedges will scatter sound in a wide frequency range according to the wedge dimensions. If porous/fibrous layer is not accepted due to increased costs, installation or flammability concerns, then the micro-perforations could be designed and tuned to absorb mid and high frequency sounds as cavity (Helmholtz) resonators.

Configuration B) is very similar to A) providing scattering and porous absorption except that the wedges' faces will be designed of larger dimensions to act as diaphragmatic absorbers to cater for lower frequencies ranges. Wedge faces can be micro-perforated for acoustic transparency if porous absorbent in laid underneath. Scattering frequency range will vary according to new wedges dimensions.

Configuration C) features wedges with tuned dimensions to provide scattering and diaphragmatic absorption to a selected range of frequencies. The cavity behind the wedges will be partitioned so to form sealed empty chambers of different volumes that with the incorporation of a small opening will form tuned cavity absorbers. This tunable configuration is particularly convenient and efficient at absorbing low frequencies (e.g. remote fan and plant noise, train noise, figure 3.9) which are ineffectively absorbed by porous materials and have the potential of masking important upper frequencies.

Configuration D) is the simplest option offering scattering and diaphragmatic absorption.

Diaphragmatic absorption is more efficient for low frequencies requiring larger wedge dimension. Hence configurations C and D will be better suited for low frequencies while A and B can cater for a wider frequency ranges.

Cables could run inside of a CMS box running along the ceiling length containing the platform lamps, forming a protruding block adding scattering on the ceiling. The long box itself could be designed to contain tune resonators or/and its walls to act as diaphragmatic absorbers.

7.4 Conclusions

This chapter has investigated several key design variables and their impact on acoustical performance of platform VA system

It has been shown that the reverberation on the platform space is not dependent on the type of loudspeaker array configuration. Loudspeaker configurations using directional projector types characterized by a narrow dispersion at mid and high frequencies, achieved higher early-to-late energy ratios which resulted in higher D50 and STI scores. D50 and STI parameters appeared well correlated.

The only effective approach to improve the predicted speech intelligibility on underground platforms when the incorporation of acoustic treatment absorption is not an option, are the reduction of the loudspeaker-receiver distance and the selection of highly directional loudspeakers. These approaches require significantly less amplifier power related cost however they can lead to a lower coverage spatial uniformity and lower sound quality perception.

Hence the design team and other stakeholder must be prepared to prioritise, compromise and balance among conflicting performance targets and other related requirements.

Computer simulation predictions have shown the significant increase in reverberation time at high frequencies with temperature and humidity. Consequently speech related parameters were seen to decrease with rising temperatures and humidity values. The 8kHz was the octave frequency band most sensitive to air climatic variations. Variations in humidity had a greater influence on the evaluating parameters than temperature variations. The neglect of temperature and humidity effects on platform total SPL and speech intelligibility predictors could lead to mild prediction errors in the design which could become critical in marginal contractual compliance situations.

Reverberation time parameters appeared moderately sensitive to a realistic range of estimated surface scattering coefficient assignments while speech related and total SPL parameters were unaffected by changes of scattering coefficient within that range.

The scattering coefficient assignment leading to most accurate acoustical parameters prediction was “5%”. However accurate results were also obtained with values within an upward tolerance. Reverberation time showed a clear trend to decrease with increasing scattering coefficient. This effect can be optimized and utilised as an alternative or additional means of reverberation control to increase the STI.

It has been shown that a deep underground platform is highly sensitive to minimal variations of sound absorption. This makes the application of acoustic treatment the most effective solution to reduce reverberation and allow platform VA system achieve the minimum STI target. A minimum 4% of platform surface area (104m^2) of an LU approved acoustic treatment would be required to be applied to a ceiling strip along the platform length to virtually achieve the required STI minimum performance target without significant SPL attenuation. The platform end walls showed to be a highly efficient location for the application of acoustic treatment. It was observed that gradual increases of acoustic treatment applied on the platform would not correspond to linear increments in STI scores. The life-critical benefits provided by an intelligible and therefore effective VA system in case of an emergency in underground spaces, out weights the argument of the relative high cost of retrofitting acoustic treatment to ensure such effectiveness.

It is hoped that the knowledge and data produced in this chapter will encourage the reconsideration of the incorporation of acoustic treatment on renovated underground platforms as one of the most suitable and cost efficient measures to achieve effective VA systems.

Chapter 8 Review and verification of the quasi diffuse sound field theory for underground platforms

This chapter reviews, verifies and expands on the quasi diffuse sound field theory as a valid concept for underground platforms equipped with distributed VA systems.

8.1 Introduction

Chapter 3 introduced the fact that the acoustic field generated by a single source in a reverberant long space does not satisfy the conditions and characteristics of diffuse fields, thus classic acoustic theory cannot be applied in those spaces (Kang,1996a).

Early research works on the acoustics of reverberant long spaces equipped with distributed multi source systems, observed that reverberation, SPL and STI spatial distribution was significantly different from results obtained when a single source was used (Barnett, 1994b), (Orlowski, 1994), (Kang, 1996d,1998). It was observed that with distributed sources the spatial uniformity of those parameters along the platform length was high.

Yang (1997) provided for the first time a brief theoretical interpretation of those observations using the image source technique on a simple approximation of a platform. Employing her own ray tracing model developed for long spaces, she demonstrated that the sound field in a long enclosure with a sound system of evenly distributed sources, resembles a diffuse field when most of the surfaces are acoustically hard. In those conditions, Yang suggested that the sound field of an underground platform space equipped with an evenly distributed sound system is analogous to a quasi cubic room in which classical room acoustic applies. The theory was briefly validated comparing calculations based on the classic acoustics Sabine formula with predictions from the Yang's ray tracing model and actual measurements. Harrison (2001) made used of the theory to develop a simple model for predicting of RASTI on platforms (see chapter 4). In chapters 6 and 7 measured and predicted parameters were consistently shown to have a high spatial uniformity as well as EDT/RT ratios close to 1 on real world platforms equipped with distributed loudspeaker VA systems. The diffuse field indications observed in those chapters agree with the general theory proposed by Yang.

8.2 Methodology

The verification of the theory is undertaken using and improving upon the validation process initially utilised by Yang (1997). This is carried out by employing a validated hybrid prediction model of an accurate representation of an underground platform equipped with a real world distributed VA system. Moreover more detailed input data (platform geometry, loudspeakers' characteristics and representative loudspeaker quantity and spacing) are also used. Reliable experimental data measured on a representative London underground platform is also employed in the validation process. Calculations and predictions in this validation are undertaken according to the STI calculation procedure BS EN 60268-16 :2003, which includes the influential auditory effects specially relevant to VA systems on underground platforms.

The quasi diffuse sound field theory is explained and illustrated by showing the presence of successive local sound fields of equivalent and diffuse characteristics. The effects of gradual increase in the number of sound sources on key parameters are investigated to support the validation of the quasi diffuse theory and derive practical design knowledge.

The validated Plat-1 original (Plat-1) CATT-Acoustics prediction model has been employed in this chapter as an investigation tool. The configuration and settings were used on the prediction model as in previous sections 6 and 7 unless stated otherwise.

Based on the high spatial uniformity and homogeneity of the sound field created by a distributed VA system on underground platforms (see 8.1 and chapters 6 and 7) only two sample receivers (receivers 8 and 9) are used in these investigations. Receiver 9 (R9) was placed on axis of its nearest loudspeaker B7 while receiver 8

(R8) was off axis in between its two nearest loudspeakers B6 and B7. Both receivers were situated at the listener plane height (1.5m) and along the platform mid width line. Platform loudspeakers (equal sources) were gradually placed in the same configuration and location as Plat-1 (equally spaced every 5m, same aim and height), (figure 8:1).

The investigation is not frequency dependent hence only 500Hz and 2 kHz were considered. The quasi diffuse sound field has been verified in this chapter in two ways:

a) by comparing Plat-1 original model predictions at the two receivers against calculations based on diffuse field assumption (Sabine and STI according to BS EN 60268-16 :2003 formulae)

b) by comparing calculations based on diffuse field assumption at receiver 8 against measurements taken at that receiver.

Simulated loudspeakers of the same type and producing the same output power were introduced gradually in turn and in pairs at the same locations as on Plat-1 complete loudspeaker configuration. The procedure started with a simulation of Plat-1 equipped only with loudspeaker B7. The next simulation featured the addition of two adjacent loudspeakers at each side of B7 which corresponded to loudspeakers B6 and B8 of the complete Plat-1 loudspeaker arrangement (figure 8:1). For each consecutive simulation, pairs of equal loudspeakers were added on each side of the previous set so gradually matching Plat-1 compete loudspeaker arrangement. In this way the simulations showed symmetrical increase in number of loudspeakers centered at B7. A total of 12 consecutive simulations were undertaken, with the last one incorporating the full 23 loudspeakers arrangement representative of the complete VA loudspeaker configuration of Plat-1 original (full platform).

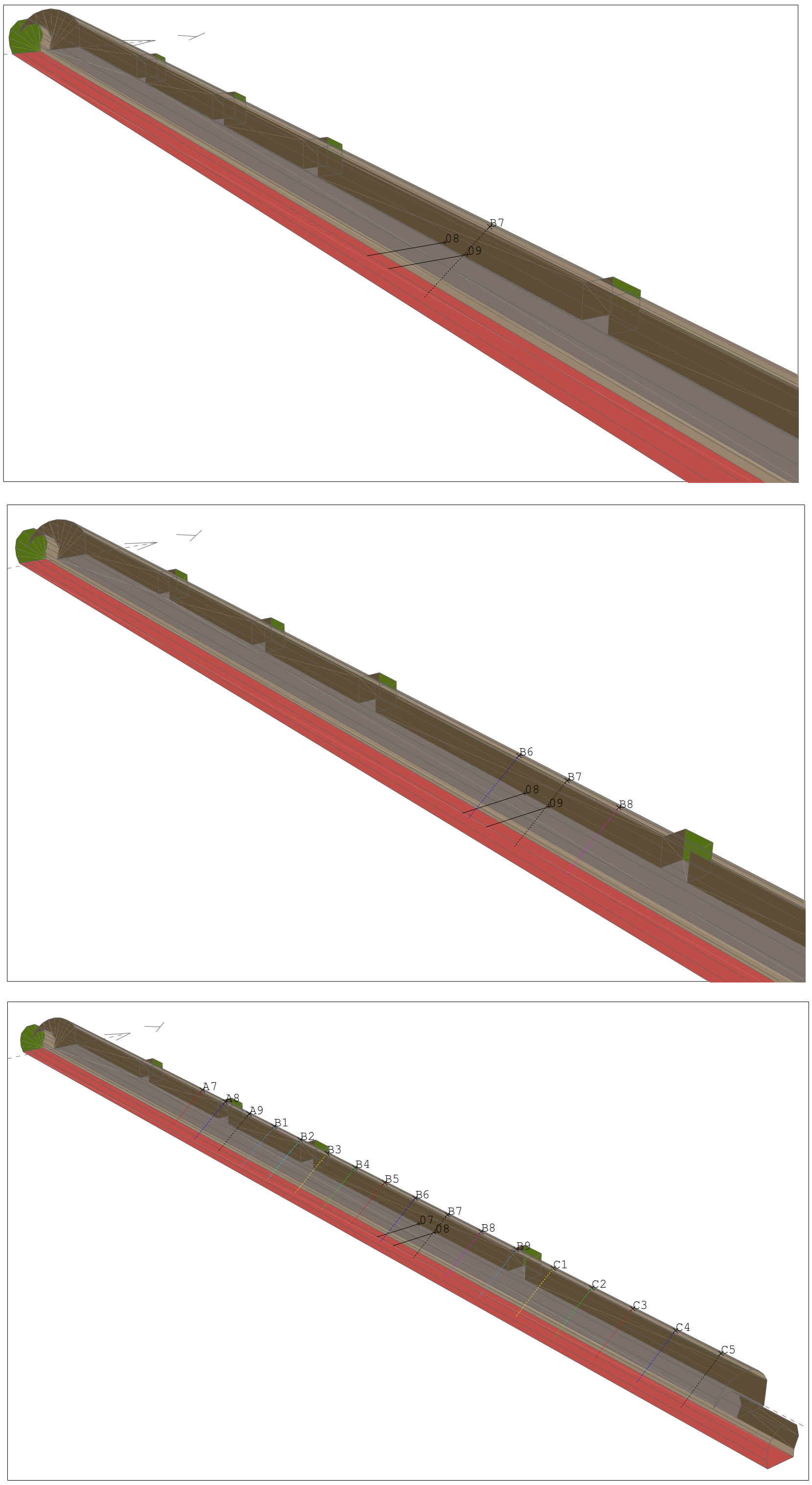

Figure 8:1 CATT view of Plat-1 original showing the gradual increase of number of loudspeakers (examples for 1, 3 and 17 sources)

The sound field of an underground platform equipped with a VA system of an evenly distributed loudspeaker array is hypothesized here as being formed of a succession of similar local sound fields (slices) making up the full platform volume and overall acoustic field (figure 8:2). Each slice is assumed to be characterized by quasi-diffuse sound field characteristics as is expected from a proportional shaped and uniformly distributed low absorption volume. Each local acoustic field is therefore created by the nearest sound source, its local reflective boundaries as well as from the contributions of adjacent and distant sources of the array. The predicted results from the slice acoustic simulations will provide evidence to explain the quasi-diffuse field conditions observed on the full length platform.

CATT-Acoustics was also used in a geometry extracted as a volume slice of the Plat-1 original (figure 8:2) centred at loudspeaker B7. The length of this slice corresponded to Plat-1 original loudspeaker spacing (5m) making the volume slice of quasi cubic proportions. Receivers R8 and R9 were positioned on axis and off axis of B7 as for Plat-1 full length with one source (figure 8:1). The internal side walls of the slice were assigned the absorption coefficient of the platform floor to obtain highly reflective side surfaces. Based on geometrical acoustics conceptualization, the highly reflective side walls of the slice geometry would produce a reflection pattern somewhat similar, in terms of energy and arrival time, to the sound arrivals from nearby and distant loudspeakers in the full length platform geometry. In other words the reflections of the highly reflective side walls represent the arrivals from other loudspeakers outside the slice. A similar validation study as performed for the full platform geometry was undertaken to the volume slice describe above.

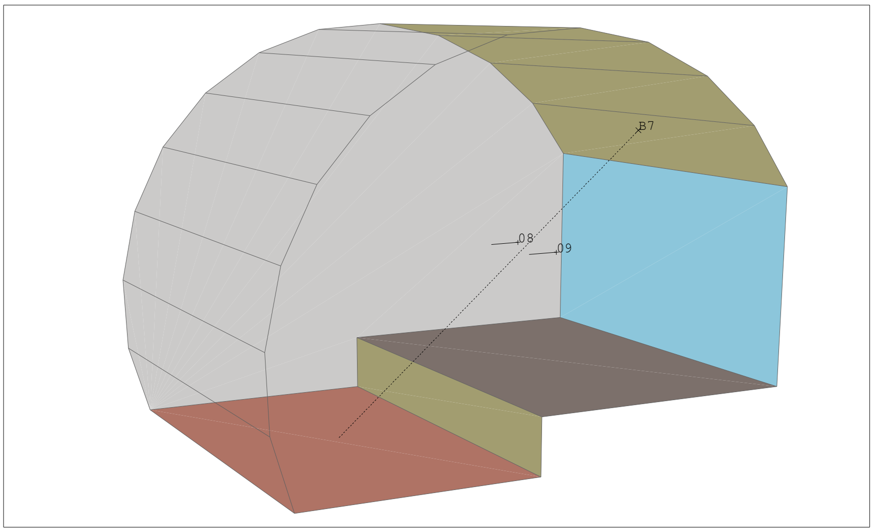

Figure 8:2 CATT view showing a Plat-1 volume slice of 5m long centred at loudspeaker B7 showing receivers R9 (on axis) and R8 (off axis).

The RT and STI calculations in the slice were based on Sabine reverberation theory presented in chapter 2 (equation 2:7) and the STI calculation according to BS EN 60268-16 :2003 (chapter 2, appendix B.2). Sabine reverberation was obtained from CATT-Acoustics classic acoustic calculation option which uses calculated room volume and area-weighted mean absorption coefficient. The calculation of STI according BS EN 60268-16:2003 assumes diffuse field conditions for which the Sabine calculation of reverberation is a valid input. For the STI calculation the signal level in octaves bands were the SPL predicted by Plat-1 at the receiver positions R8 and R9 when B7 source was present. Again, noiseless conditions were assumed in the STI calculation.

8.3 Results

Presented are the predictions and calculations based on classic acoustics for Plat-1 and slice geometries.

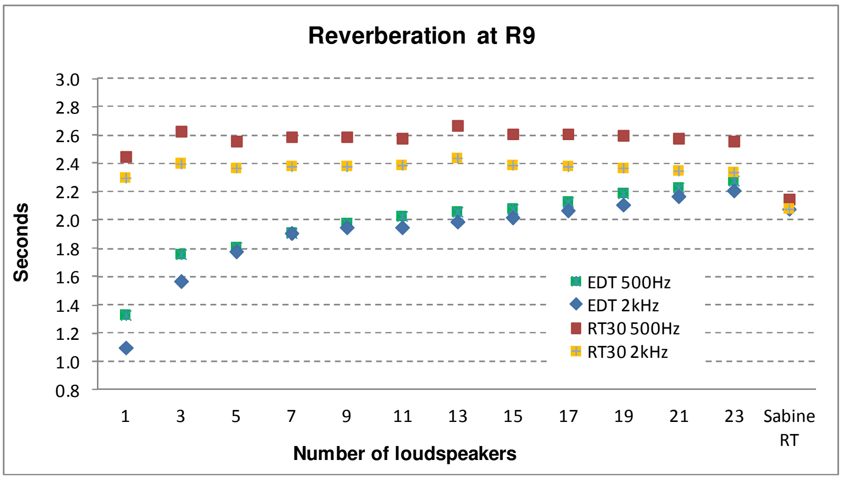

Figure 8:3 Reverberation predicted at R9 (on axis) as a function of number of loudspeakers on the platform. Also calculated Sabine reverberation time.

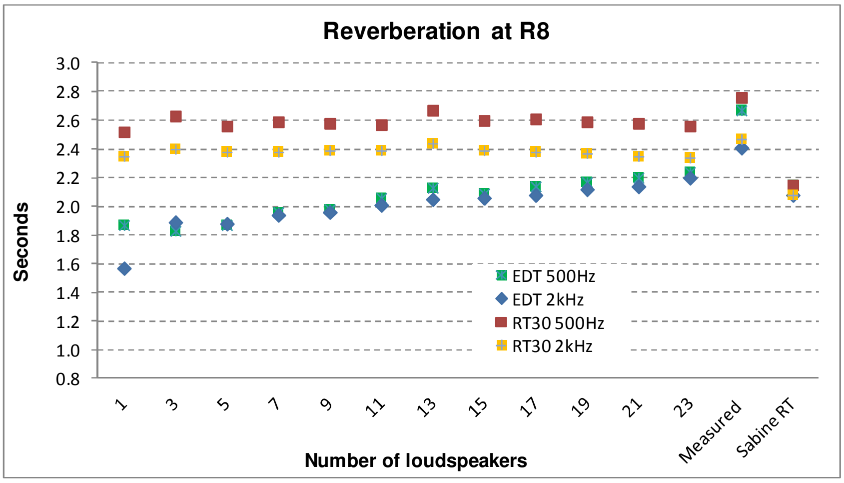

Figure 8:4 Reverberation predicted at R8 (off axis) as a function of number of loudspeakers on the platform. Also measured and calculated reverberation time.

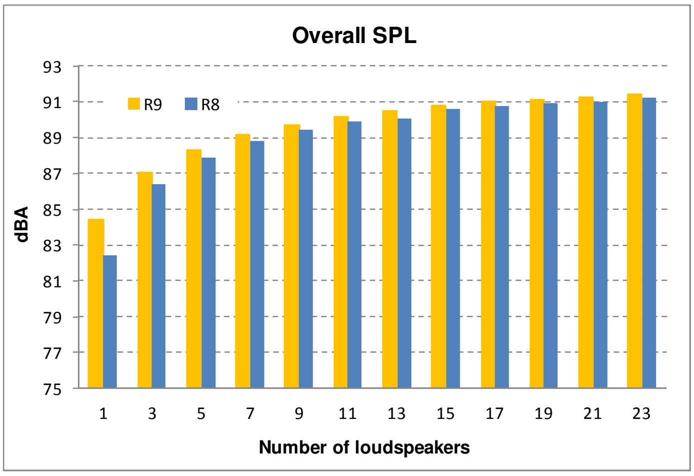

Figure 8:5 Predicted overall SPL at receivers R9 and R8 as a function of number of loudspeakers on the platform.

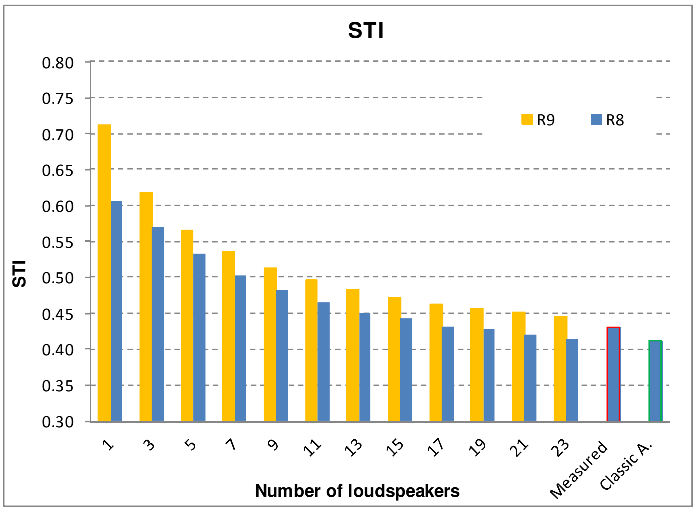

Figure 8:6 Predicted STI at receivers R9 and R8 as a function of number of loudspeakers on the platform. Also measured and calculated STI values.

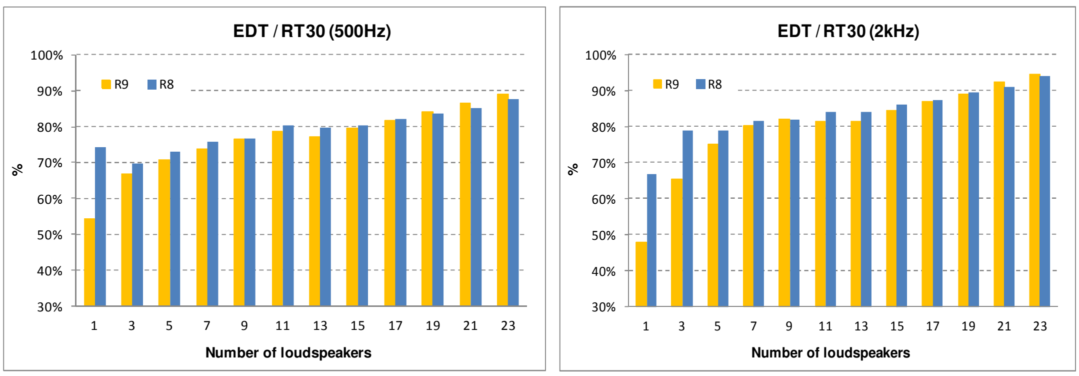

Figure 8:7 Predicted EDT / RT30 ratio at receivers R9 and R8 for 500Hz and 2kHz as a function of number of loudspeakers on the platform.

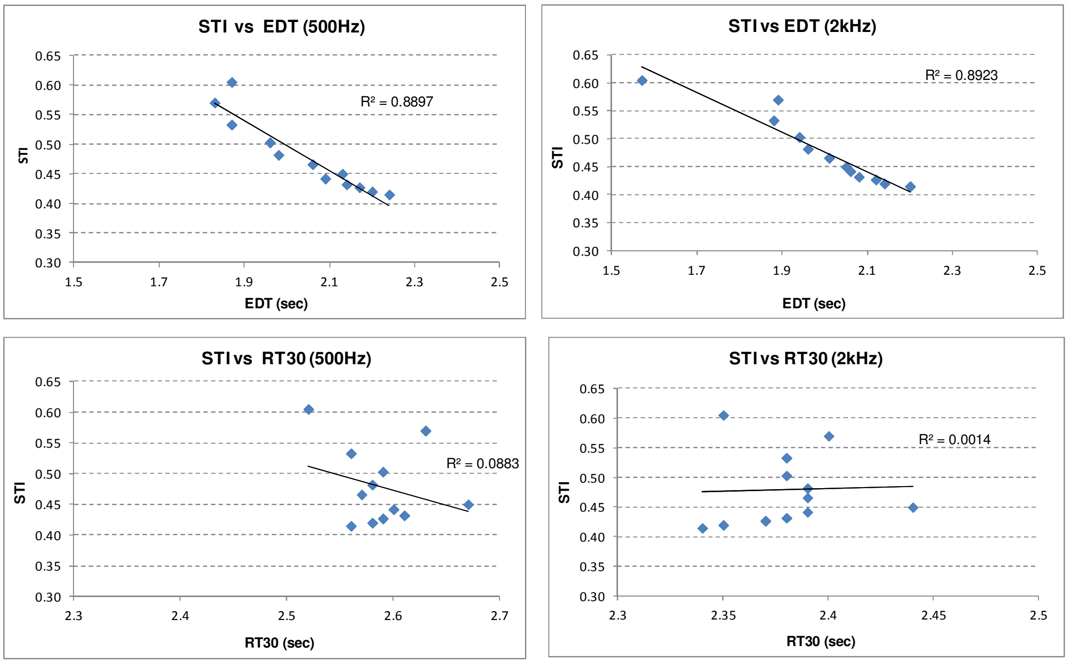

Figure 8:8 Correlation analysis of predicted STI with EDT and RT30 at receiver 8 as a function of increasing number of sources

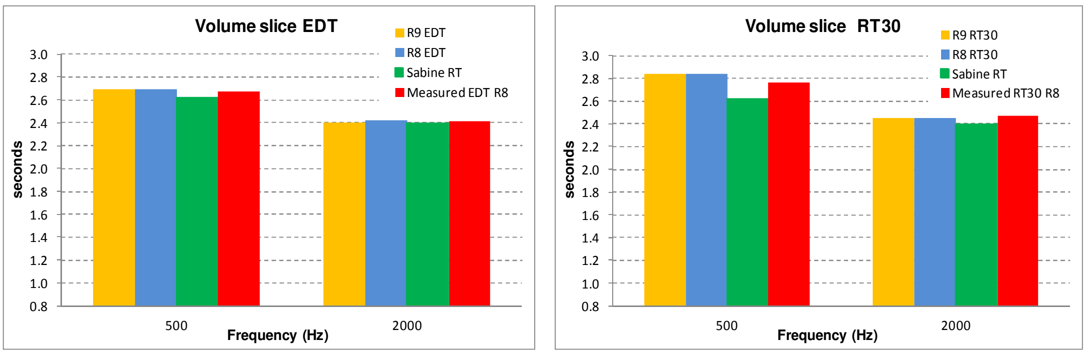

Figure 8:9 EDT and RT30 volume slice predicted, calculated and measured at R8.

Volume slice	500Hz	2kHz
R8 EDT /RT30	0.95	0.99
R9 EDT/RT30	0.95	0.98

Table 8:1 Predicted EDT/RT30 ratios at the two receivers in the volume slice

	EDT (sec) 500(Hz)	RT30 (sec) 500 (Hz)	EDT (sec) (2kHz)	RT30 (sec) 2(kHz)	STI (R8)	STI (R9)
Measured Full platfrom (R8)	2.67	2.76	2.41	2.47	0.43	-
Plat 1 (23 speakers) (R8)	2.29	2.62	2.24	2.37	0.414	-
Plat 1 (23 speakers) (R9)	2.24	2.56	2.2	2.34	-	0.415
Full platform Sabine	2.15	2.15	2.08	2.08	0.41	0.415
Slice volume Sabine	2.62	2.62	2.4	2.4	0.39	0.39
Slice Volume CATT (R8)	2.69	2.83	2.42	2.45	0.42	-
Slice Volume CATT (R9)	2.68	2.83	2.41	2.45		0.435

Table 8:2 Summary of measured, predicted and calculated EDT, RT30 and STI values in volume slice and full length Platform 1 (23 speakers) at R8 and R9

8.4 Analysis and discussion

This section provides the analysis and subsequent discussion of the results presented in section 8.3. Tabulated results are presented in appendix G.

8.4.1 Full length platform

In the results section it can be seen that RT30 was independent to the number of sound sources provided along the platform (figures 8:3 and 8:4). RT30 also appeared insensitive to location relative to nearest source as seen by the negligible difference between R8 and R9 predictions which indicates a high spatial uniformity (figures 8:3, 8:4 and table 8:2).

EDT was significantly dependant to the number of sources when this number was very low. As more sources were introduced on the platform the local direct field and early reflections became less dominant making the EDT value increasingly higher and closer to RT30 values. EDT appeared to be sensitive to location respect the nearest source when the number of sources was between 1 and 3, corresponding to cases where the local direct and early reflection components were still dominant. Understandably this effect was more pronounced for on axis receiver R9. As the number of sources was significantly increased the EDT parameter at both receivers converged with RT30 due to increasing prevalence of late energy arrivals with the number of sources (figure 8:3 and 8:4). This numerical convergence shown in figure 8:7 is due to the increasing straightening of the decay curves as the sound field becomes more diffuse caused by the increase in number of sources. Figure 8:10 shows an example of the decay curve straightening at R8 due to the increase in number of sources from 1 to 17 sources. The numerical convergence between EDT and RT30 as indicated by the EDT/RT30 ratio clearly suggests that as the number of sources increase the sound field becomes more diffuse (figure 8:7).

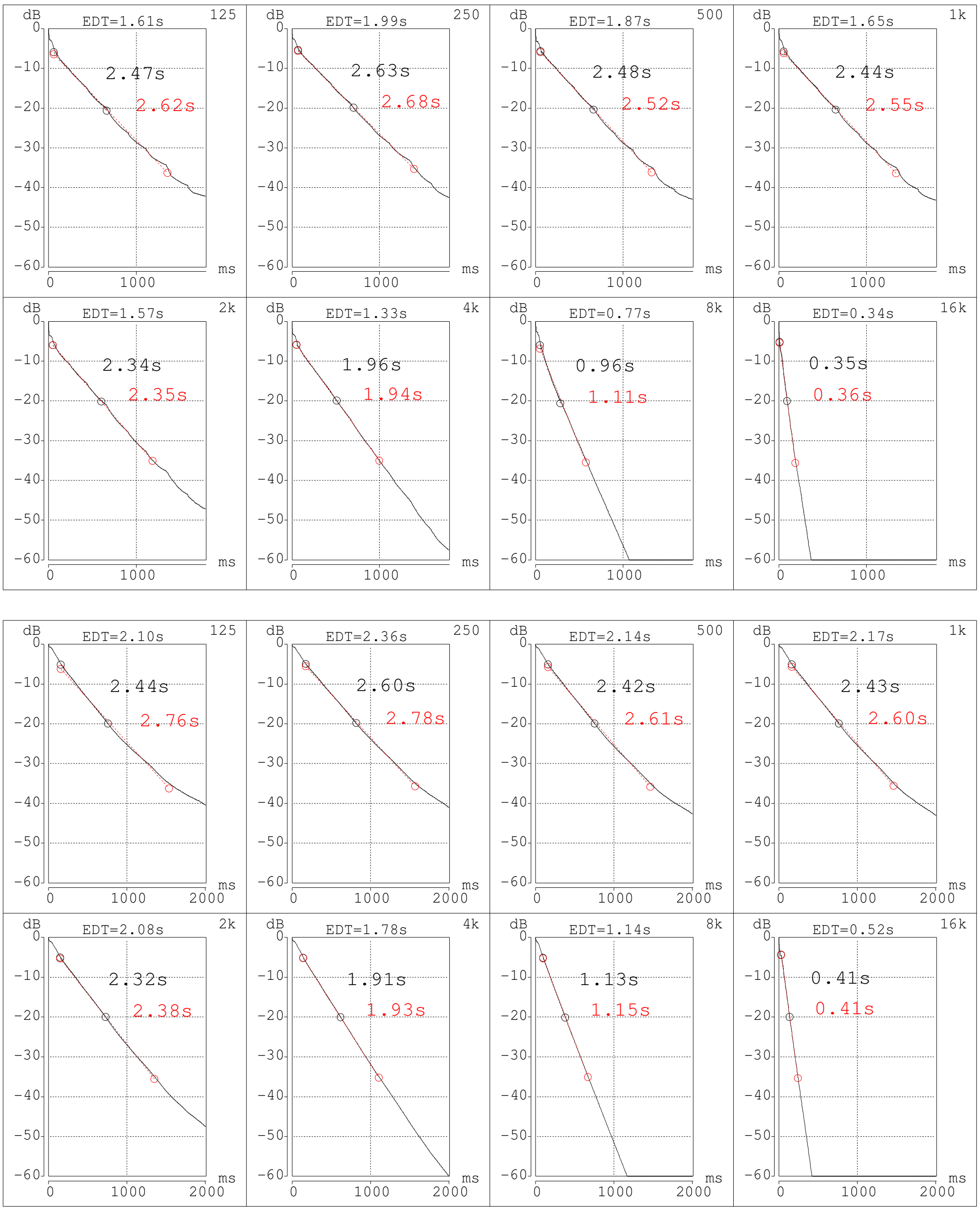

Figure 8:10 Plat-1 predicted decay curves at R8 for 1 source (top) and 17 sources (bottom)

From the analysis of figures 8:3 and 8:4 it appeared that there is a maximum number of sources from which no significant increase in EDT will be reached. This plateau appeared with 23 sources where EDT values closely meet RT30 values. The effect of increasing the number of sources was observed to be frequency independent (figures 8:3 and 8:4).

It can be observed, from figure 8:6, that the STI scores reduced with increasing number of sources in a similar but opposite fashion to EDT values (figures 8:3, 8:4, 8:8). From figures 8:3,8:4 and 8:8; it can been seen that this trend does not hold for RT30 values. This analysis demonstrates that the STI parameter on platforms is highly correlated to EDT in high SNR conditions. It can be clearly derived from figure 8:6 that the increase in sources is detrimental for the STI and this is more significant on receivers on axis and in cases of very few sources. This detrimental effect on STI appeared to stabilize at around 19 sources where increasing the number of sources did not lower the STI score appreciably at both receivers (figure 8:6).The results above showed how influential adjacent and distant sources are on the early decay curve and therefore on STI.

The initial addition of sources increased the overall SPL significantly with few sources although this increment diminished notably as the number of sources was gradually increased (figure 8:5). This trend was very similar to the general EDT trend, the graduation reducing to nothing from 19 sources. Apart from the cases of 1 and 3 sources, the diminishing increase was very similar between the two receiver positions indicating a high spatial uniformity irrespective of position (figure 8:5 and appendix G:3). The high SPL spatial uniformity in the cases comprising more than 19 sources provides further evidence of diffuse field conditions at receiver locations from a high number of adjacent equally spaced sound sources (appendix G:3).

From the results above, it can be seen that the less additional sources present on the platform the less diffuse the sound field becomes, although better speech transmission quality is obtained at the receiver of interest due to the lower reverberation caused by adjacent and distant sources.

Predicted results from the full length platform comprising 23 loudspeakers equally spaced have confirmed that a typical underground platform (Plat-1 original) possesses diffuse field characteristics.

Calculations of the platform reverberation time and STI using Sabine theory showed a close match when compared against values predicted by Plat-1 comprising the complete loudspeaker array (23 loudspeakers) (figures 8:3, 8:4 and table 8:3).

		500Hz	2Khz
R8	EDT	-0.61	-0.71
	RT30	-1.79	-1.22
R9	EDT	-0.57	-0.59
	RT30	-1.60	-1.11

	STI (receiver 8)	STI (receiver 9)
Plat 1 Predicted (23 sources)	0.414	0.415
Sabine + IEC 60268-16 :2003	0.410	0.415
Discrepancy	0.004	0
Relative discrepancy (JND)	0.08	0

Table 8:3 Reverberation time (left) and STI (right) discrepancy in JND units between Sabine calculations and Plat-1 (23 loudspeakers) predictions at R8 and R9

The good agreement of reverberation and STI seen in tables 8:3 between Plat-1 (23 sources) predictions and the Sabine calculations, provides further evidence on the validity of the theory of quasi-diffuse sound field conditions on underground platforms equipped with a distributed VA loudspeaker system.

Table 8:4 presents the Sabine reverberation and STI (BS EN 60268-16:2003) calculation errors in JND when compared against actual measurements taken at R8.

	500Hz	2Khz
EDT	1.9	1.4
RT30	2.2	1.6

	STI (receiver 8)
Measured	0.430
Sabine + IEC 60268-16 :2003	0.410
Discrepancy	0.020
Relative discrepancy (JND)	0.08

Table 8:4 Sabine reverberation time (left table) and STI (right table) calculation errors at R8 in JND units.

The good agreement between measurements and STI calculations based on the diffuse field assumption further verifies the quasi diffuse theory. These results are consistent with the validation and conclusions of Yang (1997).

8.4.2 Volume slice space

CATT predicted EDT and RT30 parameters in the volume slice showed a high spatial uniformity for both frequency octave bands as shown by the negligible deviations between R8 and R9 (figure 8:9 and table 8:2).

Overall predicted SPL deviation between R8 and R9 was 0.2dB which together with the above results provides evidences on the spatial uniformity of acoustic parameters in this space.

The EDT/RT30 ratios presented in table 8:1 showed near perfect decay linearity at both receiver positions for both frequency octaves. These results together with the spatial uniformity seen above are clear indications of the presence of diffuse sound field behaviour in the volume slice.

The discrepancy between calculated reverberation by the Sabine formula and values predicted using the CATT simulation of the space were within 0.7JND for both receiver positions (table 8:5). This agreement further denotes the presence of diffuse field in the slice space.

		500Hz	2Khz
R8	EDT	-0.26	-0.08
	RT30	-0.77	-0.20
R9	EDT	-0.22	-0.04
	RT30	-0.74	-0.20

Table 8:5 Reverberation time discrepancy in JND units between Sabine calculation and CATT predictions at R8 and R9 in the volume slice.

Table 8:6 shows the low EDT, RT30 and STI discrepancies between CATT predictions in the volume slice and predictions in the full platform volume, clearly suggesting the acoustic similarity of a sample slice to the full platform volume.

	EDT (500Hz)	RT30 (500Hz)	EDT (2kHz)	RT30 (2kHz)	STI
Volume Slice -Full Plat 1 (R8) (sec)	0.40	0.21	0.18	0.08	0.01
Volume Slice- Full Plat 1 (R9) (sec)	0.44	0.27	0.21	0.11	-0.01
Volume Slice -Full Plat 1 (R8) (JND)	1.7	0.8	0.8	0.3	0.02
Volume Slice- Full Plat 1 (R9) (JND)	2.0	1.1	1.0	0.5	-0.2

Table 8:6 Absolute and relative discrepancies of EDT,RT30 and STI at both receivers between predictions in the volume slice in the full length platform.

The prediction and Sabine calculation errors of EDT and RT30 parameters at R8 were remarkably low for both frequency octave bands, (table 8:7). These results provided the most robust evidence of the effective presence of quasi-diffuse sound field conditions in the volume slice.

	EDT (sec) (500Hz)	EDT (500Hz) Error (JND)	RT30 (sec) (500Hz)	RT30 (500Hz) Error (JND)
Measured	2.67	-	2.76	-
JND	0.27	-	0.28	-
CATT Prediction	2.69	0.07	2.84	0.29
Sabine RT	2.62	-0.19	2.62	-0.51
	EDT (sec) (2kHz)	**EDT (2kHz) Error (JND)**	**RT30 (sec) (2kHz)**	**RT30 (2kHz) Error (JND)**
Measured	2.41	-	2.47	-
JND	0.24	-	0.25	-
CATT Prediction	2.42	0.04	2.45	-0.08
Sabine RT	2.40	-0.04	2.40	-0.28

Table 8:7 EDT and RT30 predicted, calculated, measured and relative error values.

All the above results confirmed the hypothesis that a slice volume simplification of a long platform equipped with a distributed VA system possesses the same quasi diffuse sound field characteristics observed on the full platform. Those results also confirmed the speculation that the acoustic field of an underground platform equipped with a distributed loudspeaker VA system can be conceptualized as a succession of equivalent local diffuse fields each centered at each loudspeaker and all possessing similar acoustic characteristics to those of the full length platform.

8.5 Conclusions

This chapter has reviewed and verified the quasi diffuse sound field theory proposed by Yang (1997). Related knowledge has been expanded by providing additional and practical findings.

The theory is an important concept in the investigation and practical understanding of the acoustic field created by VA system on underground platforms. It allows the application of the classic acoustic theory as an approximate calculation technique which can become very useful in the efficient design of underground platform VA systems for optimal speech transmission quality.

It was found that the sound field at a given receiver on a platform becomes increasingly more diffuse as more sources are added along the length of the platform. This effect is frequency independent and virtually spatial independent. The gradual increase in the number of sources gradually increase the EDT (but not the RT30) and this has a correlated detrimental effect on the STI scores. The addition of a few sources increases notably the overall SPL at a given receiver however this increase diminish and stabilizes as the number of extra sources is increased.

On an underground platform equipped with a distributed VA system providing high SNR conditions, loudspeakers added adjacent to the nearest loudspeaker to a receiver act as sources of late arriving energy degrading the speech transmission quality.

It has been demonstrated that the theory is a valid basis for the utilization of classic acoustics as an approximation methodology in the calculation of acoustic and speech intelligibility related parameters in underground platforms equipped with distributed VA systems. It has also been shown that the created sound field can be approximated by a succession of equivalent local diffuse fields, centered on each loudspeaker position.

Chapter 9 Conclusions, recommendations and further work

9.1 Introduction

This study has examined the topic of design and optimisation of VA systems in real world underground stations to provide practical tools, knowledge and guidance to design an effective system.

9.2 Overall conclusions

9.2.1 Context of VA underground stations

Despite the critical role of VA systems in subsurface underground stations acknowledged two decades ago, their performance still fails to achieve desired targets. several factors have been identified :

- Lack of preferential consideration of VA systems in the complete station design /refurbishment (prioritisation) over arguments of cost and other station design requirements (e.g. aesthetics and ease of maintenance).
- Lack of awareness and understanding by designers and decision makers of the many and different types factors and constraints involved in the delivery of an effective system
- Inherent acoustic complexity of the spaces where VA systems are installed.

Efficient VA systems can provide additional value when not in use as a life-saving function by assisting passenger flow control, facilitating customer information and providing perception of quality and well being.

9.2.2 Performance specifications and standards

The interrelation among performance parameters and the influence of other non electro-acoustic factors such as operational and system integration constraints make the establishment of clear, suitable and balanced performance specifications a crucial and often neglected task for successful VA designs.

Attempting to meet all performance specifications as laid in current contractual documents can prove unattainable and counter–productive. A critical review has shown that contractual performance specifications were ambiguous, conflicting and occasionally unsuitable. A full revision and update of those documents is recommended with a view at balancing and prioritising specifications if necessary in light of practical experience. A list of specific amendments and generic objectives are suggested by the author for the improvement of those key reference design documents.

A novel design parameter aiming at providing a more relevant and realistic indicator of useful energy coverage (DERL) at the design stage is proposed by the author.

9.2.3 Acoustic baseline of London Underground deep platforms and empirical model development

An extensive database containing geometrical, architectural and acoustic measured data of a large sample (74) of London Underground deep platforms equipped with conventional VA systems has been created from data gathered during site surveys. The database has been statistically analysed to provide a baseline overview on the acoustic and speech transmission quality conditions as well as influencing geometrical and architectural characteristics found on deep platforms.

The analysis has shown that most deep platforms have similar geometrical and architectural characteristics. This is reflected on the concentrated range of similar RT and STI values. Of the geometrical and architectural platform features considered only the presence of ceiling metal cladding showed a significant impact on acoustic and speech transmission quality parameters. This physical similarity is the precursor idea of developing a dedicated simple model (DEPEM-1) to remove the need to systematically take acoustic surveys and utilise commercial acoustic simulation programs to design and predict platform VA system performance in platform of similar conditions.

It was found that the vast majority of platforms surveyed did not achieve the minimum speech transmission quality performance specification for deep platform VA systems (STI ≥0.5). It was shown that over-bridges and the limited cross-passage open area have not significant influence on reverberation or STI parameters (this agreed with findings in chapter 7 and 8). Similarly it was shown that platform RT20 and STI are not sensitive to moderate variations in platform volume.
The database was utilised as the basis for the construction of an empirical simple model (*DEPEM-1)* for the prediction of speech transmission quality (STI) using measured RT as a sole input and under certain applicability conditions. The model was validated against measurements yielding a prediction error below 1 JND. The prediction performance of the *DEPEM-1* was comparable or higher than other simple models or methodologies found in the literature. In addition, it was found that *DEPEM-1* can conveniently use RT input data calculated by the *Quasi-diffuse sound theory* as an useful alternative to measure RT data.

The DEPEM-1 has laid the basis for future developments to become a quick, reliable and flexible prediction tool for the early design stages of VA system on London Underground deep platforms.

9.2.4 Assessment of computer acoustic simulation programs

A validation and comparative evaluation of the two most widely used commercial acoustic simulation programs (CATT-Acoustics v8.g and Odeon v9.2.) in the design of underground platform VA systems has been undertaken for the first time. The study involved five underground stations spaces (cases) of different geometrical complexity.

Overall, both programs demonstrated their ability to predict relevant parameters used in VA electro acoustic design with similar satisfactory accuracy. Prediction errors observed on EDT, RT and STI parameters shown by both programs were within or around the target reference of 1JND for all case studies, except for the platform exhibiting sharp curvature where higher errors were seen for STI (1-2.2JND) and RT (up to 4.7JND) parameters. The sharp curvature of Plat-3 had an impact in the programs prediction quality being more notable in CATT-Acoustics. Odeon processing time was greatly shorter than CATT-Acoustics for all cases.

It was consistently observed that predictions of STI were not sensitive to moderate variations of reverberation particularly at low frequencies bands and this agrees with similar studies in the literature. On all platform simulations it was observed a high spatial uniformity for all predicted parameters resembling the characteristics of a diffuse sound field as suggested by Yang (1997).

Until further development on the existing dedicated simple models, commercial computer programs remain the most accurate, reliable and flexible tool for the design of VA system in underground spaces.

9.2.5 Design variables and impact on performance

An investigation on the impact on the acoustic and predicted speech intelligibility of the most significant variables and design factors in the design of platform VA systems has been carried out.

The loudspeaker array is the only controllable variable in the design. It was found that the spatial variation of all predicted acoustic parameters for seven loudspeakers array configurations tested is remarkably low at all frequency bands supporting the Quasi diffuse sound field theory. It was also shown that different configurations of a distributed loudspeaker system involving different loudspeaker position, aim and density do not affect the reverberation in the platform space.

The sound field on a platform equipped with a distributed VA system is almost completely dominated by reverberant energy, resembling a diffuse sound field.

Variations on the loudspeaker array configuration including different type of wide dispersion loudspeaker, aim and speaker density did not improve significantly the STI score. Only more drastic approaches such as significantly reducing the loudspeaker-receiver distance and the use of highly directional loudspeakers were able to achieve the specified performance target. It was shown that those approaches would reduce the amplifier power requirement by a factor of 6 and 4 respectively involving the corresponding cost saving in related equipment. However these configurations would involve the reduction of the coverage uniformity, perceived sound quality and aesthetics.

It has been shown that speech intelligibility from conventional designs VA systems on deep platforms is practically limited to a maximum achievable by the dominating reverberation condition and uncompromising changes in the loudspeaker array proved ineffective.

Computer simulation investigations have shown the significant increase in reverberation time at high frequencies with temperature and humidity. Consequently speech related parameters were seen to decrease with rising temperatures and humidity values. Variations in humidity had a greater influence on the evaluating parameters than temperature variations. The neglect of temperature and humidity effects on platform total SPL and speech intelligibility predictors could lead to mild prediction errors.

The reduction of reverberation time by increases of scattering coefficients within a representative range of estimated existing conditions were not sufficiently high to influence significantly the speech intelligibility related parameters. This lack of sensitivity to moderate changes in scattering coefficient can be interpreted as a useful tolerance knowledge in the estimation of values when accurate coefficients of platform surface materials data remains unavailable. Nevertheless, a marginal trend in increasing values of STI with greater values of scattering coefficient was observed. In order to substantially increase the surface scattering and thus contribute to the increase of the STI score, some practical approaches are proposed for consideration in the early design or renovation stages.

Acoustic simulations have shown that deep underground platforms are highly sensitive to variations of sound absorption. This makes the application of acoustic treatment the most effective solution to reduce reverberation and allow platform VA systems achieve the specified STI target. A minimum 4% of platform surface area (104m²) of an 30mm thick LU approved acoustic treatment would be required to be applied to a ceiling strip along the platform length to virtually achieve (0.49 STI) the required STI minimum performance target (0.5) without significant SPL attenuation at an estimated total cost of £21K. The platform end walls showed to be a highly efficient location for the application of acoustic treatment and increase of STI. A combination of those two arrangements would ensure STI scores higher that the target at an estimated total cost of £28K. It was observed that the gradual increase of acoustic treatment applied on the platform would not correspond to linear increments in STI scores. A novel acoustic treatment design concept for highly reverberant underground spaces is proposed.

9.2.6 Review and verification of quasi diffuse sound field

In a verification study utilizing CATT-Acoustics computer simulations and actual measurements has been demonstrated that the quasi diffuse sound field theory proposed by Yang (1997) is a valid basis for the utilization of classic acoustics as an approximation methodology in the calculation of acoustic and speech intelligibility related parameters in real world underground platforms equipped with distributed VA systems. It has also been shown that the created sound field can be conceptualized as a succession of equivalent local diffuse fields each centered at each loudspeaker and all possessing similar acoustic characteristics to those of the full length platform.

In this study it was also found that the sound field at a given receiver on a platform becomes increasingly more diffuse as more sources (loudspeakers) are added along the length of the platform. This effect is frequency independent, virtually spatial independent and appeared to level gradually from 19 sources. The gradual increase in the number of sources gradually raised the EDT (but not the RT30) and this had a correlated detrimental effect on the STI scores. The addition of a few sources increased notably the overall SPL at a given receiver however this raise diminished and stabilized as the number of extra sources was increased. RT30 appeared independent to the number of sources introduced along the platform, unlike EDT which was significantly dependent to the the number of sources when this number was very low. It was also generally observed in this and previous investigations that the STI on platforms is well correlated to EDT provided high SNR conditions.

9.3 Recommendations

A list of recommendations arisen from the research work is presented here for consideration in future designs of VA systems in subsurface underground spaces.

i) Review and update current performance specifications to produce clearer and more balanced specification targets and requirements taking into account practical constraints (see suggested amendment list in 3.4.4).

ii) It is recommended that the design team educate at an early stage other stakeholders on the practical and acoustical constrains as well as conflicting requirements so the management can make informed and balanced compromises for a more efficient and effective design delivery.

iii) A novel design parameter, Direct plus Early Reflections Level (DERL) coverage indicator is proposed by the author as a more relevant and realistic indicator of useful energy coverage (see 3.3.4).

iv) Considering the high occupancy density, the significant proportion of users with English as second language and increased likelihood of terrorist attacks, the author based on latest guidance (BS EN 60268-16 :2011) suggests to upgrade the STI performance target to achieve qualification band E (0.56 to 0.6) for subsurface public areas (see 3.4.4).

v) An harmonisation among national and international standards providing VA electro-acoustic guidelines is recommended as well as the provision of complete and clear information on performance test methodologies in particular for objective speech intelligibility measurement instrumentation widely performed with the STIPA method (see 3.4.4).

vi) In the design of new or renovated platform spaces it is recommended where possible to consider the introduction of functional and decorative furniture/hardware//art work, rough textured (decorative) surfaces or dedicated acoustic treatments (see author’s customised conceptual design in

7.3.5) to greatly increase the surface scattering and contribute to a higher speech intelligibility.

vii) Existing platform billboards could be redesigned to act as tuned diaphragmatic and/or micro-perforated cavity absorbers aimed at reducing challenging reverberant and background low frequencies components.

viii) The introduction of acoustic treatment in the design of new or renovated platform spaces is strongly recommended to enable VA systems to perform effectively (see author's customised conceptual design in 7.3.5).

ix) It is recommended to build in loudspeakers into the CMS panels or even into the passenger wall facing so that to reduce the highly critical loudspeaker-receiver distance.

9.4 Further work

A number of further investigations are suggested:

i) Evaluation on the effectiveness of the proposed new design parameter DERL

ii) Extension of *DEPEM-1* model database with the addition of more measured data entries to increase prediction accuracy and applicability range (e.g. metal cladding ceiling or flat ceilings platforms). Commercial simulation models could be used to further improve the model and expand the applicability range.

iii) Investigation on the validity and accuracy of the blind estimation of RT technique (Kendrick et al, 2007) on deep platforms to provide an alternative and convenient source of calculated RT input data for the *DEPEM-1.*

iv) Extension of the assessments to include: other challenging underground spaces interconnecting concourses, stair cases, flat ceiling platforms and acoustically coupled areas.

v) Investigation into the determination of a larger range of programs' operational settings aimed at reducing run time while maintaining prediction accuracy.

vi) Development and evaluation on the cost-effectiveness of the proposed acoustic treatment design concept.

vii) Development of auralisations as an efficient communication and subjective evaluation tool of design changes on platforms (e.g. the dramatic benefits of the introduction of acoustic treatment).

viii) Investigation into the effect of platform occupancy (space volume reduction, absorption and scattering variations, total SPL reduction, occupancy noise) on STI.

ix) Investigation into the potential benefits of flat panel sound radiators on cost, speech intelligibility and coverage uniformity of flat sound radiators (figure 9:1).

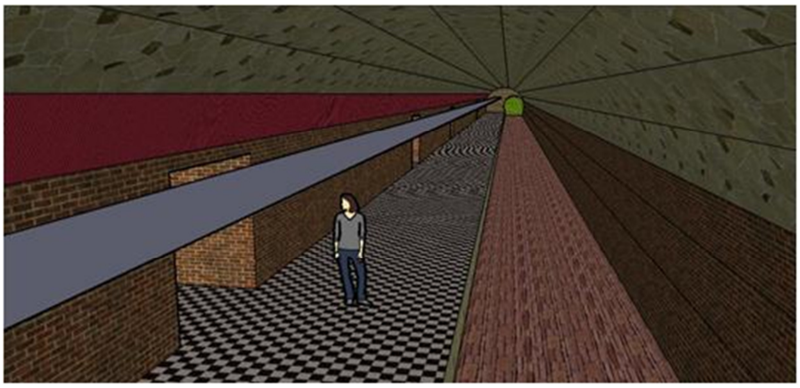

Figure 9:1 Suggested flat sound radiators application. Two suggested locations: on wall (red plane) and hanged from ceiling (grey plane).