Improving Television Sound for People with Hearing Impairments

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Key to Abbreviations Used

AAC	Advanced Audio Coding (audio codec)
AC-3	Dolby Digital (A/52) audio codec
AHRL	Age Related Hearing Loss
BSI	Bit Stream Information
DD+	Dolby Digital Plus
DRC	Dynamic Range Control
DVB	Digital Video Broadcasting
DVB CM-AVC	Digital Video Broadcasting Commercial Module - Audio Video Coding
E-AC-3	Enhanced AC-3 (Dolby Digital Plus codec)
EBU	European Broadcasting Union
ETSI	European Telecommunications Standards Institute
FRN	FascinatE Render Node
HDTV	High Definition Television
HE-AAC	High-Efficiency Advanced Audio Coding (audio codec)
HI	Hearing Impaired
HoH	Hard of Hearing
HTML5	Hypertext Markup Language v5
IP	Internet Protocol
IPTV	Internet Protocol Television
ITC	Independent Television Commission
ITU	International Telecommunication Union
LoRo	Left only, Right only (stereo downmix)
Lt/Rt	Left total, Right total (stereo downmix)
MPEG	Moving Picture Experts Group
MUSHRA	MUltiple Stimuli with Hidden Reference and Anchor
NICAM	Near Instantaneous Companded Audio Multiplex
NIHL	Noise Induced Hearing Loss
PCA	Principal Components Analysis
PTA	Pure Tone Audiogram

S/PDIF	Sony/Philips Digital Interface Format
SPIN	Speech Perception In Noise
SPL	Sound Pressure Level
STB	Set Top Box
UKCAF	UK Clean Audio Forum
WDRC	Wide Dynamic Range Compression

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Abstract

This thesis investigates how developments in audio for digital television can be utilised to improve the experience of hearing impaired people when watching television.
The work has had significant impact on international digital TV broadcast standards; it led to the formation of the UK Clean Audio Forum whose recommendations based on the research have been included in ETSI international standards for digital television, adopted into ITU standards for IPTV and also into EBU and NorDig digital television receiver specifications. In this thesis listening tests are implemented to assess the impact of various processes with a phantom centre channel and with a centre loudspeaker. The impact of non-speech channel attenuation and *dynamic range control* on speech clarity, sound quality and enjoyment of audio-visual media are investigated for both hearing impaired and non-hearing impaired people. For the first time the impact of acoustical crosstalk in two channel stereo reproduction on intelligibility of speech is quantified using both subjective intelligibility assessments and acoustic measurement techniques with intelligibility benefits of 5.9% found by utilising a centre loudspeaker instead of a phantom centre. A novel implementation of principal component analysis as a pre-broadcast production tool for labelling AV media compatible with a clean audio mix is identified, and two research implementations of accessible audio are documented including an object based implementation of clean audio for live broadcast that has been developed and publicly demonstrated.

1. Introduction

1.1. Research Aim

A significant proportion of any population suffers from some form of hearing loss. This has an impact on the ability of people to separate speech from competing sources therefore there is a need to develop means of improving this groups' quality of experience of broadcast programming.

This thesis sets out to develop such algorithms and methods, to assess their effect and investigate if it is possible to take advantage of the increased broadcast of surround sound and utilise the additional data associated with surround sound to improve television sound for hearing impaired viewers.

1.2. Thesis Structure

The origin of this thesis can be found in the Clean Audio Project (funded by the Independent Television Commission (ITC)) although this work has been continued and developed considerably beyond the remit of the original project including research funded by Dolby Labs and Ofcom with the same aim. The thesis presents research from the initial project and from continuing research into the problems associated with TV audio for hearing impaired people and investigates a number of possible solutions. For this reason this thesis does not follow a traditional PhD thesis structure but experimental work is instead presented in several phases with each containing results, analysis and discussion leading to the next phase.

Chapter one defines the problem that was investigated, identifies areas in the broadcast chain where solutions may be implemented and presents the contribution that this research has made into international broadcast standards, It also outlines the contribution of the author and the contribution of others in some of the research carried out.

Chapter two reviews the research to date across a range of fields related to the research carried out for this thesis. This includes critical analysis of previous work directly related to TV audio for hearing impaired people, and relevant material on hearing impairment, clarity and speech intelligibility.

Chapters 3, 4 and 5 present the methods, analysis and findings of the three main phases of the experimental part of the research.

Chapter 3 documents the first phase of the ITC funded Clean Audio research project, which led to the formation of the UK Clean Audio Forum[1] (UKCAF) and liaison with international broadcast standards bodies. Test methodology is developed for assessment of audio reproduction conditions using AV media for hearing impaired people. Experiments investigate the impact of *non-speech channel* attenuation, and of stereo downmixing, on ratings of speech clarity, overall sound quality and enjoyment for hearing impaired and for non-hearing impaired people.

In chapter 4 further research is presented that implements solutions identified in chapter 3 for existing two channel stereo reproduction equipment.

Chapter 5 investigates the poor ratings of stereo in listening tests carried out in chapters 3 and 4 and documents both subjective assessments and objective measurements of the impact of acoustical crosstalk on intelligibility of speech in background noise in order to identify possible causes of these ratings. The chapter uses audio-only test material in order to investigate the impact of a centre channel for speech on intelligibility based on key word recognition.

In chapter 6 a solution proposed in response to published research documented in chapter 3 (utilising principal component analysis to separate speech from noise) is evaluated for fitness of purpose. The technique is then adapted for offline usage indicating some usefulness for preprocessing broadcast material prior to broadcast.

Chapter 7 documents two implementations of variations on the clean audio solutions proposed here. Firstly an experimental process developed by Dolby Labs was assessed by the author using listening tests, and secondly a solution for accessible audio was developed and demonstrated for an object-based future broadcast system as part of the EU FP7 FascinatE project.

[1] The UK Clean Audio Forum was sponsored by the UK communications regulator, Ofcom, and consisted of broadcasters and broadcast technology providers. Members included, Dolby, SVT, S4C, BSkyB, CH4, ITV, BT, ST Microelectronics, BBC, University of Salford.

Chapter 8 presents conclusions and future directions that have been identified for continuing the research.

1.3. Contribution of the Author

Some of the experimental work documented in the thesis was carried out with the assistance of additional researchers. This section clarifies the contribution of people contributing to the work. All literature reviews, background research and all text presented in this thesis is, however, the sole work of the author. In each instance all decisions relating to the research including test methodology and implementation were made by the author. In each instance listening tests and statistical analyses were carried out by the author, where research assistants were employed to assist in tests this is identified here for clarity.

For the initial work on the project, documented in chapter 3, a research assistant (Paul Kendrick) was employed, primarily to assist in the execution of listening tests. Tests were designed, all control software developed, and the statistical analysis presented here was carried out solely by the author. Listening tests investigating stereo reproduction, documented in chapter 4, were carried out by the author with the assistance of another research assistant (Claire Churchill) who also assisted in identifying appropriate test material based on criteria set by the author. All test design, control software used and the statistical analysis presented here is the sole work of the author. Some experimental measurements in chapter 5, investigating the impact of acoustical crosstalk, were carried out with the assistance of a research assistant (Paul Kendrick) who also developed a comb filter, implemented as a DirectX plugin, for listening tests. Listening tests for Dolby Labs, documented in chapter 7 were designed by the author although methods for selecting test material were developed collaboratively with Hannes Musch from Dolby Labs, San Francisco. Processing for these tests was carried out using experimental software developed by Hannes Musch. All statistical analysis and control software were the sole work of the author. For the FascinatE project (documented in chapter 7) clean audio implementation research was carried out solely by the author, software development for audio object extraction was carried out by a research assistant (Rob Oldfield) under the direction and supervision of the author.

1.4. Defining the Problem

There are estimated to be around 10 million people who are deaf or hard of hearing in the UK . Of these around 8.3 million suffer from mild to moderate deafness (Hearing Concern, 2004) and would benefit from any improvements that may be made in television audio. The ITC received many complaints from hard of hearing people about the quality of sound on television, primarily that the dialogue is unclear and hard to understand owing to the level of background ‘noise’. This noise consists of background music, sound effects and speech and it can have the effect of masking the dialogue and making it difficult or impossible to understand. This level of complaints to the ITC and Ofcom has been mirrored in complaints to broadcasters. A survey carried out by the BBC in 2010 indicates that 60% of viewers had difficulty in understanding speech on TV (VLV, 2011). Digital TV, and especially the increasing availability of surround sound, has the potential for much improved TV sound quality and could therefore be of great benefit to hearing impaired viewers. However the wish to create “rich multilayered soundtracks” may instead lead to increased problems for hearing impaired people (Armstrong, 2011) .

1.5. Server Side or Client Side Solution

An important element in the planning of any research into broadcast accessibility solutions is defining the appropriate point, or points, in the route from producer to viewer at which change should be implemented. Changes in appropriate legislation, recommendations and guidelines can be implemented with a ‘top down’ approach; this can be carried out at an international level and so retain or improve compatibility between the broadcast systems of different countries. Standards committees and professional bodies can be influential in bringing accessibility and inclusivity issues to the fore and in promoting solutions. In collaboration with the major international audio and broadcast companies they are responsible for publishing the standards to which all of these companies should comply. Broadcasters themselves can play a key role in ensuring mixes are appropriate to the programme and avoiding exacerbation of the problem by poor mixing practice.

An alternative approach is to bring about improvements in the set top box (STB), at the viewers' end of the broadcast chain. This approach may be capable of providing more in the way of a 'quick fix' solution; an add-on to a set top box could perform appropriate audio processing and could be retro-fitted to existing equipment. STB manufacturers can re-programme the software of much current receiver and decoder hardware and there is also potential for solutions that would involve viewers altering settings and choosing equipment based on its accessibility to them and on their individual needs. Much is possible in this domain but it is sometimes difficult to persuade industry to commit development funds to benefit what is often seen as a niche market.

1.6. Digital TV and Surround Sound Broadcast

One of the features of digital audio broadcast is the capability of a far greater dynamic range than was possible with analogue broadcast, the difference in level between the quietest sounds and the loudest can be far greater. This capability is being utilised to the full by producers, not least because more and more viewers are listening to their TV sets connected to hi-fi or home cinema equipment which can cope with reproduction of a greater dynamic range than built in TV loudspeakers. This increase in dynamic range has obvious implications for viewers suffering from loudness recruitment[2]. In loudness recruitment the difference in level between the quietest sound that can be heard and the level at which sound level becomes painful is reduced making increased dynamic range in broadcast uncomfortable or painful for sufferers. Increased use of louder music, effects and other background noise can make understanding of speech much more difficult for a range of hearing impairments.

[2] *"Recruitment is the otological condition in which weak sounds are not heard while strong sounds are heard as loudly as by a normal ear… in recruitment the dynamic range of hearing is narrowed"* Yost, W. A. (2000). *Fundamentals of Hearing: An Introduction.* San Diego, California: Academic Press.

Alongside the implementation of digital TV, although not yet ubiquitous, is the continuing growth of surround sound broadcast for television. The most widely used surround sound format for digital TV in the UK is currently Dolby Digital 5.1 surround sound. At the heart of the Clean Audio project was the premise that by utilising the presence of additional channels and the extra information contained within the metadata of the Dolby Digital format it should be possible to improve the clarity of TV sound for hard of hearing viewers.

1.7. Possibilities Offered by Surround Sound Broadcast

Surround Sound Broadcast offers a number of potential solutions to create 'clean audio'. There is additional audio data and there is additional data about the audio data (metadata). These may both be utilised in an attempt to improve dialogue clarity and speech intelligibility.

1.7.1. Dolby Digital Surround Sound

Dolby Digital 5.1 is the format chosen by Sky for their current surround sound broadcasts in the UK and, with around 28 million Dolby Digital receivers in use throughout the world, Dolby Digital looks set to continue as a market leader for surround sound TV broadcast (Rumsey, 2009b). Because this is the most common format in the UK this thesis focuses on Dolby Digital however the same potential for solutions exists with other codecs such as AAC and HE-AAC.

The Dolby Digital format minimises bandwidth by using data compressed audio and allows for the use of multiple full frequency range audio channels and one low frequency effects channel. Loudspeakers for 5.1 are arranged with one central front channel (normally used for dialogue), front left and right loudspeakers and rear left and right surround loudspeakers arranged as shown in figure 1 and defined in ITU-R BS 775-1, *Multichannel stereophonic sound system with and without accompanying picture*, (ITU, 1994).

The audio is broadcast as an AC-3 or E-AC-3 (Advanced Television Systems Committee, 2012) bit stream and it is the format and content of this bit stream that may enable implementation of changes that could be beneficial to hard of hearing viewers. The original Dolby Digital AC-3 standard has now been superseded by the *enhanced* AC-3 format (E-AC-3), also known as Dolby Digital Plus (DD+). E-AC-3 incorporates a number of enhancements to the AC-3 standard. The changes include, “increased flexibility, expanded metadata functionality, and improved coding tools” (Fielder et al., 2004) including the provision of up to 15 full range audio channels instead of AC-3’s 5 full range channels.

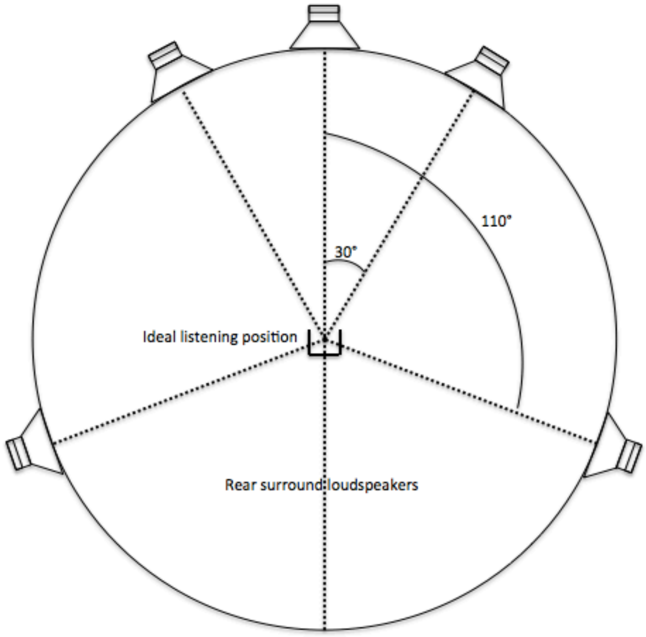


Figure 1 Loudspeaker layout compatible with ITU-R BS 775-1, Multichannel stereophonic sound system with and without accompanying picture,

For the purposes of this thesis, and as it is only concerned with investigating 5.1 channel delivery, the term AC-3 will be used throughout on the understanding that, for current surround sound broadcast using 5.1, this could be delivered using either AC-3 or E-AC-3 and that, for the purposes of 5.1 broadcast and the scope of this thesis the difference is unimportant.

The AC-3 bit stream consists of between 1 and 6 discrete channels of audio, and metadata. AC-3 metadata can be described as data about the audio data and a full description of defined metadata can be found in Appendix A, Dolby Digital Metadata Parameters and in *The Dolby Metadata Guide issue 2* (Dolby Labs, 2003). The audio is compressed in the encoding process and AC-3 streams of various bit rates encompass multi-channel and single channel formats. Additional audio channels can be included for multiple language support and there is provision to include Hearing Impaired (HI) and Visually Impaired (VI) audio channels for viewers with sensory impairments. Accompanying metadata contains information about all of these audio channels; their format, how they should be decoded, downmix parameters required to convert from 5.1 to stereophonic or mono-aural and the type of audio compression that should be applied (if any), based on a proprietary compression system, *dynamic range control,* which is discussed in more detail later in this thesis.

Unlike some surround sound systems, the AC-3 format maintains a separation between audio channels in the encoded bit stream, in other words, there are 6 discrete and separate audio channels present in a 5.1 encoded AC-3 stream. This in itself means that relative channel levels can be altered relatively simply at the decoder and individual channels can be attenuated or amplified independently, often this can be accomplished by changes in metadata interpretation. In most 5.1 encoded material the centre channel is used as a dialogue channel therefore gains in dialogue clarity should be possible by attenuating the level of the left, right and surround loudspeakers relative to the dialogue, at least for material where this would be appropriate.

1.7.2. Multi-channel Audio

The first solution to be investigated in the project was the simplest; Dolby mixing guidelines state that, *"Traditionally, dialog is placed only in the Centre speaker to tie the on-screen sounds to the picture. When a Centre speaker is used, all centre-panned dialogue appears to come from the screen regardless of the listener's position. If the dialogue comes from the Left or Right speakers, the stereo image differs depending on the listener's position. This is highly undesirable. It does not bar voices from the other channels, but generally only effects or incidental voices should be in any channel other than centre.*" (Dolby Labs, 2005)

Therefore, with a few notable exceptions[3], in the vast majority of material that includes speech content and implements Dolby Digital Surround Sound the entirety of the dialogue resides in the centre channel and, for the viewer at home, is reproduced from a loudspeaker very close to the television screen. Almost all sound effects, music and other peripheral audio is contained within the left and right front channels (reproduced from the front left and right loudspeakers) and in the rear surround channels (reproduced from the rear left and right loudspeakers). A notable exception to this is that in some film sound the centre channel is also used to convey sound effects that are synchronous and 'attached' to objects on the screen. Usually these effects are Foley[4] although there are other fairly rare examples of other sound effects in centre channel when they are important to the understanding of the visual scene or where it is critical that the effect is anchored to the cinema screen. As mentioned earlier it should be possible to make the dialogue clearer by reducing the level of the left, right and surround channels relative to the dialogue channel although the effect of this on the enjoyment and on the perceived sound quality for both hearing impaired and non-hearing impaired people was unclear. Given that most television sets are shared by several members of a household the impact on both groups is not an inconsequential factor. It is critical that improvements for one person are not to the detriment of other users. Details of an investigation into this potential solution were the subjective of the first phase of the Clean Audio Project and are covered in chapter 3 of this thesis. Further work assessing the impact of this process for two channel stereo reproduction are documented in chapter 4.

1.7.3. Hearing Impaired (HI) Audio Channel

The AC-3 stream has the capability to contain an audio channel intended as an aid to hard of hearing people. The HI channel is intended to be used as a single mono-aural audio channel containing only dialogue processed so as to make it more intelligible for hearing impaired viewers. Other than a statement that the HI channel should contain

[3] Examples from television sound that illustrate these exceptions are identified in later research documented in chapter 6 of this thesis.

[4] Foley, named after the American film pioneer, is the name given to the process of adding live or synchronous sound effects such as footsteps, rustling of clothes etc to film sound.

processed dialogue there is no available guidance as to how this improved intelligibility should be gained (Dolby Labs, 2000a). It was hoped that the Clean Audio Project could bring some clarity to this subject and that this may be of benefit in applications beyond the narrow 'broadcast' scope of the project, such as DVD production where bandwidth is not as much of a limiting factor. In the broadcast environment however bandwidth is severely limited, every bit of data has an associated cost, and a separate audio feed for hearing impaired people has not provided a solution that has been taken up by broadcasters. In this research a decision was made to concentrate on solutions that would not increase the bandwidth and therefore would be relatively cost neutral for broadcasters. It seemed likely however that any solution that can be delivered by the project as a real time process would also be useful in automatically generating appropriate audio for an HI channel that could be utilised in circumstances where bandwidth was less of a constraint.

1.7.4. Metadata

In addition to the additional audio channels available in the AC-3 and E-AC-3 formats the Dolby bit stream also contains information about the audio. In addition to format, codec and channel information this metadata is also concerned with performing three main functions:

- Allowing changes between TV programmes and TV channels with no sudden changes in level.
- Controlling the downmix of the 6 channels in 5.1 surround for stereophonic or mono-aural reproduction.
- Determining how the programme material is compressed for playback in less than ideal listening environments when appropriate options are set by users. An example of this is so-called 'midnight mode' (Dolby Labs, 2000b) used to reduce the dynamic range of audio to avoid disturbing neighbours late at night.

The first of these, ensuring consistent volume levels between programmes, is accomplished by the use of a value within the metadata that gives an average level based on the level of the dialogue in the programme material. This value, known as the dialogue normalisation level, or *dialnorm*, provides a reference in order that broadcasters can ensure a standard level between programmes and between channels. This reference level is based on a dialogue level measured using Dolby's proprietary LM100 loudness meter, rather than on the level of the audio content overall. The second metadata function is to describe how the six channels of 5.1 audio media should be downmixed for reproduction over a smaller number of loudspeakers. The capability to downmix the 5.1 surround audio to stereo or mono is vital in a broadcast context in order that material can be played back on non-surround reproduction systems without requiring additional audio channels to be broadcast.

The metadata contains parameters that determine the level of rear surround channels compared to the dialogue channel and also the relative level of front left and right channels. The information contained within the metadata is known as the Bit Stream Information (BSI) or the Extended Bit Stream Information depending on whether some more recent optional parameters are implemented. It seemed likely that metadata contained within the AC-3/E-AC-3 stream could have potential to help provide a solution with no extra bandwidth required for broadcasters. Any processing or downmixing implemented at the STB end of the broadcast chain could potentially be controlled by values in the metadata that would be set at the broadcast or production end of the chain.

This potential was explored more fully in phase 2 of the Clean Audio Project during discussions with Dolby Labs and is documented in chapter 4. The use of metadata to improve sound for hearing impaired viewers, and particularly the use of the *dialnorm* parameter, relies heavily on producers and broadcasters using the metadata appropriately and the extent to which they comply is discussed in section 2.3 of this thesis.

A complete list of metadata parameters for Dolby AC-3 and E-AC-3 is contained in Appendix A.

1.8. Contribution of the Research into Broadcast Standards

Standardisation activity beyond the original project has included presentations by the author to the Digital Video Broadcast Group Commercial Module (DVB CM-AVC) in Geneva[5]. The Commercial Module of this group has the responsibility to develop commercial requirements for audio/visual multimedia services both in broadcast and network contexts. Documented outputs from DVB CM-AVC resulting from this presentation are presented in Appendix M (Sheppard, 2006) (*CM-AVC0084 DVB CM-AVC Commercial Requirements: Delivering "Clean Audio" for the Hard of Hearing*). In this document DVB CM-AVC clearly identify the need for clean audio provision and, referencing research documented in chapter 3 of this thesis, present a system diagram developed by the author and others on the UKCAF.

The research presented here led directly to the formation of the UKCAF, and recommendations on Clean Audio provision stemming from this research were presented by UKCAF to the International Telecommunications Union (ITU) (UK Clean Audio Forum, 2007) (reproduced in Appendix K). The recommendations from UKCAF have been published in ETSI TS101154, *Digital Video Broadcasting (DVB); Specification for the use of Video and Audio Coding in Broadcasting Applications based on the MPEG-2 Transport Stream*, (Annex E.4 - Clean Audio) (ETSI, 2009) standard for digital broadcast.

The ETSI Clean Audio provision, as defined by the research documented here, is referenced as a requirement in *Open IPTV Release 2 Specification, Volume 2 – Media Formats [V2.0]* (Open IPTV Forum, 2011) for broadcasts where Clean Audio is provided. These recommendations are now being implemented by some European broadcasters. For example the recent implementation of the NorDig specification for

[5] *"The Digital Video Broadcasting Project (DVB) is an industry-led consortium of around 250 broadcasters, manufacturers, network operators, software developers, regulatory bodies and others in over 35 countries committed to designing open technical standards for the global delivery of digital television and data services.*Digital Video Broadcasting Project. (2011). *DVB - The Global Standard for Digital Television*. URL: http://www.dvb.org/index.xml.

Nordic digital TV which makes Clean Audio support as specified by ETSI TS101154 mandatory for all NorDig profiles where STBs are capable of implementation (NorDig, 2013). The EBU standard that specifies "the minimum HDTV receiver requirements of EBU members (the broadcasting organisations)" (EBU, 2009), also now explicitly requires Clean Audio as defined by ETSI TS101154 for HDTV receivers.

1.9. Contribution to Knowledge

In addition to contributing to international broadcast standards the research documented here has contributed the following to knowledge in the field of broadcast audio for people with a hearing impairment.

Improvements in perceived speech clarity, sound quality and enjoyment of video clips for hearing impaired people have been identified as a result of simple remixing of 5.1 surround sound audio-visual media that could be carried out at the STB, potentially by the use of metadata to flag content appropriate for remixing.

5.1 surround sound material downmixed to stereo has been shown to lead to poorer speech clarity, sound quality and enjoyment for both hearing impaired and non-hearing impaired people when compared to three channel (left, centre, right) reproduction.

Issues with stereo reproduction have been identified and investigated, listening tests have been devised that confirmed statistically significant reduced intelligibility for material utilizing a phantom centre channel compared to a real centre channel based on keyword recognition. Acoustic measurements have been carried out that confirmed this to be the result of attenuation of frequencies key to intelligible speech due to acoustical crosstalk.

Dolby's *dynamic range control* compression processing has been shown that it can significantly improve speech clarity and overall sound quality for hearing impaired people when compared with the default stereo reproduction found on set top boxes.

The potential of object based audio to provide accessible audio has been demonstrated and novel methods for extracting audio objects from a complex acoustic scene demonstrated.

2. Literature Review

There is limited previous research into television sound for hard of hearing people other than that focussing on the use of subtitles and other non-audio cues. For this reason the existing research in that area has been supplemented by relevant work from a variety of subject areas which could inform the research. The review is presented in several sections: a section on the research that *has* been carried out on TV audio for hearing impaired people, sections on hearing loss, lessons that could be learned from hearing aid processing development, the impact of TV audio standards and discussion of various common standards and formats of relevance to the research including downmix methods, and more general work on speech intelligibility and clarity. A section is also presented on the psychoacoustic test methodologies adopted in this research and the particular problems of adapting listening tests and subjective assessment for hearing impaired participants.

2.1. TV Audio for Hearing Impaired People

Considerable research has been carried out into usability issues for digital and interactive TV for older and sensory impaired people including earlier research commissioned by the ITC (Freeman et al., 2003) and several qualitative methodologies have been developed for this purpose (Eronen, 2006; Freeman and Lessiter, 2003). A limited amount of research has been carried out specifically about TV audio for hearing impaired people, principally by the broadcast community; the ITU have questioned how TV sound could be made better for hearing impaired people (1994b) and studies have been carried out by the BBC. A BBC study by Mathers (1991) responded to complaints from viewers over a number of years complaining about “background sound (e.g. audience laughter, crowd noise, background music etc.)”. Mathers’ research carried out subjective testing where participants were presented with three different levels of background sound; a reference mix between speech and background thought appropriate by a BBC mixing engineer and mixes with background sound at -6dB and +6dB relative to this recording. The research was carried out prior to the introduction of NICAM stereo in 1991 and the audio was reproduced using whatever equipment participants normally used while watching TV. It is assumed because of this that the reproduction systems used were monaural. 336 participants in total were asked to assess the perceived intelligibility of speech in excerpts across a range of programme material however only 25% of these were tested under controlled conditions. Results from the research were inconclusive and led to an expressed view that intelligibility was not *highly* dependent on background level and that very high or very low levels of background sound would be needed for a significant impact to be shown.

A second BBC study by Meares (1991) suggests that multichannel sound systems associated with HDTV could be utilised in providing a Hard of Hearing (HoH) channel, at least for programming where a discrete commentary channel exists, and potentially for other programming where access to the original mixes were available. Meares suggests that hard of hearing listeners would benefit enormously from such a provision but identifies additional cost in broadcasting additional HoH services. An additional HoH channel is identified as being an ideal solution by others (Hoeg and Lauterbach, 2009) however where only the premixed audio is present there is no clarity as to how a clean dialogue channel could be derived from material that is already mixed, as is more usually the case.

More general research on audio processing to improve intelligibility can be found in Armstrong's useful literature review (Armstrong, 2011) which points out the considerable difficulties inherent in separating speech from competing noise. Hu and Loizou are cited as carrying out three studies investigating the effect of speech enhancement algorithms on intelligibility, as opposed to speech quality (2007b; 2007a; 2006) however these studies are based on single channel separation methods. Other research (Kendrick and Shirley, 2008) illustrates several implementations of algorithms that provide some separation for multiple microphone configurations. Furthermore chapter 6 and Zielinski (Zielinski et al., 2005) document processing that can separate sources for *produced media* under certain specific conditions. Armstrong's unequivocal conclusion that, "Whilst audio processing can be used to create cosmetic improvements in a speech signal it cannot be used to improve the ability of an audience to follow the words" is therefore rejected. It is possible that this may be an accurate statement for a single channel or for a two channel stereo condition, although this is a very active research topic and some success is claimed using knowledge of individual person's speech frequencies, for example (Stark et al., 2011). However for some multichannel conditions the research presented in this thesis shows that actual intelligibility improvements can be reliably demonstrated under experimental conditions from *produced* multichannel media. The further assertion that "Audio processing cannot be used to create a viable 'clean audio' version for a television audience" is also rejected as it is based on the same argument. It is possible that a surround sound, or even stereo mix, where mixing parameters are not simply the result of source and microphone placement but are produced subject to known guidelines and conventions, may present a special case where sufficient research had not at that time been carried out for a conclusion to be drawn.

The methodology and detailed results of the BBC's own large scale survey into hearing impairments and TV audibility is unpublished however some key outputs have appeared in news releases (VLV, 2011). One of the headline findings was that 60% of viewers "had some trouble in hearing what was said in TV programmes". Background noise and background music accounted for 25% of these, other major factors being foreign language and dialects, poor diction and speech being 'too fast'. The survey led to a series of guidelines and training materials for the BBC in order to alleviate problems as far as was possible through improved production techniques (BBC, 2011).

Over a number of years RNID research published in their annual survey report has held background noise accountable for a higher proportion of problems with dialogue on TV than BBC research suggested. Reports indicate that the number of people finding that background noise affected their ability to hear speech on TV rose from 83% of respondents in 2005 (RNID, 2005) to 87% in 2008 (RNID, 2008). The problem was worse for older people with 88% of the over 65 age group reporting problems compared to 55% of those aged 16-24. Interestingly 45% of those surveyed who had no reported hearing loss also noted background noise as affecting their ability to hear speech

(RNID, 2005) indicating that different mixes, rather than the use of subtitles, is more likely to be a useful solution for many.

Some research has been carried out directly aimed at improving intelligibility of speech on TV audio for hearing impaired people, some of it following recommendations made after research documented in this thesis.
Early work by Carmichael (2004) on the DICTION project indicated that, at the time of the research, although signal processing could make speech sound clearer, it could not improve measures of objective intelligibility in terms of word recognition. Müsch (2008) has argued that this can still reduce the cognitive effort required for comprehension and has discussed algorithms developed by Dolby which utilised several techniques to detect the presence of speech in centre channel and to attenuate other competing sounds in that, and other, channels. The aim of the techniques used was twofold; to decrease listener effort and, as a consequence, to improve intelligibility. Müsch explains that older listeners tend to prefer higher listening levels to younger listeners because of elevated hearing thresholds but also that the effect of loudness recruitment reduces the level at which listening becomes uncomfortable or painful. There is therefore what Rumsey refers to as a reduced “window of comfortable dynamic range” (Rumsey, 2009a) for older listeners. Müsch argues that the cognitive load caused by the mental processing used to filter out background sound and ‘clean up’ the speech means that there is reduced attention for the higher level cognitive processing used to contextualise sentences and therefore fill any ‘gaps’ caused by words not heard. He suggests that the problem for older people in understanding speech on TV is not usually cognitive impairment but is primarily of a sensory nature. Reduced frequency resolution affecting the recognition of speech formants is cited as one reason, another being reduced ability of hair cells in the inner ear to detect phase effectively. The argument here is that audibility is key, that signal processing may not be able to improve individual word recognition but may be able to reduce listening effort and therefore the cognitive load that may play a part in preventing comprehension for hearing impaired people. Others, cited by Carmichael (2004), have argued that there are more factors at work than simply the sensory impairments themselves. Cervellera (1982) points to age

related degradation of nerve pathways as adding significant ‘noise’ to perceived sounds; Stine et al (1986) and Rabbit (1991) point to evidence that slower and less efficient cognitive systems resulting from the ageing process also add to the problem. Certainly the combination of high dynamic range audio, competing background noise, reduced comfortable dynamic range, lack of frequency resolution and other effects brought on by physiological change, degraded nerve pathways and reduced, or slowed, cognitive performance explain why older viewers may find it difficult to understand speech on TV and also the number of complaints received by Ofcom and television broadcasters.

Methods developed during work carried out by Uhle et al (2008) as part of the European project “Enhanced Digital Cinema” (EDCine, IST-038454) aimed to reduce background noise in the centre channel and so improve speech quality however the methods used, based on single channel separation had limited success.

The DTV4ALL project, funded under the EU ICT Policy Support Programme, was set up to “facilitate the provision of access services on digital television across the European Union” and looked at accessible audio for hearing impaired people as part of this remit. It recognised that much accessible audio would ideally be produced at production stage where original unmixed audio was available (thus avoiding the difficult speech separation problems discussed by Armstrong) and highlighted Dolby Digital Plus (DD+) (ATSC, 2005)(Fielder et al., 2004), which has the capability to mix additional audio channels into a 5.1 mix, as being a possible useful implementation platform. The project specifically excluded the unmixing of stereo and mono broadcast audio from its scope but did suggest making use of the fact that in much 5.1 audio, speech is primarily in the centre channel. In tests on material with no separate speech channel a panel of 18 participants rated processed and unprocessed AV test material with a “normal level of background noise” including soft rumbling, rustling and footsteps for ‘audibility’. The nature of background sound was carefully chosen based on pre-test results that indicated poorer acceptance of the clean audio processes used for louder non-speech content, such as cheering. The processing was implemented using the Cedar 1500 processor which enables attenuation of spectral components not utilised by speech. The processor requires setting up for each acoustic situation and it had to be carefully monitored to ensure effective operation (Brückner, 2010) and so, although it was working in real time, it was essentially a manual process used to demonstrate the current state of the art in the context of clean audio. Some additional manual processing was carried out by the iZotope (iZotope) processing tool in order to remove additional piano tones from some sections as the Cedar software was found to be insufficient for this material. Results indicate that ratings were quite variable both across media clips, and across different participants. In a number of cases where participants were able to differentiate between unprocessed and clean audio sections, very varied ratings were obtained. However generally the clean audio processing achieved high ratings. It was concluded that demonstrating improvements for some individuals could indicate a basic principal of providing an optional audio presentation at the set top box for those people. It was anticipated that this mix would be part of broadcast preparation and would be provided as a separate clean audio channel. The DTV4ALL project concluded that "clean audio is a very good solution" but that there was not currently a clear understanding of the difficulties of delivery. Investment in clean audio was considered desirable "particularly if clean audio could be generated automatically" (DTV4ALL, 2010).

A number of researchers have suggested that clean audio could be provided via an IP link in parallel with broadcast and a NEM position paper (MEUNIER et al., 2012) suggests that this could be undertaken as part of individualisation and personalisation of connected TV services. Although bringing its own potential synchronisation issues this is a potentially attractive solution although beyond the scope of this thesis.

Very recent developments by Fraunhofer published during the writing of this thesis have utilised a 'dialogue enhancement' algorithm that pre-prepares material for broadcast in such a way that it can be unmixed to two individual components at the set top box. In this implementation dialogue enhancement parameters are generated as part of the mixing process and these contain sufficient information to effectively derive original sources from the mixed AV media at the STB. Instead of transmitting separate speech and background, an AAC or HE-AAC bitstream is transmitted that contains mono, stereo or 5.1 mixed content. Metadata containing the unmixing parameters required to separate out sources required to create a clean audio output are transmitted as part of this transmission. The advantage to this solution is that it could be made backwards compatible with existing equipment, where no decoder was present in the set top box, the default stereo or 5.1 mix would be heard. This solution was demonstrated for two channel stereo material as part of BBC Wimbledon coverage (Fuchs et al., 2012) and viewers were able to use a PC based software application to adjust levels of commentary compared to court-side ambience. Although the process brought some additional complexity to the production process audience response was favourable. The technology is further described in a paper not published at the time of writing (Fuchs and Oetting, 2013) as an implementation of MPEG SAOC (Spatial Audio Object Coding) where the dialogue or commentary is considered as an audio object which can be extracted from the pre-mixed audio based on the parameters transmitted with the broadcast audio mix.

2.2. Hearing Loss and Audio Reproduction

It is important to define at an early stage what types of hearing loss may be addressed by the potential solutions discussed within this thesis. There are broadly two types of hearing loss; conductive and sensorineural. It is also possible to have a combination of these types of hearing loss. Conductive hearing loss results when sound vibration is unable to pass to the inner ear for some physical reason, often as a result of a build up of ear wax or fluid or by a damaged ear drum. In many, but by no means all, cases this may be corrected by surgery or other treatment. Sensorineural, or cochlear, hearing loss is caused by damage to the cochlear or to the auditory nerve which may have a combination of a number of contributory factors. The most common reason for sensorineural hearing loss is the ageing process, typically resulting in a loss of high frequency perception. Other contributing factors include (but are not limited to) prolonged exposure to noise (Noise Induced Hearing Loss, NIHL), disease and infections, and some medications (Ototoxicity) (Roland and Rutka, 2004). Sensorineural hearing loss accounts for the majority of hearing loss in the population (often estimated at close to 90% (Meier, 2007)). Because of its prevalence, people suffering from sensorineural hearing loss were considered the main potential beneficiaries of any solutions generated from this research.

2.2.1. Prevalence of Hearing Loss

Age Related Hearing Loss

The number of people suffering from some form of hearing loss can be difficult to assess accurately. Action on Hearing Loss (previously the RNID) estimate the number of people suffering from hearing loss in the UK to be around 10 million (Action on Hearing Loss, 2012). Davis (1989) carried out a survey across several cities in the UK and concluded that 16% of UK adults have a bilateral hearing impairment and 25% have a bilateral or unilateral impairment. Of these only 10% self reported that they had bilateral hearing difficulty in a quiet environment which indicates the difficulties of reliance on self reported statistics. The Medical Research Council's statistics show clearly the correlation between hearing loss and age.

Adults aged	with mild, moderate, severe or profound hearing loss
16 – 60	6%
61 – 80	47%
81 & over	93%

Table 1 Age distribution of self assessed hearing loss as published in International Journal of Audiology (Noble et al., 2012)

Even allowing for inaccuracies from self reporting the use of different classification systems across different countries has increased the complexities involved in understanding the prevalence of hearing loss (Roth et al., 2011). A review carried out in 2010 found that 30% of men and 20% of women in Europe were found to have a hearing loss of 30dB HL or more by age 70 years, and 55% of men and 45% of women by age 80 years (Roth et al., 2011) indicating the high prevalence of age related hearing loss (ARHL) in populations. The review also noted the difficulties in assessing prevalence of hearing loss, namely a lack of standardised method in assessing or of counting AHRL, commenting that, there were “more information gaps than information that would allow gaining a meaningful picture of prevalence of ARHL”. Nevertheless, the data available makes clear that hearing loss as a result of the ageing process is widespread. The US *National Health and Nutrition Examination Survey* (Agrawal et al., 2008) suggests that hearing loss is more prevalent in US adults than had previously been reported estimating that in 2003-4 16.1% of adults (29 million Americans) had some degree of hearing loss. The study stated that because the survey results were self reported the results probably underestimate the true scale of hearing impairment. Again it is difficult to draw comparisons with UK and European populations owing to the methods and definitions used in each study; audiograms used during Agrawal’s study in the US concentrated on speech frequencies and high frequency loss whereas UK definitions cover a wider range of frequencies.

Noise Induced Hearing Loss

In addition to ARHL, the national burden of hearing difficulties attributable to noise at work is considered to be substantial in the UK (Palmer et al., 2002) and this incorporates some demographic variation. Causes for older and for younger people in particular show considerable variance. Several sources provide indications as to the prevalence of NIHL in the UK, a Health and Safety Executive (HSE) sponsored report (Palmer et al., 2001) collates data from a number of sources as follows. In 1995 14,200 people were in receipt of benefit for industrial deafness (HSE., 1997), however this does not reflect the number of people suffering from NIHL, a Medical Research Council survey quoted by HSE (HSE.) estimates the true number to be closer to 509,000; the discrepancy being mainly because of the conditions needed to in order to claim benefit including a high degree of hearing loss (>50dB in both ears). A self-reported survey by HSE gives some credence to this with 140,000 people being estimated to have deafness or tinnitus made worse by their employment (Jones et al., 1998) and in the four year period 1991-1995 the UK Association of British Insurers handled 230,000 NIHL claims. Given that all of these surveys only included people currently in work it is likely that, once those people no longer working are taken into account, the numbers would be much higher. It could indeed be argued that the prevalence of NIHL in older people will be considerably higher than in those currently of working age owing to the lack of a stringent health and safety at work regulatory framework at the time that they were working.

Amongst young people the impact and cause of noise induced hearing loss (NIHL) may be different, one substantial component being entertainment. Studies into hearing impairments from recreational activities (Clark, 1991; Maassen et al., 2001) have found noise levels substantial enough to cause some permanent hearing damage with repeated exposure across a wide range of activities. Detailed studies into musicians (Axelsson and Lindgren, 1978; Axelsson et al., 1995), employees in the entertainment industries (SADHRA et al., 2002) and young people listening to music on headphones (Peng et al., 2007) largely indicate substantial impact of entertainment on the hearing of younger people.

2.2.2. Application of Hearing Aid Processing

Hearing aid processing design has used a number of approaches that could have application to improving television sound for hearing impaired people. Turner and Hurtig (1999) investigated using proportional frequency compression as an aid to intelligibility and found some improvements but concluded that it was less effective than high frequency amplification in most participants. In a smaller study Mazor et al (1977) found that frequency compression actually reduced intelligibility in most cases. Roch et al (2004) discuss the benefits of frequency compression for some listeners with sensorineural hearing loss and propose a pattern recognition system to compensate for the material dependent nature of this method. The research found that voices with different fundamental frequencies required different degrees of frequency compression to attain the best intelligibility improvements.

Multichannel amplitude compression solutions have been investigated and have shown superior benefits to conventional linear hearing aids in some studies (Moore, 1987; Moore et al., 1992; Laurence et al., 1983) although this is not universally accepted. Plomp (1988) argues that fast acting multichannel amplitude compression has a negative effect on speech intelligibility and the subject has been the source of considerable debate. Humes et al (1986) also compared conventional linear hearing aids with 2 channel, wide dynamic range compression (WDRC) aids and used a longer test period to allow for acclimatisation effects. This research utilised the Connected Speech Test designed by Cox et al (2001) and found benefits to both types of hearing aid but with greater improvements being shown using WDRC, particularly for lower speech levels. Moore and Glasberg (1986) compared the performance of single channel and two channel compression in hearing aids and found benefits to both but significantly better results from the two channel system in noisy situations. Barford (1978), on the other hand, found multichannel compression to have less intelligibility benefits than an optimally fitted linear hearing aid. It is important to state that the characteristics of these multichannel aids are tailored to each individual and may therefore be of limited benefit in developing any ‘hard of hearing output’ for digital television. However, Moore’s research (2003) does indicate that compression may be beneficial even when not aiming to match the characteristics of an individual’s hearing loss. There is also some debate as to whether hearing aids significantly improve understanding of speech on TV, one study found no significant benefits to intelligibility for older adults using hearing aids although significant results were obtained indicating benefits for closed captioning (Gordon-Salant and Callahan, 2009). Various factors were cited for this, one being that speech on television is simply too degraded for understanding even with hearing aids used.

Some recent assistive technology approaches for people with hearing impairments take advantage of the increasing processor power available in mobile devices and three main approaches are common: hearing enhancement, visual augmentation, and multi-modal augmentation. Apps such as BioAid and Aud-1 (Clark, 2012) bring hearing aid technology to the mobile phone and allow users to adjust parameters according their own needs. Speculation as to the impact of Google Glass for the hearing impaired community is rife; bone conduction audio could be beneficial for users suffering from conductive hearing loss and the potential for *real-life* closed captioning based on automatic speech recognition has been discussed (Flacy, 2012) although as yet little research has been done to investigate further. Other approaches take a multimodal approach with pattern recognition techniques used to identify important or useful events and generate appropriate displays (Mielke et al., 2013).

2.2.3. The Impact of Multichannel Audio

The Dolby Digital 5.1 surround sound format may in itself bring advantages for hearing impaired and other television viewers. Some research suggests that there may be some benefits for television sound by the addition of a central loudspeaker, as is used in 5.1 surround sound systems, compared to a central 'phantom' stereo image.
Often, where both 5.1 surround sound and two channel stereo broadcasts take place only one mix is carried out in 5.1 and an automated down mix used for stereo broadcast. Increasingly though, the 5.1 mix is the only available broadcasted format and downmixing occurs at the set top box in the users' home. It is suggested by Dressler (Dressler.R., 1996) that the downmix process, whereby a 5.1 surround sound audio stream is converted for 2 channel playback, may distort the mix in such a way as to reduce intelligibility by altering "the subjective balance of the mix". Holman (1991) suggested that the addition of a central loudspeaker made the material easier to understand although stated that this may not actually produce greater intelligibility. This effect, leading to an apparent difficulty in understanding, is a result of acoustical crosstalk (Holman.T., 1996) that occurs when two identical signals arrive at the ear with one slightly delayed compared to the other. This produces a comb filtering effect that cancels out some frequencies in the audio.

Additionally the comb filtering effect has been found to be detrimental to the listening experience more generally. Commenting on frequency response problems caused by signal path delays David Clark states that "Clearly the 'phantom' center is an entirely different listening experience than pure left or pure right. One might ask if stereo is deeply flawed as [a] sound reproduction technique or if interference notches should simply be ignored" (Clark, 1983 cited in Vickers, 2009a). Impacts for listeners such as these that go beyond intelligibility mean that considerable efforts have gone into attempts to remove, or reduce the impact of crosstalk. Methods have been proposed to reduce the impact of this crosstalk by Cooper and Bauck (1989) and Bauck and Cooper

(Bauck and Cooper, 1992) but these may be impractical in the context of television viewing as they utilise crosstalk cancellation techniques that rely heavily on the listener being in the ideal listening position. Clearly in a home environment this is very rarely the case. Vickers (2009a) recognises this and goes further pointing out that “when the listener is not equidistant from the speakers, the comb filter peaks and nulls will be in different frequency locations at the two ears”. The resultant comb filtered perception in any given location in the room then becomes very unpredictable and impossible to compensate with additional comb filters. There is some debate as to the specific cause of intelligibility problems resulting from crosstalk. Bucklein (1981) suggests that intelligibility difficulties may actually be made worse by peaks resulting from the crosstalk effect, rather than the troughs, as might be assumed however the underlying problem remains regardless of which effect of crosstalk is most detrimental. Other approaches to reduce crosstalk impact have been suggested; decorrelation methods (Kendall, 1995; Boueri and Kyriakakis, 2004) have been suggested so as to randomise the effects of crosstalk and so make the effects less prominent, this can be seen as a signal processing equivalent of relying on room reflections to even out responses, but others have found artefacts and distortions from these methods which, with musical content, have manifested themselves as unacceptable timbre change (Augspurger et al., 1989).

Vickers also suggests a further possibility for defeating crosstalk; by deriving a centre channel from two channel stereo content which would then be presented as a real, rather than a phantom, source. He suggests a method for accomplishing this using frequency domain upmixing (Vickers, 2009c) and provides a useful review of upmixing methods (Vickers, 2009b). Clearly this would be a useful direction if it was effective as side channels (L and R) could be reduced with reference to the new centre channel content in order to improve intelligibility for people with a hearing impairment. His research suggested that existing upmixing algorithms either provided inadequate centre channel separation or produced ‘watery sound’ or ‘musical noise’ artefacts (Vickers, 2009b) although formal subjective testing was not applied to assess this thoroughly (Vickers, 2013). These methods are specifically about spatial decomposition, rather than signal separation, a different approach which is beyond the scope of this thesis. Goodwin and Jot (2007) make reference to primary-ambient decomposition in extracting ambient information from stereo signals using principal component analysis and a variation on principal component analysis for accessible TV audio is implemented in chapter 6 of this thesis.

Much of the limited literature around the subject of broadcast audio for hearing impaired people covers signal processing methods but is speculative as regards the impact on people, particularly hearing impaired people. There is a substantial gap in the literature of robust subjective assessment of how such processes affect people with hearing impairments and this thesis aims to fill some of these gaps.

When carrying out subjective assessments for perceptual aspects such as clarity there is an issue of the degree that visual cues can influence understanding of test material, in addition to the intelligibility of the audio information. Grant at al (1998) found great variability between participants in their ability to utilise audio/visual integration to improve understanding of material but estimated potential improvements using visual content of up to 26% in some individuals. Early research by Sumby and Pollack (1954) indicates that the visual contribution to speech intelligibility increases significantly as speech to noise ratio decreases. In the VISTA project (Carmichael et al., 2003) a high degree of ‘speech reading’ was recognised as being attempted by older participants in attempting to understand an avatar with a synthetic voice, this was partially unsuccessful owing to lip sync problems although this in itself indicates a degree of reliance on visual cues for older users. Other research (Beerends and De Caluwe, 1999) shows biasing in assessments of AV media quality from both audio and visual interactions. The research indicates that, in their study, quality of visual presentation had more impact on assessments of audio quality than quality of audio presentation had on assessments of visual quality. In each case significant influence was demonstrated. For the audio researcher this is potentially problematic and care must be taken to ensure that video quality is consistent throughout AV media presentation. Audio quality assessments should therefore ideally be carried out in *audio only* conditions. For some tests however, for example where audio quality may not be the only descriptor under scrutiny, audio-visual presentation will be necessary and any test procedures incorporating visual material must also be carefully designed to eliminate any bias resulting from visual cues in the media.

2.3. Audio for Digital TV Broadcast

As well as these acoustic and psychoacoustic factors a further consideration is the implementation of international standards in delivering audio for television. This section gives an overview of systems currently in place in order to present a clearer understanding of the potential for broadcast standards to provide a workable solution. The fast paced rate of change of the broadcast television landscape means that standards and formats have undergone some changes during the research period this thesis covers and this section also serves to illustrate the continuing relevance of the research to the current standardisation situation.

2.3.1. Television Audio Systems and Formats

International television standards are covered by a range of standards bodies however this thesis focuses on standards applying to the UK. Implementation of any recommendations and guidelines for 5.1 audio produced as part of the research carried out is equally applicable to other digital television standards although the detail of the implementation may differ slightly depending on the specific metadata and audio channel implementation mandated by a given standards body.

Until the introduction of digital TV broadcast all audio for TV in the UK was either mono or NICAM stereo (ETSI, 1997). Since digital switchover in the UK between 2008 and 2012 an increasing number of options have become available for broadcasters. One development has been the spread of high definition (HD) TV services, many of which have been accompanied by 5.1 surround sound. It is this introduction of 5.1 audio that this thesis takes as its starting point in order to improve TV sound for hearing impaired people. 5.1 surround sound is therefore the focus of this review of TV standards.

At the time the research documented here commenced the most popular and widely accepted standard for broadcasting audio with digital HD television in the UK was AC-3 (Adaptive Transform Coder 3). This standard, developed by Dolby Laboratories, is mandatory on DVD and HD-DVD, and optional in Blu-Ray disks. AC-3 is also utilised in digital TV broadcast in many countries including UK satellite broadcasts from British Sky Broadcasting (BSkyB) and UK cable TV from VirginMedia. During the course of this research broadcasters in the UK commenced broadcast of terrestrial HD content which utilises MPEG I Layer II, MPEG I Layer III (ISO/IEC, 1993) and the AAC and HE-AAC codecs initially developed by Coding Technologies and specified in ISO/IEC 14496-3 (ISO/IEC, 2006) (The Digital TV Group, 2012). The AC-3 codec is optional for UK terrestrial HD broadcast but was considered likely to be adopted for terrestrial broadcast because of its widespread use in satellite and cable broadcast in the UK (de Pomerai, 2009). AC-3 / E-AC-3 is specified as mandatory by the ATSC (Advanced Television Systems Committee) A/52B standard (ATSC, 2005) and is included as an equivalent second audio coding option within DVB (Digital Video Broadcasting Project) standards. The DVB specification is published by the European Telecommunications Standards Institute (ETSI) as TR 101 154 (ETSI, 2000).

An interesting feature of Dolby Digital audio is the way in which it deals with relative levels of speech in programme material. In addition to the audio material itself the programme stream contains metadata, or data about the audio data, and contains information used by the decoder in order to effectively decode the transmitted material. As briefly mentioned earlier the metadata contained in the AC-3 audio stream contains a value for *dialogue normalisation* or *dialnorm* which gives a value for the average level of speech in transmitted material. According to Dolby Labs Guide to Metadata (Dolby Labs, 2003), "*The consumer's Dolby Digital decoder reproduces the program audio according to the metadata parameters set by the program creator, and according to settings for speaker configuration, bass management, and dynamic range that are chosen by the consumer to match his specific home theater equipment and environmental conditions*." "*This control, however, requires the producer to set the metadata parameters correctly, since they affect important aspects of the audio—and* *can seriously compromise the final product if set improperly*." One study carried out by Dolby (Riedmiller et al., 2003) reveals that only 1 out of the 13 digital services surveyed in one area of the US had set the dialog normalisation value correctly and, as a result, the audio level for these services varied by as much as 16dB, much higher than the 7.8dB "comfort zone" found by Dolby in listening tests (Riedmiller, 2005). In the case of the single example where the dialnorm value was correctly set the appropriate dialnorm value happened to be the factory set default. Therefore in the study carried out *no TV broadcaster was utilising and setting this value in an appropriate way.* This misunderstanding of the importance and use of broadcast metadata has serious implications for the implementation of any metadata controlled processing at the set top box and therefore for the perceived clarity of speech and intelligibility within programmes, especially, but not exclusively, for people with some hearing loss.

The Dolby Labs decoder specification for the AC-3 audio stream utilises the dialnorm value in order to apply what it calls Dynamic Range Control (DRC) to the programme audio. DRC is usually utilised where audio monitoring is via inferior reproduction equipment which may not manage high dynamic range content appropriately, and also to enable what is sometimes referred to as 'midnight mode'. Midnight mode allows extreme volume levels of sound effects in the sound track to be reduced so as not to cause disturbance to neighbours for late night viewing without affecting the level of the dialogue of the programme material. DRC functions by compressing the audio relative to the level of the dialogue in the audio stream such that levels below the dialogue level have some gain applied, levels above the dialogue level are attenuated and a null gain area is retained for dialogue content which remains unchanged. The level at which this null area resides is set by the dialnorm value. It is clear that any method for improving speech clarity that relies on metadata being set appropriately in the broadcast and production chain, such as utilising DRC compression settings, may fail unless broadcasters take adequate care in setting appropriate metadata values.

As already stated enhancements to the AC-3 standard are contained in the more recent Dolby Digital Plus standard (DD+ or E-AC-3 (Enhanced AC-3)) (Fielder et al., 2004).

These enhancements, although including the potential for independent control of channel levels based on metadata, still rely in equal measure on the validity of the metadata generated by producers and broadcasters.

Where broadcast is in 5.1 surround sound STBs commonly downmix the multichannel audio in order to enable reproduction on two channel stereo and mono reproduction systems rather than broadcast a separate stereo mix of the programme. There are two downmix methods available on Dolby compatible STBs as follows.
Lt/Rt (left total, right total) stereo downmixing capability is found in every Dolby compliant STB and DVD player and is the default stereo downmix for Dolby compliant equipment. By implication this means it is the default implementation for the large majority of surround audio reproduction equipment currently available. The Lt/Rt mix contains elements of all 5 full range channels; centre channel is sent to both left and right at -3dB, surround channels are combined into a single surround channel which is added to left and centre out of phase and to right and centre in phase. LoRo stereo (left only right only) is normally generated by attenuating and then adding each surround channel into its respective front channel, Ls to L, Rs to R. Centre channel is added to left and right at -3dB as shown in figure 2 although the degree of attenuation can be altered in metadata at the encoding stage.

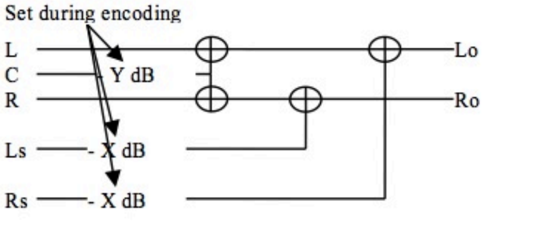
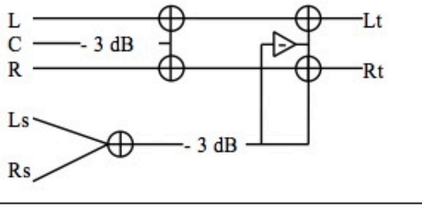
Tests	Details
Stereo Lo/Ro (down-mixed)	
Stereo Lt/Rt (down-mixed)	

Figure 2 Derivation of Lt/Rt and LoRo stereo downmixes from 5.1 multichannel audio material.

2.3.2. Future Developments

Current use of 5.1 surround sound in broadcast is likely to be predominant for some time to come as broadcasters and broadcast manufacturers have invested substantially in infrastructure developed around these formats. Test broadcast in Japan utilizing a 22.2 audio system suggest that an increased number of channels may be one route forward as high bandwidth connectivity becomes more widespread. Any such system would certainly be accompanied by similar metadata that could be utilized in the same way as any accessible audio system developed for 5.1. Another possibility is that object based audio may replace channel based audio as a broadcast audio standard. Object based audio treats individual sound sources as discrete objects with coordinate locations, regardless of reproduction system, and developments from companies like DTS, Fraunhofer, Dolby and others suggest that this may become more mainstream although not in the short term. This is covered in more depth in chapter 7 of this thesis.

2.4. *Psychoacoustic Test Methodology*

In order to design appropriate test methods to assess the viability and effectiveness of processes and conditions for TV audio it is firstly important to be clear on definitions of what is being assessed. At this stage a distinction is drawn between intelligibility of speech and clarity of speech. Within this thesis the term ‘clarity’ is defined as *perceived* clarity, i.e. a measure of how clear the speech appears to be to a listener. Intelligibility will be used to refer to a measure of how well speech can be understood and assessed either by correct identification of words or by comprehension of the meaning of phrases or sentences. Although there are clear similarities between these terms and many instances in literature where each are used interchangeably there are some examples where processed speech can appear clearer and yet be no easier to understand (Carmichael, 2004) so the distinction is an important one. The relationship between the two factors would be expected to be close in most cases however using the descriptor ‘speech clarity’ could be expected to be more influenced by other factors such as the more ambiguous ‘quality’ than the more objective, score based measure of ‘intelligibility’. Similarly, the effect of each factor on user experience more generally could be expected to be similar however there are circumstances where this may not be the case. A completely isolated speech channel would undoubtedly produce better intelligibility ratings but the complete absence of music and sound effects important to scene comprehension may produce poorer ratings for enjoyment for some AV media with some participants.

In designing subjective assessments for audio conditions it is critical to understand the nature of the data or information required that will allow the most useful analysis. For this thesis the use of quantitative methods combined, where appropriate, with objective measurement of conditions has been used. This approach has however been informed by informal semi-structured interviews with participants in the research. The outcomes of the interviews have not been used to derive clear research conclusions but instead have been used to gain some insight into why particular results may have been obtained and to inform the development of test methods used during the research.

Zielinski's review of biases in audio listening quality tests identifies much of the potential for gaining meaningless or misleading data from listening tests (Zielinski et al., 2008) and all of this is relevant in the design of tests during the research presented here. The biases identified include recency effects, listener expectations and preferences, stimulus related biases such as uneven frequency of presentation, scale and range related biases and biases resulting from the appearance of the experimental interface.

The nature of research involving hearing impaired participants mitigates against the adoption of standard test methodologies used for audio assessment and also creates substantial challenges for the researcher. When discussing listening tests for loudspeaker assessment Toole (1985) identified a number of what he called 'nuisance variables' that could cause large variability in subjective assessments, these were split into those associated with the listening environment, those related to the listeners themselves and those related to experimental procedure or test design and are presented here in order to assess the ‘nuisance variables’ that can and can not be excluded from the research documented in this thesis.

Toole’s ‘Nuisance Variables’

Listening environment factors:

- *Listening room*
- *Loudspeaker position*
- *Relative loudness (of compared sounds)*
- *Absolute loudness (of all sounds)*
- *Program material*
- *Electronic imperfections*
- *Stereo (peculiar technical problems)*

Listener factors:

- *Knowledge of the products*
- *Familiarity with the programme*
- *Familiarity with the room*
- *Familiarity with the task*
- *Judgement ability or aptitude*
- *Hearing ability (physical impairment)*
- *Relevant accumulated experience*
- *Listener interaction and group pressure*
- *Stereo (conflicts between spatial and sound quality aspects of reproduction)*

Experimental Procedure:

- *Identification of perceptual dimensions*
- *Scaling of the perceptual dimensions*
- *Anchoring or normalisation of individual scales*
- *Effects of context and contract*
- *Effects of sequence and memory*
- *Experimenter bias*

(Toole, 1985)

Clearly there are considerable challenges for the researcher in assessing audio reproduction conditions using subjective test methodologies. Listening tests documented in international standards have the advantage that where the potential for biases exist, they are well understood and can therefore be mitigated against to a large degree. For subjective assessment of audio systems with hearing impaired participants there is considerably more potential for unpredicted biases to appear and great care must be taken in developing test methodologies for this group. Perhaps unsurprisingly most research into improving TV sound for hearing impaired people has focused largely on signal processing methods and carefully controlled subjective assessments with hearing impaired participants have been rare. This thesis aims to fill some of these gaps and develop robust test methods for assessments of potential answers to the problems that hearing impaired people experience in viewing TV in their homes.

3. The Clean Audio Project Phase 1: Non-Speech Channel Attenuation

This chapter is based on research largely funded by the ITC and then by Ofcom. The research investigated the problem of TV audio for hearing impaired viewers and potential solutions that could improve their experience of TV audio. Within the chapter the use of production guidelines is discussed, Dolby Digital metadata is discussed as a means of conveying information to the receiver, the effect of attenuating non-speech channels in a surround sound system is evaluated and a review of test procedures is carried out in the light of the results obtained in order to identify biases and inform future test design.

3.1. *Introduction*

Although many complex solutions have potential to improve TV sound for hearing impaired people, the potential for metadata to enable independent control of channel levels makes the need for an investigation into the impacts of reducing non-speech channels on both clarity and perceived sound quality clear. The fact that most hearing impaired viewers watch television on a shared television set makes it critical to understand the effect of this processing on listening pleasure, or perceived sound quality for both hearing impaired and non-hearing impaired people.

Some of this research has been published in the Technology and Disability journal article, *The ITC Clean Audio Project* (Shirley and Kendrick, 2006) on which this chapter is based.

3.2. *Aims of Clean Audio Phase 1*

The aims of Clean Audio phase 1 as agreed with the research funders (Independent Television Commission) were as follows:

- To assess the effect of attenuating left and right channels in a 5.1 surround sound system for hearing impaired viewers.
- To assess any benefits of 5.1 surround sound compared to downmixed stereo.
- To assess the effect of this remix for non-hearing impaired viewers.

- To produce recommendations for hard of hearing viewers as to how they may improve their viewing experience.
- To produce guidelines for broadcasters.

3.3. Methodology

The test methodology chosen was a two way forced choice comparison and participants assessed audio-visual media, rather than audio only. This choice of test method introduces a number of potential issues for the research however, on balance, other possible methods were rejected. A summary of these choices and rationale is as follows.

3.3.1.Audio-Visual Stimuli

In most assessments of audio quality and in standard test methodologies audio is assessed in isolation and for good reason. As has already been discussed in section 2.2.3 visual material can have a significant effect on the ratings of audio quality. It was also considered likely that some degree of lip reading may also have an impact. However in these tests a further descriptor of the media was being assessed beyond audio quality. It was considered important that the tests also assessed hearing impaired and non-hearing impaired participants' *enjoyment* of the media overall as this would be a critical factor in viewers' acceptance of any recommendations developed form the research. It was also considered critical that participant's experience of the audio-visual media in the tests mirrored as closely as possible their experience of TV in the home. For these reasons test presentation was of AV media.

3.3.2.A/B Comparative Tests

An option considered during test design was to use a variation on MUSHRA (ITU, 2003) test methods. The MUSHRA test design typically presents the participant with a selection of media clips simultaneously and the participant is permitted to switch between conditions at will. Each condition is rated using a descriptor scale. This type of presentation that allows the participant to take control of the order of playback of the conditions is useful for assessments of intermediate audio codec quality in that it reduces the impact of the *recency effect* noted by Zielinski et al (2008). A disadvantage of this method becomes apparent when presenting material that varies substantially over time. For the clips used speech was not constant throughout the duration of the clip.

Inevitably the AV material utilised in the tests had sections where no person was talking and some sections with clearer speech than others. It is likely that a participant could inadvertently switch between sections such that one condition had substantially more clear speech than other conditions and so produce an unknown variable in to the test procedure. Utilising an AB comparison test was considered to allow more control over unwanted variables because the impact of recency effect could be factored out by randomisation of presentation order and by maximising section equivalence across the section duration. Avoiding more than two stimuli would also assist in reducing the centering effect discussed in Zielinski et al's review of listening test biases. The centering effect refers to a tendency for subjective assessment scores from multiple stimuli to vary such that the mean of the scores of all stimuli tends to the mean of the score data. The effect has been shown to be reduced by not using multiple stimuli .

3.3.3.Forced Choice Comparison

A variation of the CMOS and CCR scaled paired comparison methods (ITU, 1996) was used for the tests. The tests used a forced choice comparison in order to present a rating of 'how much better' rather than a continuous scale between stimuli A preference and stimuli B preference (as would be the case for CMOS testing) so that the centre point, where the stimuli were the same, was not visible. Additionally previous scores were presented on separate sheets and were not visible while subsequent judgements were being made by participants thus removing any visual cues that may encourage centering. This was considered likely to further reduce the centering effect.

3.3.4.Other Test Design Factors

The challenges for the researcher carrying out subjective testing involving hearing impaired participants become clear when considering Toole's extensive list presented earlier and although many of the 'nuisance factors' were able to be excluded from tests in this research, some could not, and for this reason care needed to be taken to ensure validity of test results.

The subjective assessments presented in this thesis were carried out in a listening room conforming to ITU-R BS1116 therefore factors stemming from the listening room acoustics and loudspeaker positions were of limited concern. Great care was taken in ensuring that relative reproduction levels between conditions tested were consistent, however because of the nature of the people participating in tests, absolute loudness was unavoidably variable from participant to participant. Preferred loudness levels between participants varied by as much as 21dB; those with severe hearing loss required loudness levels that would be uncomfortable, if not painful, for participants with mild or no hearing impairment. For this reason it was necessary for participants to conduct assessments individually and no attempt was made to carry out group subjective assessment. This mitigated against the problem of listener interaction and group pressure although added considerably to the time taken to run listening tests. Considerable care and time was invested in identifying appropriate equivalent programme material for tests - an extensive review of test methods has been carried out to attempt identification of biases by studying test outcomes with regards to analysis of programme material. In order to avoid sequencing and memory biases, these were then presented in a pseudo-random manner ensuring that clips and processes were both presented A-B an equal number of times as B-A for each subject. Also every condition was tested an equal number of times with each media clip. Participants were appropriately briefed by use of a standard script in order to avoid inadvertent experimenter bias and were trained in the task with test examples prior to commencing assessments. Perceptual dimensions, and the interface and scales on which these were graded, were carefully chosen to make clear exactly what was being assessed at any given time and to ensure the tests were as simple as possible to understand from the participants' perspective. There were however factors identified by Toole that could not be removed from the assessments. Participants did not have 'normal hearing' and there was no consistency of hearing impairment between participants. Many of the participants were older people and did not feel comfortable working directly with a computer interface so programme selection and condition switching was carried out by the researcher under instruction from participants who marked results on paper instead of putting the subject in full control of playback and choice of condition. Despite this, wherever possible, experimental procedures were automated using control software in order to avoid experimenter error. Also, although an anchor or reference item was inappropriate for most of the conditions under assessment[6], presentation was designed as far as possible to avoid the centering effects discussed by (Zielinski et al., 2008). It was clearly understood that the outcomes from tests would be relative measures of perceptual scales between conditions rather than absolute measures of any given factor under test and no attempts were made to attempt absolute measures using the data. Where possible existing and proven test methods were adapted however none of the existing test standards were considered applicable for adoption in its entirety. For example the need for hearing impaired participants meant that there were no 'expert listeners' as required by ITU-R BS.1116 (ITU, 1997). The ITU standard states that, "It is important that data from listening tests assessing small impairments in audio systems should come exclusively from participants who have expertise in detecting these small impairment", and that, "An insufficiently expert subject cannot contribute good data". This would effectively exclude all hearing impaired participants who were, after all, the focus of the research therefore great care had to be taken to control all other potential effects that may influence experimental data. Standard and uniform reproduction levels specified in listening test standards are also inappropriate for this research because of variation in the hearing acuity amongst participants. The ITU-R BS.1116 standard is aimed at detecting 'small impairments' in audio systems and much of the research documented here is about comparing features of reproduced sound for which was usually no 'reference' that would be expected to be graded better than other processes. This also excluded ITU-R BS.1534 (MUSHRA)(ITU, 2003) testing which requires an unprocessed reference condition where other conditions are graded with reference to this unprocessed audio. In the case of improving audio for hearing impaired people the reference unprocessed condition is quite likely to be graded lower than processed conditions thus negating its benefits as an anchor.

[6] Typically an anchor reference of known quality - either good or bad - will be used in detecting impairments in audio systems in order to stabilise ratings scales and also to identify 'outliers' considered insufficiently capable of identifying differences between conditions. For the research presented in this thesis the aim is to improve perceptual factors from the reference position and no reasonable justification could be constructed for any given reference for a group of test participants with differing hearing abilities and impairments.

There is a further issue to be addressed regarding the listening conditions for hearing impaired people. Participants stated whether they watched TV with or without their hearing aids in place and answers were mixed. For example some preferred not to use their hearing aids at all at home, others used hearing aids so that the TV did not have to be too loud for other members of the family. In order to assess the impact of varying reproduction conditions and processes on home viewing of material the decision was made that participants in the research would experience the audio in the way that they would normally experience television audio. If they normally wore a hearing aid to view TV, they could wear their hearing aid for the tests. If they would normally remove the hearing aid to view TV they were asked to do the same prior to tests commencing. The decision to allow hearing impaired participants to wear hearing aids for the tests inevitable introduces some additional 'nuisance variables' in itself. Each participant's hearing aid would be calibrated differently each participant will then be hearing something different from the others. The impact of these nuisance variables was considered closely however other factors influenced the decision. The fact that each participant had a different hearing impairment meant that each was already hearing a different cue for the same condition, the impact of wearing an aid was considered to not make this significantly worse. Some participants' hearing was so impaired that without an aid they were unable to gain any meaning from the speech in any condition. Also, the aim of this research is to make a difference to peoples' experience of TV audio in their homes. Because substantial numbers of people wear aids to watch TV any solution developed by the research had to work for these viewers, for these reasons it was decided that the benefits of allowing hearing aid use outweighed the disadvantages.
Listening tests took place in a listening room that conformed to ITU-R BS.1116-1 *Methods for the subjective assessment of small impairments in audio systems including multichannel sound systems* (ITU, 1994a) and ITU-R BS.775-1 *Multichannel stereophonic sound system with and without accompanying picture*. (ITU, 2006)
Equipment used:

- DVD Player
- Digital Audio Mixing Console
- Dolby DP562 Reference Decoder

- 3 x Genelec 1029 Active Monitors
- Kamplex BA25 Audiometer

Tests were carried out with each subject utilising the same test material on the same day; these took the form of forced choice paired comparison assessments for each condition. An audiogram was carried for each subject in order to assess the hearing acuity of the test panel and to gauge the degree and nature of each subject's hearing impairment. Assessments carried out in this phase of the research were aimed at assessing dialogue clarity, overall sound quality and enjoyment of clips of AV media across a range of material. It was considered important that clips used for each condition should be equivalent to those used for each other process in order to avoid choices being based on the clip, rather than the process under assessment. Playback of the same clip for each condition was considered however this was thought likely to be detrimental to the tests for the following reasons:

- Assessment of the clarity of speech within a section would be biased if the section had already been viewed and the speech content was therefore known to the subject.
- Subject fatigue would be increased by repetitious viewing of the same clip.

Pilot experiments using identical clips to identify appropriate volume levels showed the former concern to be well based. When presented with the same clip for each process participants consistently set the volume level highest for the first condition played and lowest for the last regardless of the order of the conditions. In order to avoid this impacting on and biasing results consecutive clip sections were utilised for all tests.

As far as possible the differences between the sections were minimised by using two sections within the same clip and utilising clips where the two sections are very similar in terms of scene, type and amount of background noise and in the identity of the person speaking.

The clips were played from DVD and visual cues were used to indicate the start of a new clip and which section was being played. The visual cue consisted of a black screen with a title giving the clip or section number. The use of an audio cue was tested (a short steady tone) in addition to the visual cue to separate the two clip sections. Six pilot tests were carried out in order to identify any potential biases and also to identify any potential technical problems with the method and apparatus used for the tests. After some experimentation it was noted that the addition of the audio cue was distracting and hindered the participants' auditory memory of the previous section. Results were inconsistent and participants stated that they found the tone to be a distraction and an annoyance which they thought was preventing them from remembering the previous clip's audio qualities. Instead of using an audio cue, the duration of the visual cues (black screen with white text label) was increased to ensure that the change in sections remained noticeable but that the distracting tone could be removed.

An analysis of the test material was carried out subjectively with respect to the loudness of background sound and music compared with the dialogue in order to choose clips that were broadly equivalent. The amount of dialogue spoken off-camera and facing away from camera was also measured by counting words spoken when the speaker's mouth was visible compared to when it was not although it was not possible to identify clip pairs with an identical, or close to identical, proportion of face-to-camera dialogue.

Participants were asked to watch excerpts of video material with a Dolby Digital 5.1 encoded soundtrack. Each excerpt was split into two similar sections with a different condition being applied to each section. The subject was then asked to assess which of the two sections was preferred based on three criteria:

- Overall sound quality.
- Their enjoyment of the section.
- The clarity of the dialog.

Participants were also asked to show how much better their preferred section was for each of these criteria. There was no option for the two sections to be assessed as being the same. All AB and BA comparisons were assessed by each subject, with the order of the conditions changed for each subject so ensuring that every process was carried out on every video clip. An example answer sheet can be seen in Appendix D

3.3.5. Interviews with Participants

During the process of carrying out test procedures, and in order to gain maximum benefit from access to substantial numbers of hearing impaired people, further work was carried out to try and gain some insight into the experience of hearing impaired people as TV viewers. In addition to measured and controlled tests, informal interviews with participants were carried out between tests. These interviews were useful in order to give some insight into the issues considered most important by hearing impaired people and in order to identify any potential biases in test results.

3.4. Test Material

The test material consists of a series of 20 video clips with a Dolby Digital 5.1 soundtrack. Each clip was split into 2 sections and each section treated with a different process on playback. To this end it was considered important that the amount and type of side channel audio was consistent throughout the clip so that like was compared to like.

Clips were introduced by a title reading "Clip x" (where x was the number of the clip), each section of the clip was introduced with a 3 second title reading "Section A" or "Section B".

Clips were chosen according to the following criteria:

Criteria	Details
Length	Between 1 min and 1 min 30 seconds
Amount of side channel noise	Clips chosen have moderate side channel audio that could possibly mask sounds in the dialogue channel.
Type of side channel 'noise'	A variety of types of side channel audio including background speech, music and sound effects.
Mix of off-camera and on-camera speech	Consistent amount of *face to camera* dialogue between clip sections in order to avoid results being biased by lip reading

Table 2 Criteria used to choose appropriate video clips for subjective tests

It was thought important in choosing the clips that each clip should appear to be complete in itself, i.e. at the end of a clip the subject matter is brought to some sort of conclusion. This was seen as important in order to give some meaning to the 'enjoyment' factor for each section/process combination and in order to reduce potentially irritating breaks in the video sequences. Within this limitation, the length of each clip was standardised as far as possible.

Clips were chosen with a moderate amount of different types of background, side channel audio. The background audio included speech, music and sound effects. A complete listing of the clips used is included in Appendix B.

All of the paired forced choice comparison tests were carried out with levels calibrated in order that the overall A-weighted sound pressure level was identical.

3.5. Analysis of Participants

The group was composed of 41 participants with a range of ages and hearing impairments ranging from profoundly deaf (with cochlear implants) to non-hearing impaired. The profile of participants was as follows.

3.5.1. Profile of Hearing Impairments

The level of hearing impairments as measured by pure tone audiogram can be seen in figure 3.

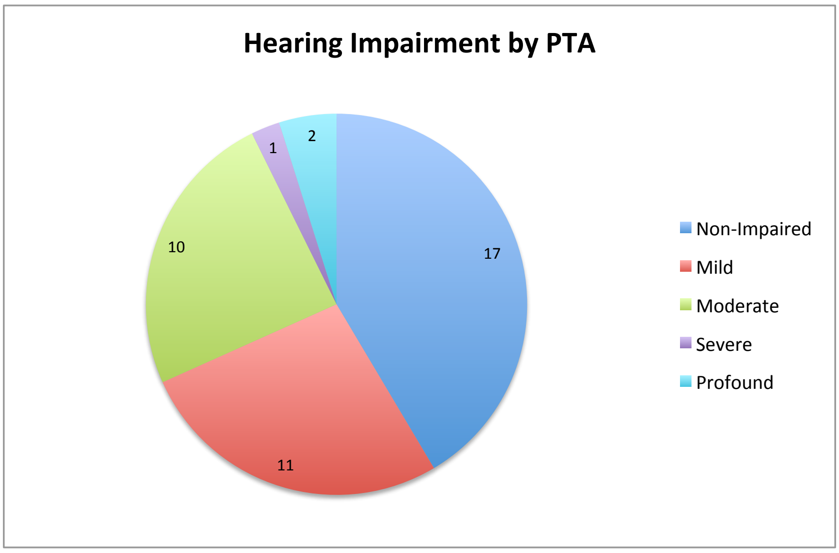


Figure 3. Ratings of participants' hearing impairment by pure tone audiogram according to categories

defined in (British Society of Audiology, 1988)

The accepted definitions of degrees of hearing loss in the UK are categorised by the hearing acuity of a persons best ear (British Society of Audiology, 1988) however within the subject group for this listening test there were 5 participants with asymmetric hearing loss who, although classified as 'non-impaired hearing' by this definition, had substantial hearing loss in one ear. As these participants had self-reported that they had hearing loss and also that they had difficulty understanding speech on TV, and as their asymmetric loss was substantial, these participants were considered as 'hearing impaired' for the purpose of data analysis.

3.5.2. Setup and Calibration

Equipment setup was as follows:
The subject's chair, the reference monitors and the television were placed in the ideal listening positions according to the *ITU-R BS.775-1 multi-channel stereophonic sound with and without accompanying picture (ITU, 1994)* recommendations. Control equipment was situated at the back of the listening room with the operator out of sight of the viewer. The digital S/PDIF output from the DVD player was converted to AES/EBU standard using a converter and connected to the AES/EBU input on the Dolby Reference Decoder. Three outputs from the decoder are used: left, centre and right. Left and right channels are fed into 3 separate channels on the mixer, centre channel to one other. One channel was used for each required output level enabling the levels to be changed by muting and un-muting channels. Each channel was then routed, via the internal busses of the mixer, to one of three bus outputs, each of which was connected to the corresponding monitor. All of these faders could be grouped together so that the initial reference volume level for each subject could be adjusted using a single fader.

In order to minimise the possibility for human error during the testing procedure the test was designed to be as automated as possible. Level setting for each subject was accomplished by developing some simple control software capable of sending MIDI controller information in response to experimenter input. The software was used to accept the input from a mouse wheel and used to produce MIDI messages that could be used to alter the output level of the group busses and therefore the input to the monitoring equipment. In this way each subject was able to set their own preferred listening level at the start of the tests without any possible bias from the test administrator and without any clear maximum or minimum levels.

During preliminary testing with 3 participants it became apparent that the overall sound pressure level (SPL) was being altered considerably by each process and with it, the subject's perception of what was the 'best' section. Many repeatable listening tests have shown that there is a significant bias toward audio with a higher sound pressure level (Nousaine, 1991). For this reason it was important that the overall sound pressure level remains constant for all of the AB comparison tests.

In order to ensure that this was the case, a test DVD was burned which consisted of pink noise on the three front channels. The A-weighted sound level was measured for the lowest level process and the overall bus output levels from the mixing console altered by the software to ensure that each of the other processes was heard at the same sound level for every test.

A challenge presented in this listening test design was that, unlike with expert listening panels called for in listening test standards, participants had varying degrees of hearing loss and no single listening level was likely to be appropriate for all participants. In order to mitigate this issue participants were asked to set a comfortable listening level for the unaltered reference LCR condition at the start of the tests and were not permitted to alter this level for the duration of the test. All conditions for that subject were then played back at this reference level.

3.6. Experimental Results

Normally where anchor points are not used it is recommended to that data should be normalised (ITU, 1998) and as previously discussed no reference anchors were appropriate for these tests. Participants' data was therefore normalised according to the procedure documented in Appendix H. The impact of this normalisation on results was an improved level of significance when compared to a simple score of how many times a condition was preferred. The research was intended to assess changes in the perception of programme material for three factors: speech clarity, overall sound quality and enjoyment. The same participants experienced all conditions that were assessed making a series of one way repeated measures ANOVAs appropriate in order to identify whether there were any significant differences between the mean ratings for each criteria. Therefore a series of one way repeated measure ANOVAs were carried out for clarity, overall sound quality and enjoyment ratings. The key to processes tested is as follows:

LCR Centre channel, plus left and right channels at standard relative levels set using reference tones.
LCR1 Centre channel, plus left and right channel at -3dB.
LCR2 Centre channel, plus left and right channel at -6dB.
C Centre channel only.
LR Lt/Rt Stereo downmix.

A marked statistical significance at $p<0.05$ was found for most of the combinations tested with the hearing impaired subject group. The non-hearing impaired results showed less significance, probably as a result of the lower number of participants; only speech clarity results were statistically significant for this group based on a repeated measure ANOVA.

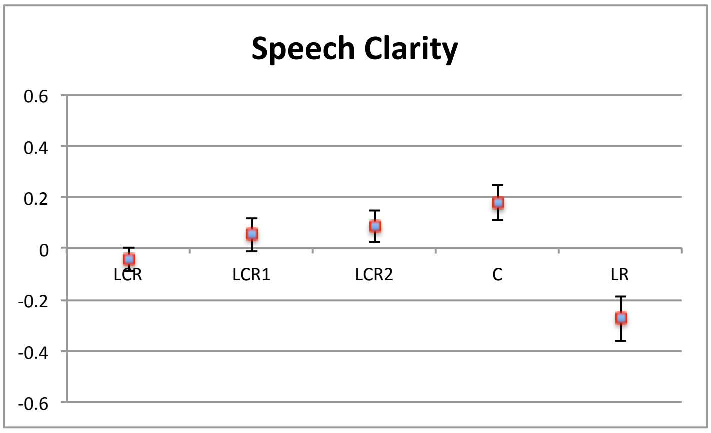
Analysis of speech clarity ratings indicate a trend for both groups that reducing non-speech channels improved the perceived clarity of speech within clips. Statistically significant findings show that for both hearing impaired and non-hearing impaired groups centre channel only (*C*) and *LCR2* were both judged as having clearer speech than the *LCR* reference condition. All conditions were considered to have clearer speech than *LR* (Lt/Rt stereo downmix) across both groups. However no statistical significance was shown in comparisons between *LCR* and *LCR1* or between *LCR1* and *LCR2*.

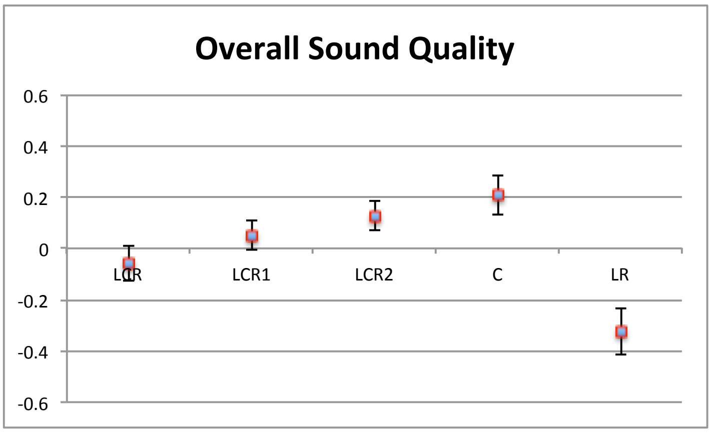
Additionally there were no statistically significant results in comparisons between *LCR1* and *C* or between *LCR2* and *C*.

For the hearing impaired group, when considering the *LCR, LCR1, LCR2* and *C* conditions as following a trend of increased non-speech channel reduction, no significance was shown for adjacent processes (ie between *LCR* and *LCR1*, *LCR1* and *LCR2*, *LCR2* and *C*) although for all non-adjacent conditions, where there was a greater change in speech to non-speech ratio, significance was unambiguous. In each case the process with more speech and less competing sounds in side channels was preferred as having higher overall sound quality. So *LCR2* and *C* were rated more highly than *LCR* and *C* was rated more highly than *LCR1*.

For ratings of enjoyment of clips under each reproduction condition there were no significant results for the non-hearing impaired group and no conclusions could be drawn for this group.
For the hearing impaired group results followed a similar trend to that of overall sound quality results: *C* was found to be more enjoyable than *LCR* and once again every other process was rated more highly than *LR.*

3.6.1. Hearing Impaired Results





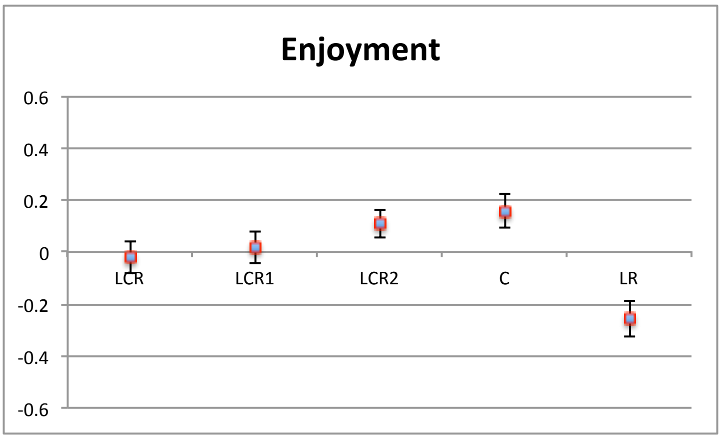


Figure 4 A, B & C Rating of speech clarity, overall sound quality and enjoyment for each condition by hearing impaired participants with error bars indicating 95% confidence level

3.6.2. Non-Hearing Impaired Results

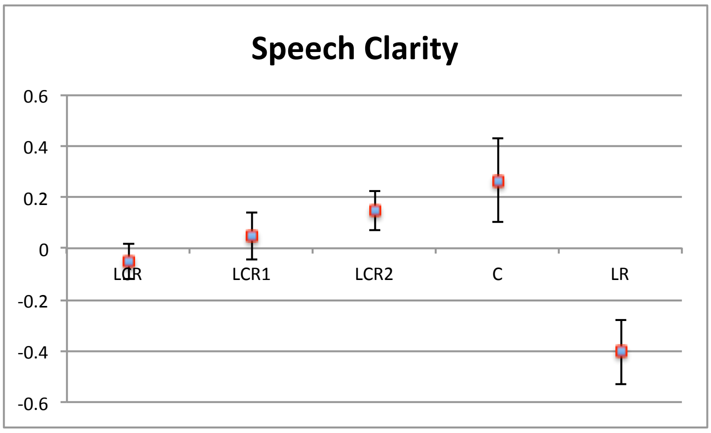
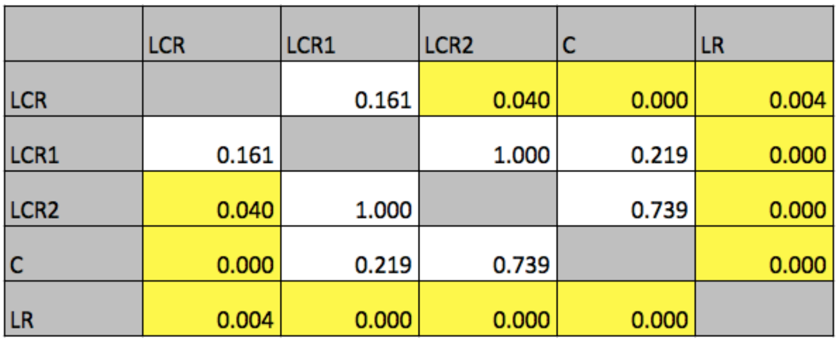


Figure 5 Rating of speech clarity for each condition by non-hearing impaired participants with error bars indicating 95% confidence level

3.7. *Significance of Results*

3.7.1. Hearing Impaired Participants

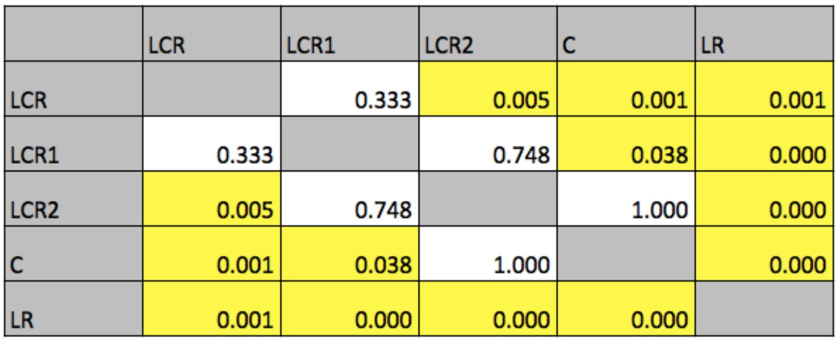
3.7.1.1. Significance: Speech Clarity



	LCR	LCR1	LCR2	C	LR
LCR		0.161	0.040	0.000	0.004
LCR1	0.161		1.000	0.219	0.000
LCR2	0.040	1.000		0.739	0.000
C	0.000	0.219	0.739		0.000
LR	0.004	0.000	0.000	0.000	

Table 3. P-values for each pairwise comparison for speech clarity ratings, highlighted values indicate statistical significance at p<0.05

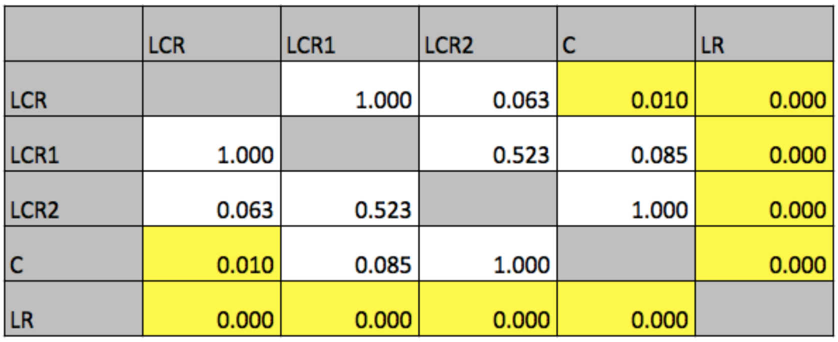
3.7.1.2. Significance: Overall Sound Quality



	LCR	LCR1	LCR2	C	LR
LCR		0.333	0.005	0.001	0.001
LCR1	0.333		0.748	0.038	0.000
LCR2	0.005	0.748		1.000	0.000
C	0.001	0.038	1.000		0.000
LR	0.001	0.000	0.000	0.000	

Table 4. P-values for each pairwise comparison for overall sound quality ratings, highlighted values indicate statistical significance at p<0.05

3.7.1.3. Significance: Enjoyment

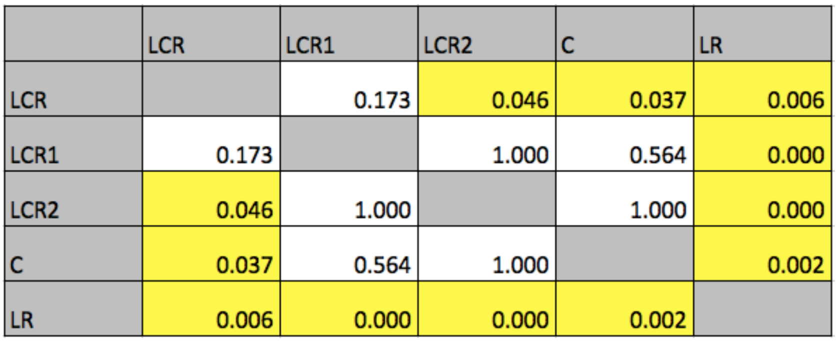


	LCR	LCR1	LCR2	C	LR
LCR		1.000	0.063	0.010	0.000
LCR1	1.000		0.523	0.085	0.000
LCR2	0.063	0.523		1.000	0.000
C	0.010	0.085	1.000		0.000
LR	0.000	0.000	0.000	0.000	

Table 5. P-values for each pairwise comparison for enjoyment ratings, highlighted values indicate statistical significance at p<0.05

3.7.2. Non-Hearing Impaired Participants

3.7.2.1. Significance: Speech Clarity

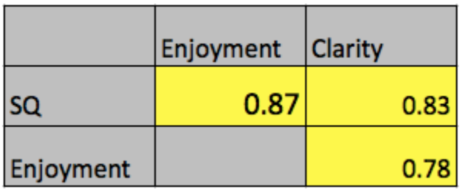


	LCR	LCR1	LCR2	C	LR
LCR		0.173	0.046	0.037	0.006
LCR1	0.173		1.000	0.564	0.000
LCR2	0.046	1.000		1.000	0.000
C	0.037	0.564	1.000		0.002
LR	0.006	0.000	0.000	0.002	

Table 6. P-values for each pairwise comparison for speech clarity ratings, highlighted values indicate statistical significance at p<0.05

No overall significance was found for the non-hearing impaired group for overall sound quality or for enjoyment using a repeated measures ANOVA.

For the hearing impaired group ratings for clarity, overall sound quality and enjoyment appeared to be closely related and the Pearson correlation coefficient for each rating combination was calculated to assess this factor. Table 7 shows the Pearson correlation coefficient of each combination, each of these is a statistically significant result at p<0.05. Although it could be argued that the correlation between these factors could have been an artifact of ratings being collected together the fact that there was no correlation between rating descriptors for the non-hard of hearing participants suggests that this was probably not the case.



	Enjoyment	Clarity
SQ	0.87	0.83
Enjoyment		0.78

Table 7. Correlation coefficients between scales for hard of hearing listeners, shaded cells indicate significance at p<0.05.

3.8. *Discussion*

3.8.1. LCR, LCR1, LCR2, C

In each case where statistical significance was found the clarity of the dialogue was perceived by both groups as having improved when the side channel levels were reduced by 6dB or more. For the hearing impaired group ratings of their enjoyment of the clips, and of the perceived overall sound quality, followed the same trend as for the perceived clarity of the dialog. For the non-hearing impaired group there was no overall significance to the ANOVA results for overall sound quality or for enjoyment.

3.8.2. LR (downmixed stereo)

This process scored significantly lower than any other in this test, speech was considered to be less clear than any other process by both groups, it was perceived to have a lower overall sound quality and to make clips less enjoyable compared to every other process by the hearing impaired group.

There are two possible explanations for this consistently poor rating of the *LR* condition for each rating scale where significance was shown. The poor rating of the stereo downmix may be as a result of the downmix process where the 6 channels in the AC-3 audio stream are remixed for 2 channel stereophonic reproduction. As has been discussed earlier, two types of downmix are specified by Dolby Labs, these are known as Lt/Rt (left total, right total) and LoRo (left only, right only) (Dolby Labs, 2003). Both downmixed formats are derived from a mix of all 5 full range channels including left and right rear surrounds. Lt/Rt is the default output for all current consumer devices and so was the chosen downmix format for these tests, LoRo, also defined by and sometimes referred to as the ITU downmix (ITU, 1994a), is the downmix that is specified for use where derivation of mono signals is required. The inclusion of rear surround information in this mix reduces the relative level of the centre channel, usually used for dialogue, and so was very likely to affect the clarity of the dialogue compared to the other conditions, none of which include rear surround audio. This could explain the relative perceived lack of clarity in the stereo mix, however if this were the only factor one might have expected non-hearing impaired participants to rate Lt/Rt stereo more highly for overall sound quality and enjoyment.

Although this result could perhaps have been predicted the default Lt/Rt derived two channel stereo provides a useful reference between what a viewer may be listening to now, and what improvements could be possible with surround sound equipment set up with a hearing impaired output derived from remixing the 5.1 channel audio at the STB. For the hearing impaired subject group, perceived overall sound quality and enjoyment was shown to be directly correlated with the clarity of dialogue. The ratings of the other processes indicate that hearing impaired viewers may benefit from reducing the level of lest and right channels, maximum benefit being gained by muting side speakers entirely. For the non-hearing impaired subject group although clarity was enhanced by reducing surround channel levels, this did not result in any statistically significant response to the perceived sound quality and enjoyment of the material.
The most striking result from the tests therefore was the low rating of the Lt/Rt stereo downmix when compared with all other conditions.

3.9. Analysis of Test Procedures

3.9.1. Introduction

When conducting subjective listening tests it is important to remember that there is unlikely to be a perfect experiment that will give conclusive, yes/no answers. Inevitably variables other than those being tested will influence results. Because of this it was considered important to review test procedures and material for these first tests using the data gained in order to attempt an analysis of any unpredicted causal effects. In designing these tests as many of these imperfections were taken into account as possible. For example, the clips and processes were rotated so that every clip was tested with every process, clips were chosen to be as similar as possible and each pair of processes was tested in A/B and B/A order to minimise any effect caused by the order of processes. It is recognised however that these measures may be only partially successful. This section is intended to enable improved test design for future work in this area and describes factors other than the conditions applied to the media that had an effect on the results.

3.9.2. Clip Dependency

Two factors that can defined as clip dependency were found to be influencing the choice of participants considerably: the order of the clip playback, and visual cues to the meaning of dialogue.

3.9.2.1. Playback order

Although it was anticipated that the order of the processes would have some effect on subject preference, the degree to which this affected results was unforeseen. The following three graphs (figures 12 to 14) show the percentage that section A was chosen over section B for each clip used, it clearly shows a marked preference for section B for most of the clips.

Clarity

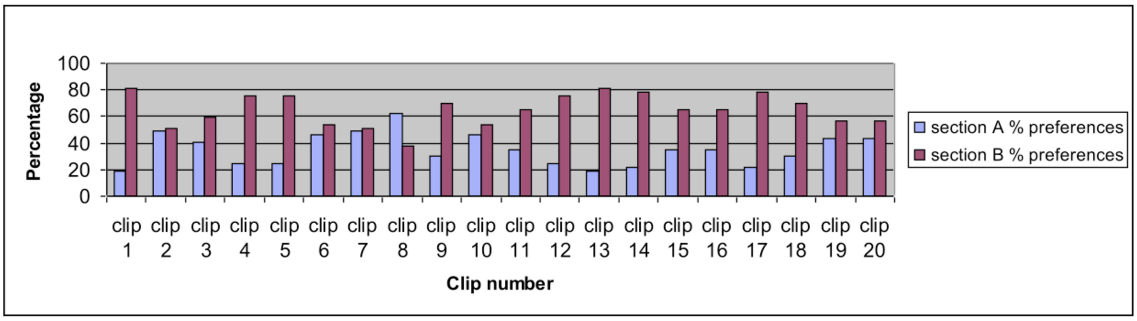


Figure 6. Percentage of preferences for clip A or B for speech clarity

Quality

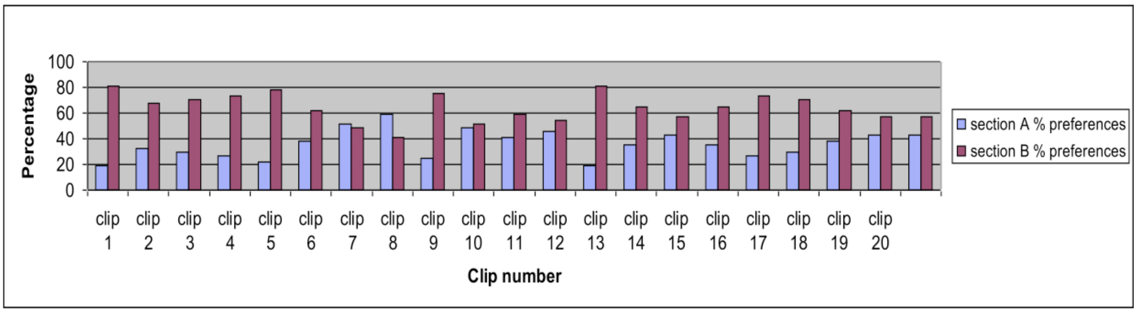


Figure 7. Percentage of preferences for clip A or B for overall sound quality

Enjoyment

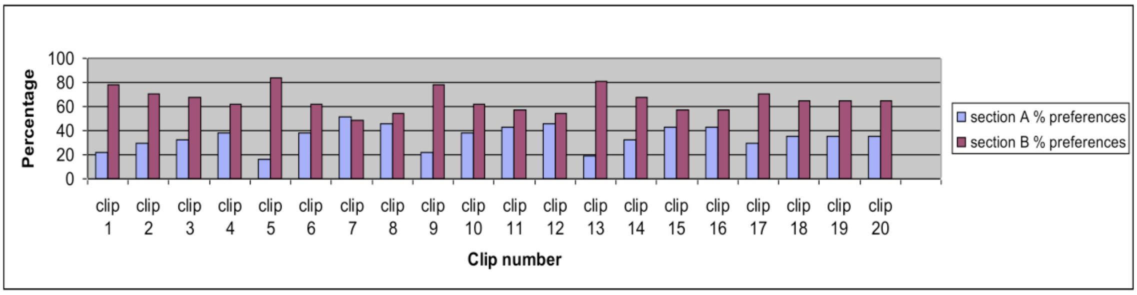


Figure 8. Percentage of preferences for clip A or B for enjoyment

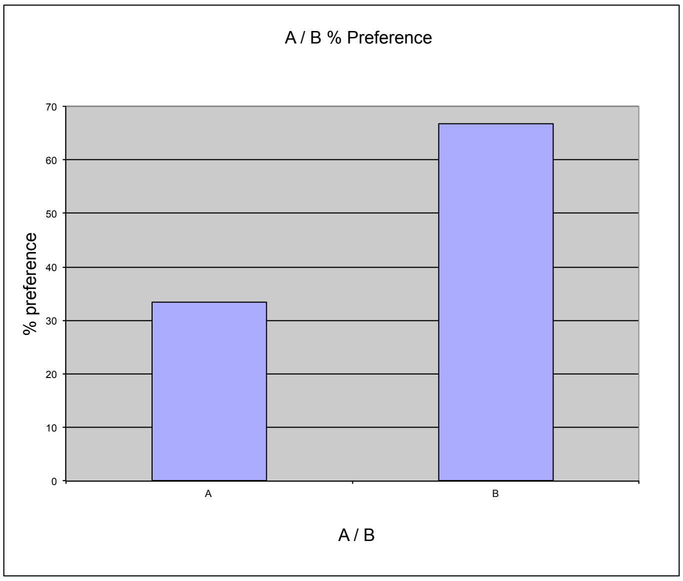


Figure 9 Overall preferences for clip A or B showing recency effect

The effect of this preference for section B was more pronounced among the hearing impaired participants although was marked for both groups (as shown in figure 9). A number of the hearing impaired participants commented that it often took most of the first section to get used to the accent of the actors and to “blank out” the background effects or music. This meant that they found section B considerably easier to understand and led to the higher preference for section B. The fact that this tendency was so prevalent with both groups suggests that there are other factors leading to a preference for the most recently heard section. One potential contributing factor is the so called ‘recency effect’ discussed in (Zielinski et al., 2008). In the tests carried out the order of the processes was arranged in order that processes were played first and second an equal number of times in order to avoid this effect biasing results. It is likely however that this unwanted variable caused some statistical ‘noise’ and so reduced the statistical significance of the results.

3.9.3. Analysis of Test Material

An analysis of the test material was carried out with respect to the type and loudness of background noise and the amount of dialogue spoken off-camera and facing away from camera compared to dialogue spoken where the speaker’s mouth was visible. Some results of this analysis are shown in Appendix C.

3.9.3.1. Visual Cues

It was considered likely that, consciously or unconsciously, people would be using visual aids to help their understanding of the dialogue; this might have included such factors as lip reading and gestures. In order to see how this impacted on the test, graphs in Appendix C are used to show a comparison of the clarity of each clip and the percentage of time the speaker was facing the camera. An analysis was carried out to assess the impact of this on participants’ rating of each clip. This was carried out for non-hearing impaired, and for hearing impaired participants. The graphs show that there was a tendency for participants with hearing impairments, especially moderately impaired hearing, to choose the section with the most ‘face to camera’ dialogue. This was only true, however for analysis of section B (the second section shown in each paired comparison). It is unclear why this should apply more to the most recently heard section however it may be an indication as to how much the bias against section A has reduced the significance of some of the data gained. Interestingly, when questioned on this, most participants were unaware of their reliance on visual cues in understanding the dialogue.

The Pearson’s correlation coefficient of the percentage of total preferences for a clip for clarity against the percentage of on-screen dialogue for the second section viewed only was calculated as 0.545 indicating a significant correlation between the two factors ($p<0.05$). A graph indicating the relationship is shown in figure 10 for all participants across all section B clips. For section A (shown in figure 11) Pearson’s correlation coefficient was calculated as -0.102 with no significance at $p<0.05$.

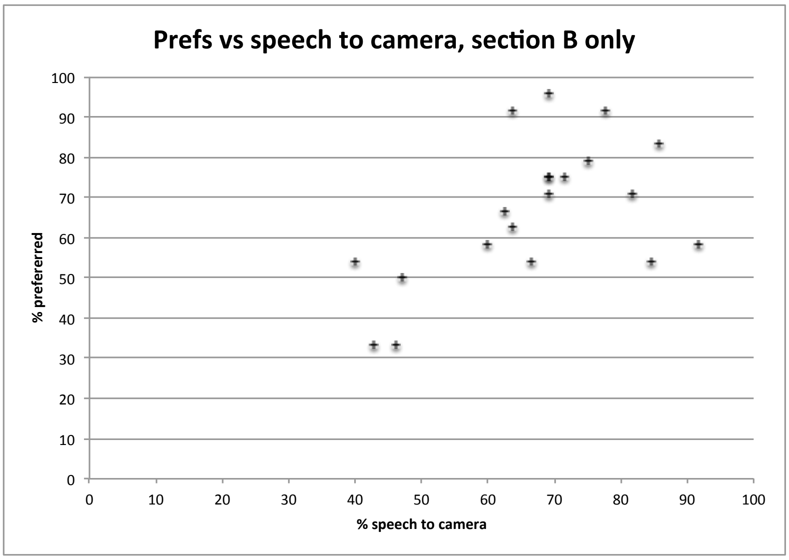


Figure 10. Scatter chart of percentage of on-screen dialogue against the percentage of times that clips were preferred for dialogue clarity across section B only (the most recent section heard)

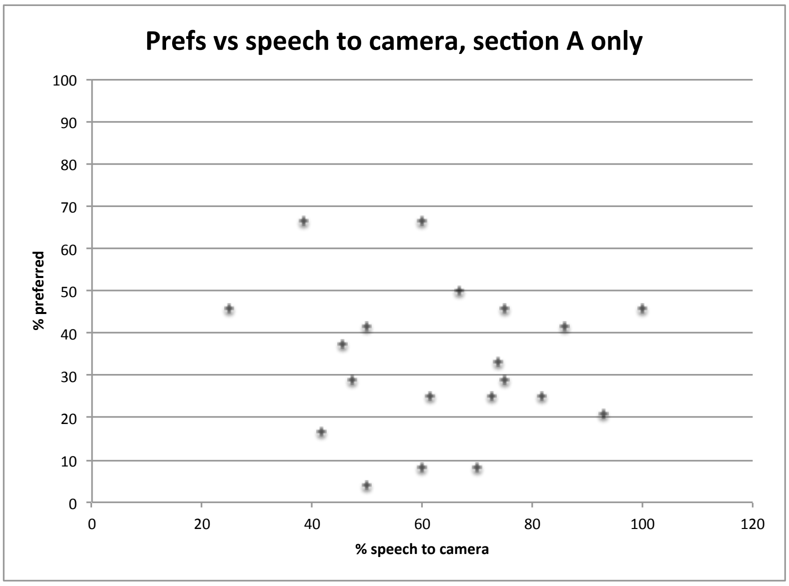


Figure 11. Scatter chart of percentage of on-screen dialogue against the percentage of times that clips were preferred for dialogue clarity across section A only (the first section heard)

Correlations for dialogue clarity and percentage of on-screen dialogue broken down by participants' degree of hearing impairment can be found in Appendix J.

This effect was considered to be unavoidable when using existing visual test material.

For hearing impaired participants enjoyment was strongly linked to the percentage of speech-to-camera with a Pearson correlation coefficient of around 0.523 ($p<0.05$) compared with a weaker, but still significant, correlation of 0.335 for speech clarity ($p<0.05$). This can perhaps be seen as a result of the fact that when asked, most hearing impaired participants were unaware that they were using lip reading at all during the tests. It could be hypothesised that for ratings of speech clarity participants had more of a conscious focus on the audio exclusively whereas enjoyment was considered as a more multi-modal rating across both audio and visual features. For non-hearing impaired participants there was a very weak negative correlation for both speech clarity and enjoyment with no statistical significance. On considering overall sound quality correlations for the hearing impaired group there is indication of some positive correlation between overall sound quality and the percentage of speech-to-camera dialogue mirroring results discussed earlier that indicated some correlation between ratings of clarity and sound quality for hearing impaired people. Again no substantial correlation is indicated for non-hearing impaired participants, for this group the amount of speech spoken with face clearly visible had little impact on perceived speech clarity, overall sound quality or enjoyment of the clip section.

3.9.4. Recommendations for Future Test Design

The analysis of test methods presented here indicates that there are potential clear biases which could have impact on subjective test results. Most of which were considered during test design stages however the strength of some of the potential biases was unexpected and some recommendations for future test design can be stated in order to reduce the impact of these factors on other subjective tests.

- The impact of the playback order of the clips should be minimised. This could be achieved by allowing the subject to switch between processes at will or by taking care to ensure that clips are presented first and second an equal number of times in any paired comparison test.

- Assessment of speech clarity and intelligibility should ideally be done without visual content where this is possible. In the case of intelligibility assessments this could be accomplished by utilising or adapting standard test material for assessment of intelligibility such as the Connected Speech Test (Cox R.M., 1987) or the Revised Speech in Noise Test (Bilger R.C. Nuetzel J.M. Rabinowitz W.M. Rzeczkowski C, 1984). In the case of audio for TV this is a step removed from a 'real-world' scenario and so unsuitable for the comparisons documented here but would ensure more accurate assessment of the audio conditions free from other influences. Where visual material is essential to tests (for example when attempting to assess 'enjoyment' of AV media) great care should be taken to ensure that clips are as equivalent as possible.

- When using visual content care should be taken to ensure that all clip/process combinations are tested. This should minimise the effects of clip dependency on results.

3.10. Conclusions

3.10.1. Downmixed Stereo

The most statistically significant finding to come out of all of the phase 1 tests has been the assessment of downmixed stereo. In A/B comparison tests it was the least preferred process, both groups consistently preferred every other process.

As has been described earlier in this report the downmixed stereo is derived from the full 5.1 channels and is not a separate stereo track such as can be found on current broadcast transmissions and many DVDs. It was not, therefore, possible to make generalised judgements on stereo from these results. It seemed likely that the perceived clarity of the downmixed stereo was affected by the presence of rear surround channels in the mix. The downmixed stereo soundtrack is the only option available for listening to many DVDs and surround sound broadcasts if one has only stereo audio reproduction equipment.

From these tests we know that downmixed Lt/Rt stereo was considered to make dialog less clear, to have a lower overall sound quality and to make clips less enjoyable than using discrete front surround and dialogue channels.

3.10.2. Level of Left and Right Front Surround Channels

For hearing impaired participants, perceived overall sound quality and enjoyment appeared to be directly related to the clarity of dialog. Close correlation between all three factors was indicated at $p<0.05$. Hearing impaired viewers could benefit from left and right channels being reproduced at lower levels, maximum benefit was gained by muting side speakers entirely. For non-hearing impaired participants clarity was enhanced by reducing left and right channel levels however there was no statistically significant evidence of impact on perceptions of overall sound quality and enjoyment. There was some weak evidence to suggest that it may have detracted from the perceived sound quality and enjoyment of the material.

3.10.3. Summary

From these tests we can say that hearing impaired viewers can improve clarity, sound quality and enjoyment of 5.1 AV media by muting side channels and listening solely to the dialogue channel where dialogue is present. Hearing impaired viewers sharing a television with non-hearing impaired viewers may be able to listen to television with improved clarity, sound quality and enjoyment by reducing the level of the side channels. It is possible that by experimenting with the level of these channels they may be able to improve clarity for everyone without significantly detracting from the enjoyment of non-hearing impaired viewers. This is not a straightforward process using current AV receiving equipment however metadata could be utilised to activate such a hearing impaired mode in future.

3.11. Recommendations for Hard of Hearing Viewers Published by Ofcom

- Where only a 5.1 surround soundtrack is available, use of discrete left, centre, right (LCR) channels can improve clarity, perceived sound quality and enjoyment of programme material, compared to downmixed stereo, for both hearing impaired and non-hearing impaired viewers.

- Hard of hearing viewers can significantly improve the dialogue clarity of Dolby Digital 5.1 programme material television by listening to centre (dialogue) channel only. This can result in a perceived improvement in sound quality and may enhance their enjoyment of the programme material.

- Hard of hearing viewers sharing a television can benefit from lowering the level of the surround channels. This may be less detrimental to the enjoyment of non-hearing impaired viewers than removing surround channels completely but can still improve dialogue clarity.

- These recommendations have the most benefit for those having a moderate hearing impairment.

3.12. Further Work

The questions arising from the research documented in this chapter led to further assessments that are documented in following chapters as follows:

- Stereo reproduction: further work was carried out in order to investigate the poor ratings of downmixed stereo derived from a 5.1 soundtrack in order to ascertain whether implementing the proposed solution would also be beneficial for people listening to broadcast audio on two channel stereo systems.

- Dynamic Range Control: it is possible that the dynamic range control available in existing DVD players and AV receivers may provide benefit for hearing impaired viewers. To this end tests are documented that ascertain the effect of dynamic range control compression processing on the perception of programme material for hearing impaired people with a range of impairments. This could potentially lead to recommendations as to how hearing impaired viewers could utilise existing equipment settings in order to improve perceived sound quality.

- Compression Techniques: In addition to the above, the literature presented in the chapter 2 raises a question as to whether band limited compression techniques, such as those used in hearing aid design, could have potential to facilitate more inclusive product design for hearing impaired people.

4. Clean Audio Phase 2: A Solution for Downmixed Stereo Reproduction?

A factor that came out of informal discussions with participants in the previous phases of this research was that although hearing impaired viewers are the most likely to benefit and hear improvements from using multichannel reproduction, they are the least likely adopters of surround sound technology. This makes it critical to understand how attenuation of non-speech channels, one potential solution, will affect outcomes for people using more common, two-channel stereo reproduction equipment. It has already been shown that the default stereo mix, Lt/Rt, is detrimental to clarity and perceived sound quality compared to reproduction with discrete left, centre and right channels. Phase 2 of the Clean Audio Project aimed to ascertain the impact of implementing the solution discussed in chapter 3 (non-speech channel attenuation) for the majority of people, i.e. when presented over Lt/Rt downmixed stereo.

An additional factor investigated in this phase involved compression. The decision to apply compression to an audio signal in order to aid speech understanding uncovers a multitude of options that must be carefully considered. Should the same compression characteristics be applied across the whole of the frequency spectrum, or should it be implemented differently over two or more frequency channels? Implementing multiple channel compression adds the complication of how to overlap adjacent channels. How fast acting should the compression be and what speed of attack and release should the compression act at in order to preserve important features of the sound such as a speech envelope? In addition it is by no means certain that STB manufacturers would add additional processing for what is perceived as a ‘niche audience’. Dolby multichannel decoders come equipped with their own type of compression, *dynamic range control* (DRC). It was unclear from reviewing previous research on compression and hearing loss what impact DRC may have for hearing impaired viewers and so this was also factored into this phase of the experimental work. Therefore the work in this phase aimed to assess the effect that the DRC option as already implemented in Dolby Digital equipment has on the sound quality, enjoyment, and speech clarity of television sound for hard of hearing viewers with the aim of enhancing the clarity of the dialogue, the overall sound quality and their enjoyment of the programme material. These tests were considered important as any improvements gained using this processing would be a ‘zero cost option’ for hearing impaired viewers and could readily be implemented on existing equipment. Phase 2 of the Clean Audio project also included preliminary negotiations with Dolby with a view to facilitating the implementation of findings from phase 1.

The type of compression used in this pilot study is encoded in the DRC profile found in the AC-3 and E-AC-3 metadata. Dolby’s DRC is unusual for compression systems in that it amplitude compresses the audio relative to the average level of dialogue. The bit stream contains metadata values giving the average level of the dialogue; audio with an amplitude significantly lower than this level is amplified, audio with a level significantly higher is attenuated. The levels of gain and attenuation are dependent on the type of programme material contained within the AC-3 stream. The values for ‘film standard’ (utilised in this research) at the knee points of the gain plot shown in figure 12 relative to the dialogue level are listed below:

- Max Boost: 6dB (below -43dB)
- Boost Range: -43 to -31dB (2:1 ratio)
- Null Band Width: 5dB (-31 to -26 dB)
- Early Cut Range: -26 to -16dB (2:1 ratio)
- Cut Range -16 to +4dB (20:1 ratio)

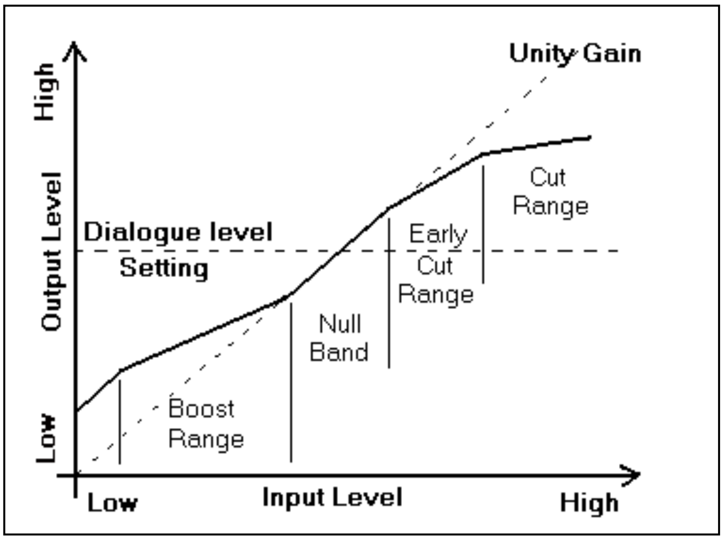


Figure 12. Dolby Dynamic Range Control compression indicating gain ranges centred around the defined dialogue normalisation level (dialnorm) from Dolby Metadata Guide (Dolby Labs, 2003)

The implications for inappropriate dialnorm settings are clear from the figure above; dialogue can either be raised or lowered in the mix depending on whether the value is set too high or too low, if dialnorm is set at too low a level dialogue will be reduced in level whereas lower level audio content will be raised leading to increased problems for all viewers especially hearing impaired people.

4.1. Methodology

This phase of the project involved assessment of dialogue clarity, enjoyment and overall sound quality for a series of DVD clips in the following listening conditions:

- Lt/Rt stereo at reference levels
- Lt/Rt stereo with left and right channels at -6dB
- Lt/Rt stereo centre channel only (reproduced as phantom centre)
- Lt/Rt stereo with Dynamic Range Control
- Lt/Rt stereo with Dynamic Range Control and left and right channels at -6dB

The methodology adopted was that of forced choice comparison blind A/B listening tests identical to that utilised in Clean Audio Phase 1. Participants were asked to compare video material accompanied by a Dolby Digital 5.1 soundtrack presented using a 2 loudspeaker stereo reproduction system.

4.2. Experimental Setup

Equipment used

- DVD Player
- Digital Mixing Console
- Dolby DP562 Reference Decoder
- 2 x Genelec 1029 Active Monitors
- Notebook PC and MIDI interface

Equipment set up was as follows:
The subject’s chair, the reference monitors and the television were placed in the ideal listening positions according to the *ITU-R BS.775-1 multi-channel stereophonic sound with and without accompanying picture* recommendations. Control equipment was situated at the back of the room with the operator out of sight of the viewer. The digital S/PDIF output from the DVD player was converted to AES/EBU standard using a converter and connected to the AES/EBU input on the Dolby Reference Decoder.

The DP562 reference decoder has a number of engineering and test modes of operation that allow flexible handling of inputs and outputs. The test procedure was carried out using the Lt/Rt three channel output (L, R & C) downmix, with and without the DRC setting. In this mode of the DP562 decoder the five full range channels of the 5.1 surround input are downmixed to Lt/Rt with centre channel remaining as a discrete output instead of being mixed at -3dB into left and right channels. The L, R and C channels were routed, via the internal busses of the mixer, to two outputs, each of which was connected to the corresponding monitor, L and R channels could then be attenuated relative to centre in order to achieve the 6dB reduction found to be useful in the previous phase of the research. Preliminary testing showed that the overall sound pressure level (SPL) was being altered by DRC as well as by L and R channel attenuation and the subject's rating of what was the 'best' section was therefore likely to be biased. It was important that there was consistent loudness for all of the AB comparison tests and so for each test the levels were equalised using an automated procedure. The level of the L/R channel pair was measured using a $\text{Leq}_{\text{(clip duration)}}$ (measured in dB(A)) and the difference calculated. This difference was applied appropriately by adjusting the overall bus output levels from the mixing console for each condition using control software capable of sending MIDI controller information in response to experimenter input.

4.3. Test Material

Previous work in phase 1 of clean audio (documented in chapter 3) demonstrated the need for testing using multiple clips so a series of 20 video clips with a Dolby Digital 5.1 soundtrack were selected for testing. Clips were selected using the same criteria as for phase 1.

Each clip was split into two sections and introduced by a title reading "Clip x" (where x is the number of the clip), followed by a title reading "Section A" for the first part of the clip and then "Section B" for the second part of the clip. Again it was thought important in choosing the clips that each clip should appear to be complete in itself, i.e. at the end of a clip the subject matter is brought to some sort of conclusion. The length of each clip was standardised as far as possible.

4.4. Participants

Twenty hard of hearing participants and twenty non-hard of hearing participants were selected to take part in the test. The age distributions of these participants is shown in figure 13, their gender distributions are shown in figure 14, and the classification of the hearing losses of the hard of hearing participants is shown in figure 15. The classification system used is explained in Appendix G, according to this system six of the hard of hearing participants had asymmetric hearing losses the other fourteen had bilateral losses.

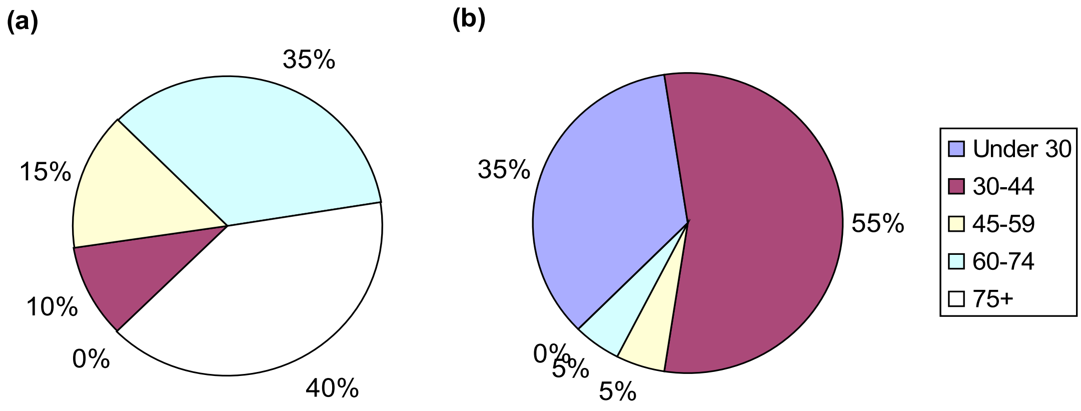


Figure 13. Age distribution of (a) hard of hearing participants and (b) non hard of hearing participants

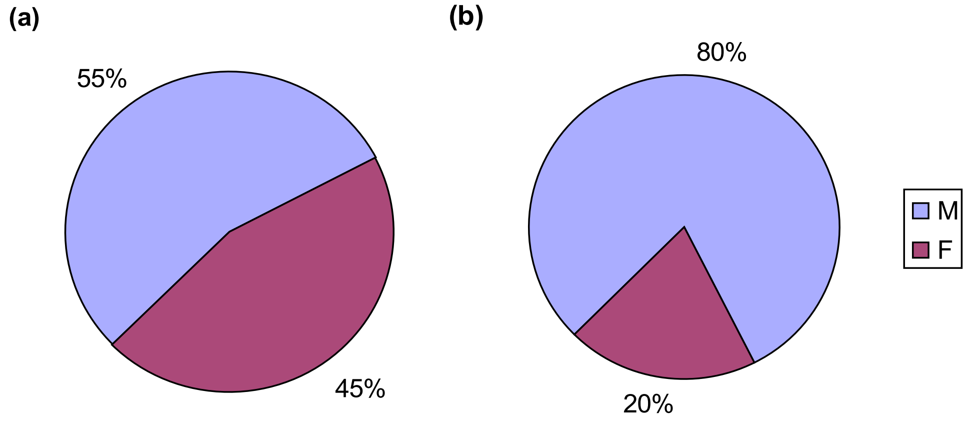


Figure 14. Gender distribution of (a) hard of hearing participants and (b) non hard of hearing participants

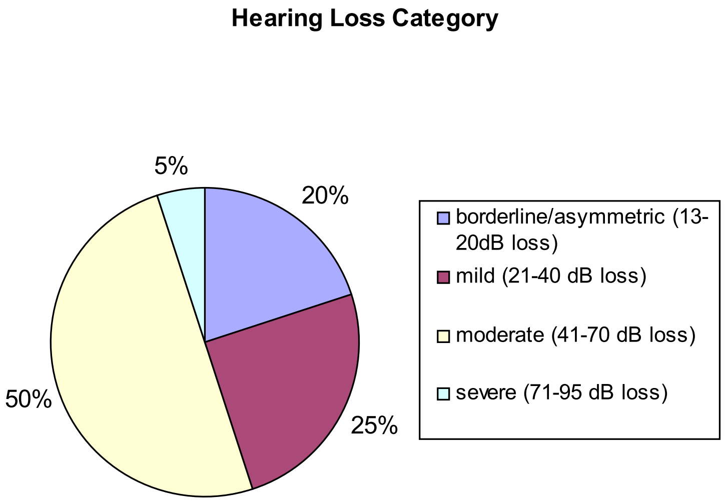


Figure 15. Hearing losses of hard of hearing participants

4.5. Experimental Results

A detailed explanation of the data normalisation and analysis can be found in appendix H. The data was normalised according to this procedure. A series of repeated measure ANOVAs were carried out for clarity, overall sound quality and enjoyment ratings. Conditions assessed were as follows:

LR	Lt/Rt stereo at reference levels
LR6dB	Lt/Rt stereo with left and right channels at -6dB
Centre	Lt/Rt stereo centre channel only (reproduced as phantom centre)
DRC	Lt/Rt stereo with Dynamic Range Control
DRC6dB	Lt/Rt stereo with Dynamic Range Control, left and right channels at -6dB

Speech Clarity

For the hearing impaired subject group the Lt/Rt stereo condition (*LR*) was rated significantly lower for clarity than the *LR6dB* condition and for reproduction with DRC applied. When compared to *DRC6dB* the result is close to statistically significant although falling short of the required confidence level of 95%. Once again this *Lt/Rt* condition has an overall mean rating lower than any other condition tested, this mirrors results from chapter 3 which used three channel reproduction. Centre channel only, when presented as a phantom centre, received the second lowest mean rating for speech clarity, its low ratings were however statistically significant only when compared to the *LR6dB* condition meaning that few firm conclusions could be drawn. The *LR6dB* condition, with left and right channels attenuated by 6dB, received the highest mean rating of all conditions and for all comparisons apart from against reproduction with DRC applied these results were statistically significant.

Considering these results alongside those presented in chapter 3 it appears that left and right channel with 6dB of attenuation once again improved perceived speech clarity for the hearing impaired group.

Overall Sound Quality

Mean ratings for overall sound quality for the hearing impaired group largely mirrored speech clarity ratings; the *LR* condition was rated on average worse than both *LR6dB* and *DRC* ($p<0.05$), ratings when compared to *centre* and *DRC6dB* had no statistical significance. Centre channel only (*centre*) was rated lower than *DRC* and *LR6dB* ($p<0.05$). Of the conditions with dynamic range control applied *DRC* was rated higher than *LR*, *Centre* and *DRC6dB* ($p<0.05$) indicating that although dynamic range control improved the Lt/Rt downmix for overall sound quality, reduction in left and right channels was detrimental to these ratings when DRC was applied.

In pairwise comparisons of overall sound quality for non-hearing impaired participants the only statistically significant ratings were that all conditions were rated more highly than the *centre* condition ($p<0.05$).

Enjoyment

As was the case with chapter 3 results the enjoyment ratings carried less statistical significance, only results for *LR6dB* when compared to *LR* and *centre* indicated any significant preference ($p<0.05$) for hearing impaired participants.

For non-hearing impaired participants all conditions except for *LR6dB* were preferred to the *centre* condition. *LR6dB* compared to *centre* indicated no statistical significance at a 95% confidence level.

4.5.1.1. Hearing Impaired Group (20 participants)

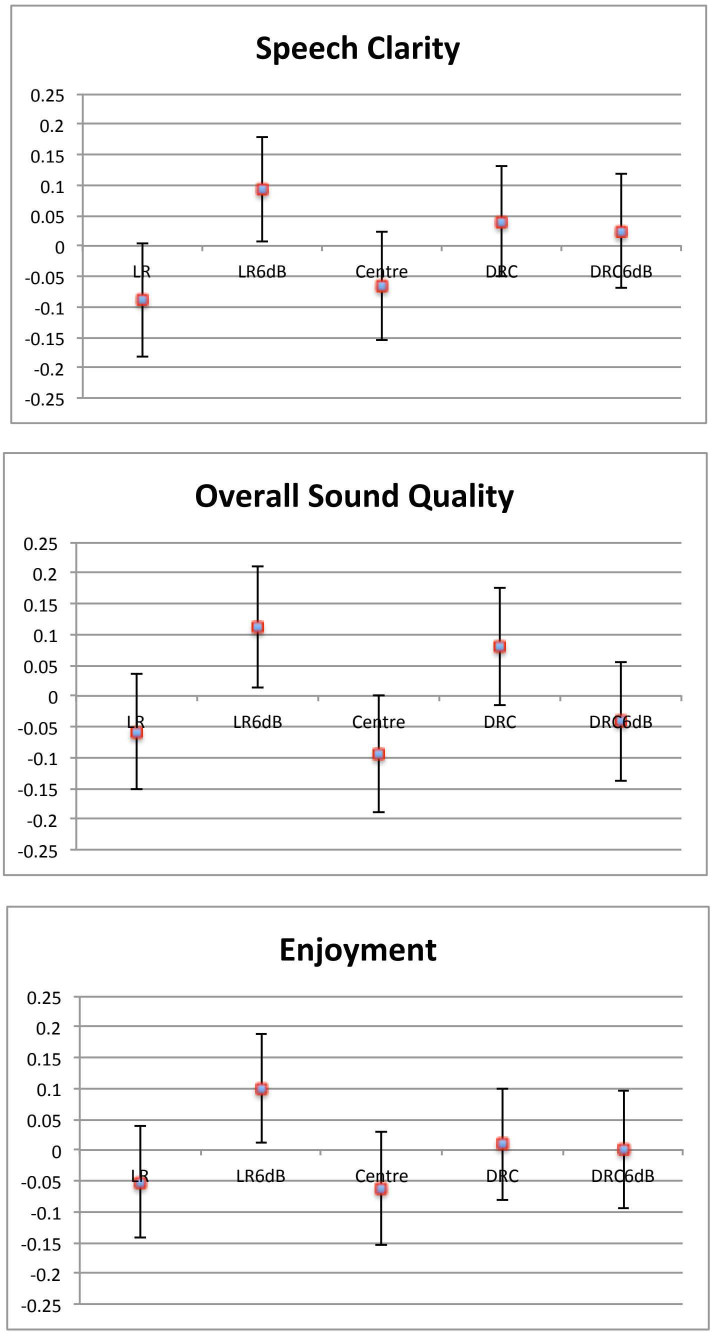


Figure 16 A, B & C. Plots showing values obtained for speech clarity, overall sound quality and enjoyment for the hearing impaired group. Error bars indicate 95% confidence interval.

4.5.1.2. Non-Hearing Impaired Group (20 participants)

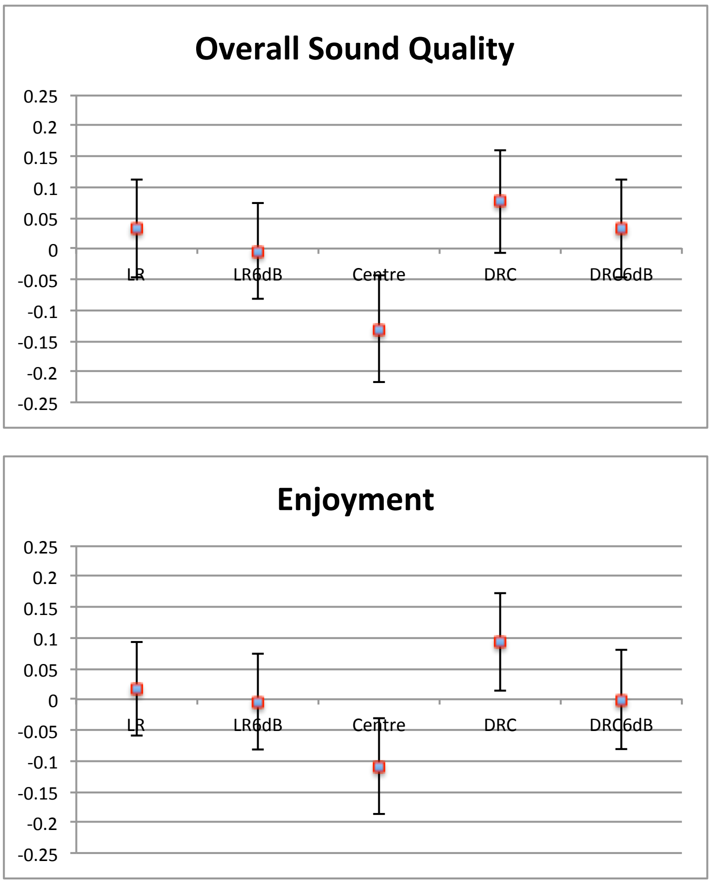


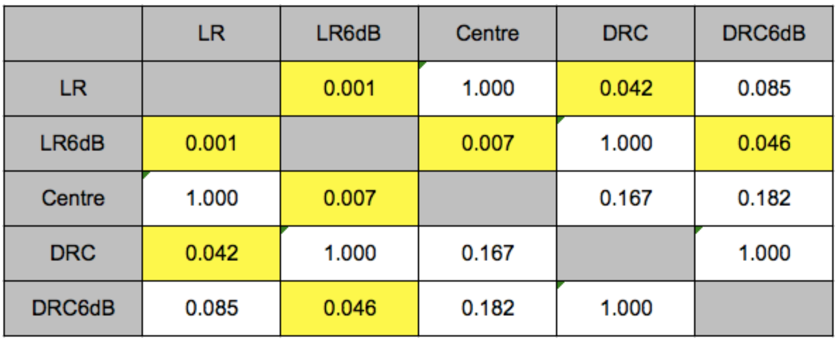
Figure 17 A & B. Plots showing values obtained for overall sound quality and enjoyment for the non-hearing impaired group. Error bars indicate 95% confidence interval.

4.6. Significance of Results

Tables 8 to 13 show the statistical significance of each pairwise comparison based on the repeated measures ANOVA. Highlighted boxes show significant data at p<0.05.

4.6.1. Hearing Impaired Participants

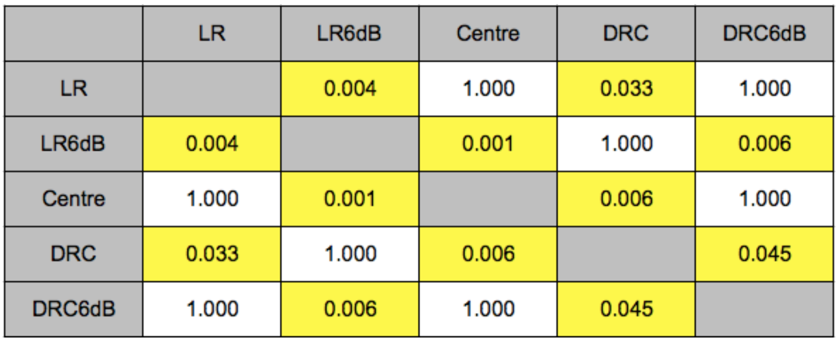
Significance: Speech Clarity



	LR	LR6dB	Centre	DRC	DRC6dB
LR		0.001	1.000	0.042	0.085
LR6dB	0.001		0.007	1.000	0.046
Centre	1.000	0.007		0.167	0.182
DRC	0.042	1.000	0.167		1.000
DRC6dB	0.085	0.046	0.182	1.000	

Table 8. P-values for each pairwise comparison for speech clarity ratings, highlighted values <0.05 indicate statistical significance

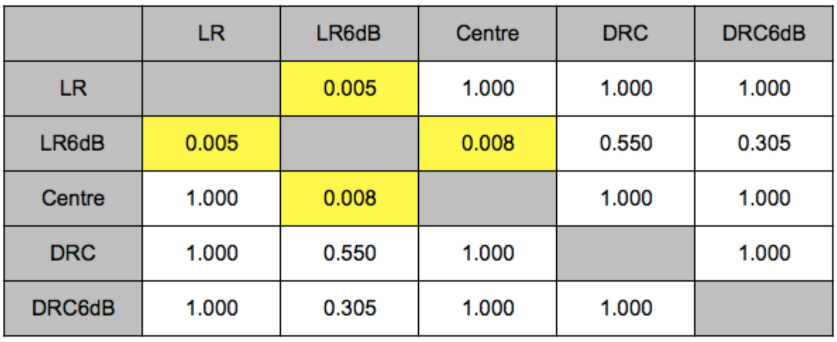
Significance: Overall Sound Quality



	LR	LR6dB	Centre	DRC	DRC6dB
LR		0.004	1.000	0.033	1.000
LR6dB	0.004		0.001	1.000	0.006
Centre	1.000	0.001		0.006	1.000
DRC	0.033	1.000	0.006		0.045
DRC6dB	1.000	0.006	1.000	0.045	

Table 9. P-values for each pairwise comparison for overall sound quality ratings, highlighted values <0.05 indicate statistical significance

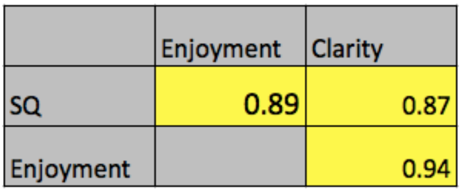
Significance: Enjoyment



	LR	LR6dB	Centre	DRC	DRC6dB
LR		0.005	1.000	1.000	1.000
LR6dB	0.005		0.008	0.550	0.305
Centre	1.000	0.008		1.000	1.000
DRC	1.000	0.550	1.000		1.000
DRC6dB	1.000	0.305	1.000	1.000	

Table 10. P-values for each pairwise comparison for enjoyment ratings, highlighted values <0.05 indicate statistical significance

Pearson correlation coefficients were calculated to investigate the apparent correlation between ratings for the hearing impaired group. Results indicated a strong correlation between all three factors rated in this experiment as can be seen in table 11. The results show that for the hard of hearing group sound quality, enjoyment and clarity are all closely interrelated.



	Enjoyment	Clarity
SQ	0.89	0.87
Enjoyment		0.94

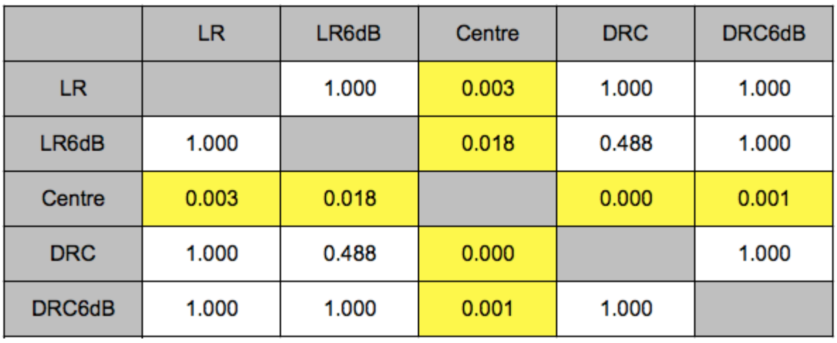
Table 11. Correlation coefficients between scales for hard of hearing listeners, shaded cells indicate significance at p<0.05.

4.6.2. Non Hard of Hearing Participants

Significance: Speech Clarity

There were no statistically significant outcomes for speech clarity rating for this subject group at a 95% confidence level.

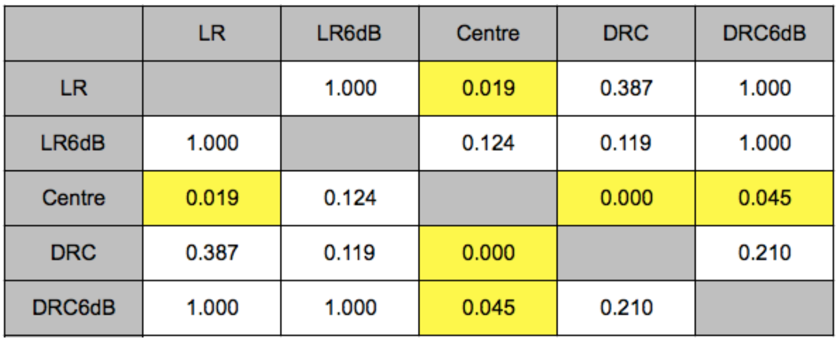
Significance: Overall Sound Quality



	LR	LR6dB	Centre	DRC	DRC6dB
LR		1.000	0.003	1.000	1.000
LR6dB	1.000		0.018	0.488	1.000
Centre	0.003	0.018		0.000	0.001
DRC	1.000	0.488	0.000		1.000
DRC6dB	1.000	1.000	0.001	1.000	

Table 12. P-values for each pairwise comparison for overall sound quality ratings, highlighted values <0.05 indicate statistical significance

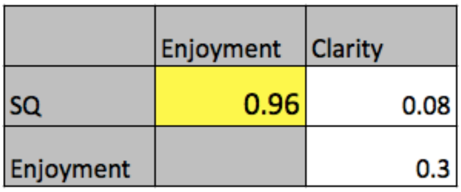
Significance: Enjoyment



	LR	LR6dB	Centre	DRC	DRC6dB
LR		1.000	0.019	0.387	1.000
LR6dB	1.000		0.124	0.119	1.000
Centre	0.019	0.124		0.000	0.045
DRC	0.387	0.119	0.000		0.210
DRC6dB	1.000	1.000	0.045	0.210	

Table 13. P-values for each pairwise comparison for enjoyment ratings, highlighted values <0.05 indicate statistical significance

Calculating Pearson correlation coefficients between the three ratings indicates a strong correlation between overall sound quality and enjoyment as can be seen in table 14.



	Enjoyment	Clarity
SQ	0.96	0.08
Enjoyment		0.3

Table 14. Correlation coefficients between scales for non hard of hearing listeners, shaded cells indicate statistical significance at p<0.05.

4.7. Discussion

4.7.1. Hearing Impaired Group

This work using the default stereo Lt/Rt downmix indicates that the recommendations generated from chapter 3 that were effective for three channel reproduction (attenuating left and right channels by 6dB) may have also improved perceived speech clarity and overall sound quality for hearing impaired viewers when presented over a two channel reproduction system. There is also evidence to suggest that applying Dolby's dynamic range control compression process may have improved speech clarity and overall sound quality for this group.

The removal of non-speech channels entirely had been expected to improve speech clarity ratings when compared to the default Lt/Rt stereo downmix however no evidence was found to substantiate this expected result. The rating of centre channel only was rated more poorly than the attenuated left and right condition (*LR6dB*) for both perceived speech clarity and for overall sound quality when presented over two loudspeakers. It could have been expected that there may be some reduction in ratings due to the centre channel being presented as a phantom centre rather than over a discrete loudspeaker however this was the case for all of the conditions assessed here. This possibility is investigated later in chapter 5 of this thesis.

The question remains however as to why the centre only condition was so poorly rated even when compared to other conditions with speech presented using a phantom centre. The results here give no firm conclusion to this question although do raise an interesting point about the differences between assessment of intelligibility versus speech clarity. It is possible that there could be some positive impact from the presence of side channel information that may hide some audible distortion or imperfection caused by acoustical crosstalk. Had the test been for intelligibility, logic would suggest that participants would have recognised more keywords with less background, side channel, sounds present. When perceived *speech clarity* is being tested it may be that participants were trying to assess the clarity of the speech signal in isolation from the background sound. With no background sound present the speech may have sounded *wrong,* or *unclear*, because of frequencies being cancelled out. Those same frequency peaks and troughs may have been disguised to some extent by the presence of side channel sounds that were sufficient to 'fill in' some of the frequency gaps but not sufficiently loud as to cause difficulty in understanding the speech content. Indeed Vickers (Vickers, 2009a) paper points out that such notches in the frequency spectrum caused by two channel reproduction are often filled in by the effect of room reflections. Furthermore he suggests that, *"When audio content includes correlated (center) and decorrelated (side) information, only the center content is subject to the comb filtering, reducing the salience of the notches."*.

4.7.2. Non-Hard of Hearing Group

Much less significance was found for the non-hearing impaired group. Centre channel only, when reproduced over two loudspeakers as a phantom centre, rated lower than all conditions for overall sound quality ($p<0.05$) and for enjoyment compared to all conditions apart from *LR6dB* ($p<0.05$) for which result no significance was found.

Correlations between ratings showed that for hearing impaired participants speech clarity, overall sound quality and enjoyment were all closely correlated and that for the non-hearing impaired group only overall sound quality and enjoyment showed correlation.

4.8. Further Work

While it would have seemed reasonable that centre channel only would have been rated highly for clarity even if this were not reflected in sound quality ratings this was not the case. There was clearly a substantial difference in perception of the conditions reproduced over two channel compared to three channel reproduction. Further work was therefore required in order to ascertain the cause of the poor performance of the phantom centre channel only when using Lt/Rt downmixed stereo in these tests and to assess if this could be shown to be the consequence of acoustical crosstalk.

4.9. Implications for TV Viewers

The research presented here lent further weight to the premise at the root of discussions undertaken with Ofcom and Dolby Laboratories aimed at providing a hard of hearing setting for STBs. Previous research showed distinct benefits to using attenuated left and right channels in order to improve clarity, perceived sound quality and enjoyment in a 3 channel reproduction system utilising a centre loudspeaker such as that found in surround sound systems. This research indicates that the same technique could be effectively implemented in a 2 channel stereo reproduction system utilising the Lt/Rt downmix that is standard on set top boxes, DVD players and other Dolby equipment. Furthermore potential benefits for hearing impaired people could be gained by utilising Dolby's *dynamic range control* compression processing. This has the potential to provide solutions for hard of hearing viewers who do not yet have surround sound reproduction equipment and who rely on the downmixed-to-stereo sound produced by STBs as default for two channel reproduction.

5. The Problem with Stereo

Results in this chapter have been published in the Journal of the Audio Engineering Society article, *The Effect of Stereo Crosstalk on Intelligibility: Comparison of a Phantom Stereo Image and a Central Loudspeaker Source* (Shirley et al., 2007).

5.1. Introduction

The growth of digital television and the implementation of surround sound broadcast make work into the intelligibility of sound formats timely. Holman (Holman.T., 1991) has carried out experiments subjectively comparing a phantom centre image and a real centre source. The results showed that the addition of the centre channel appeared to make the dialogue clearer. It was suggested that participants found each method as intelligible as the other but that more effort may have be required to understand the stereo playback. It can be argued that this seems counter-intuitive, especially when applied to older people with cognitive and other factors that may slow understanding of speech. Holman also mentions the problem of acoustical crosstalk in stereo reproduction (Holman.T., 1996). This occurs when sound from both loudspeakers reach the ear at slightly different times causing a comb-filtering effect and a dip in amplitude at around 2 kHz (Holman.T., 1996). This is particularly apparent when creating a central phantom stereo image where both sources give the same signal. At each ear there is a signal and a slightly delayed version of that signal which causes the filtering effect. This effect was originally noted by Snow (1953). The potential importance of the effect of crosstalk is also noted by Toole (1985). Dressler (1996) suggests an example of intelligibility problems where the commonplace downmixing from 5.1 surround to two channel stereo can lead to excessive level causing clipping that could “alter the subjective balance of the mix to the detriment of dialogue intelligibility”. It is possible that some of this detrimental effect on dialogue intelligibility may not be solely as a result of the process of downmixing and resultant clipping noted by Dressler. This chapter describes research assessing if there is an actual measurable improvement in intelligibility (as opposed to perceived clarity) by the addition of a central loudspeaker for the centre channel of 5.1 material and to assess any benefit to utilising a centre loudspeaker for speech. The background and the methodology of the experiments are explained, results and data analysis for pilot tests and the main series of tests are presented and results are discussed and considered with respect to measured data.

5.2. Methodology

5.2.1. Overview

Previous subjective testing described in this thesis looked at perceived speech clarity, overall sound quality and enjoyment using audio with accompanying picture from DVD, encoded with a Dolby Digital 5.1 soundtrack. An analysis of the test procedures from these tests indicated that subject ratings were influenced by the amount of 'face to camera' speech in each clip. Although this effect was predicted and the test designed to nullify any influence on the results it seems likely that this factor would have reduced the statistical significance of the results. It is also quite possible that there can be conditions whereby dialogue will appear to be clearer and yet not be any more intelligible. In Holman's experiments, he saw no improvement in intelligibility but noted that it may take more effort to understand the speech presented as a phantom centre image and in order to assess *intelligibility* as opposed to *perceived clarity* it was decided to assess audio in isolation, with no accompanying picture.

It was decided that for the purposes of investigating any intelligibility impact of a phantom centre channel only non-hearing impaired participants would be required. The inclusion of hearing impaired participants was considered to introduce further variables that would reduce the likelihood of gaining significant results. Factors such as varying degrees of hearing impairment between participants and the unknown impact of asymmetric hearing loss were considered to reduce the likelihood of finding interesting generic findings about two channel stereo reproduction effects on intelligibility.

5.2.2. Speech in Noise Testing

Audiometric measures of hearing loss are used to assess the acuity of an individual's hearing for specific frequencies however this does not necessarily correspond with the ability of a person to understand speech in everyday situations. Some research (Festen and Plomp, 1983) indicates that although the ability to understand speech in quiet is largely determined by audiometric loss, ability to understand speech in noise is more closely related to frequency resolution - our ability to resolve adjacent frequencies. Poor frequency resolution makes it more difficult to distinguish between tones, but also, in the context of speech comprehension, between different phonemes. Summerfield states that, “audiometric loss is generally associated with poor frequency resolution” (Summerfield, 1987) but also, that “impairments in gap detection and frequency discrimination are not so associated with audiometric loss, and account for significant amounts of variability in speech understanding after association with audiometric thresholds have been taken into account.” This suggests that one of the most critical factors in our ability to understand speech is one which is not measured by the usual method of assessing hearing acuity, the audiogram. Intelligibility of speech itself is dependent on many contributing factors. Identification of small phonetic information elements are a part of comprehension, but also larger prosodic parameters, such as variations in the pitch and rhythm of speech and also contextual information aid understanding. For this reason testing of speech perception in noise using sentences with and without helpful context has been used to determine the hearing ability of individuals in ‘day-to-day situations’ (Bilger R.C. Nuetzel J.M. Rabinowitz W.M. Rzeczkowski C, 1984) and a number of tests exist for that purpose.

One of these tests, the Speech in Noise test (SIN) by Killion and Vilchur (1993) is used to measure the intelligibility of speech against a background of multi-talker pseudo babble. The test uses a series of sentences with multiple keywords within each sentence. The test produces an SNR-50 (dB) value that represents the *speech to babble* ratio where the listener gets 50 percent of the keywords correct. A variation of the SIN test was proposed by Kalikow et al (Kalikow D N. Stevens K N. Elliot L L, 1977); the Speech Perception in Noise test (SPIN) has been used to assess the degree to which linguistic-situational information is utilised in everyday speech reception. The test consists of 10 *forms* of 50 sentences per form. Each of these sentences ends in a keyword that must be identified by the subject and this is used for scoring. The forms each have 25 sentences that end with low predictability key words (where the contextual relevance of the sentence to the keyword is low) and 25 ending in high predictability key words (where the contextual information relates strongly to the final keyword in the sentence). Each form has a *cognate*, or paired, form containing the same key words but reversing the predictability. For example one form presents the keyword ‘spoon’ in a contextual setting; “Mr Brown stirred his coffee with a spoon”. Its cognate form presents the same keyword in a non-contextual setting; “Mrs White considered the spoon”. The forms were later refined by Bilger et al (Bilger R.C. Nuetzel J.M. Rabinowitz W.M. Rzeczkowski C, 1984), in order to improve equivalency between forms. Equivalence assessment was carried out by analysis of phonetic content and analysis of results from extensive testing with hearing impaired participants to ensure that equivalent keyword recognition scores were obtained for each form and no forms contained easier or more difficult keywords than any other. In the SPIN tests, the noise used to mask the speech is multi-talker babble with 12 talkers. It was designed to be carried out at a speech to babble ratio of +8dB for hearing impaired participants without hearing aids. The revised SPIN test media (Bilger R.C. Nuetzel J.M. Rabinowitz W.M. Rzeczkowski C, 1984) was used as the basis of this research largely because of the extensive efforts made to maximise form equivalence.

5.2.3. Adaptation of SPIN test

This adaptation of the SPIN test was presented in stereo with the speech coming either from a centre loudspeaker or from a phantom centre image. It would have been possible to record a new babble track in stereo and utilise this in the tests however it was considered that this could adversely affect results. The babble and speech used in the SPIN test has been phonetically balanced and rigorously tested in order to eradicate any keywords found to be consistently ‘difficult’ or ‘easy’ as a result of combinations of particular sections of babble and particular phonemes. For this reason it was decided that the benefit of utilising the existing babble from the SPIN CD outweighed the possible disadvantage of processing influencing the results. The authorised SPIN CD used for these listening tests was designed to be presented in mono and comes as a two channel stereo recording with multi-talker babble on one channel and speech on the other. In order to present the babble in a stereo sound field a DirectX plug-in was developed and applied to the mono babble track in order to generate a stereo signal with a wide source width.

5.2.4. Test Setup

Sentences ending in keywords were presented using both a phantom centre image and a real centre loudspeaker in a background of pseudo stereo multi-talker babble as indicated in figure 18.

The SPIN test recommendation is that it should be presented at a level 55dB SPL above the listener's *babble threshold* in a sound field; the babble threshold being the lowest level at which the listener can detect the presence of the pseudo-babble. The tests discussed here were carried out with participants with no hearing impairment and the appropriate babble threshold calculated from a typical non-hearing impaired audiogram. All tests were carried out at 68dB SPL, the level recommended in the manual accompanying the authorised SPIN CD for participants with unimpaired hearing. The test material was extracted from the authorised version of the revised SPIN test CD (Bilger R.C. Nuetzel J.M. Rabinowitz W.M. Rzeczkowski C, 1984). Audio data was imported into a laptop computer with a multi-channel audio interface and a multi-track audio program. Three buses were created within the program that individually controlled the levels of the babble and speech while routing them to appropriate outputs.

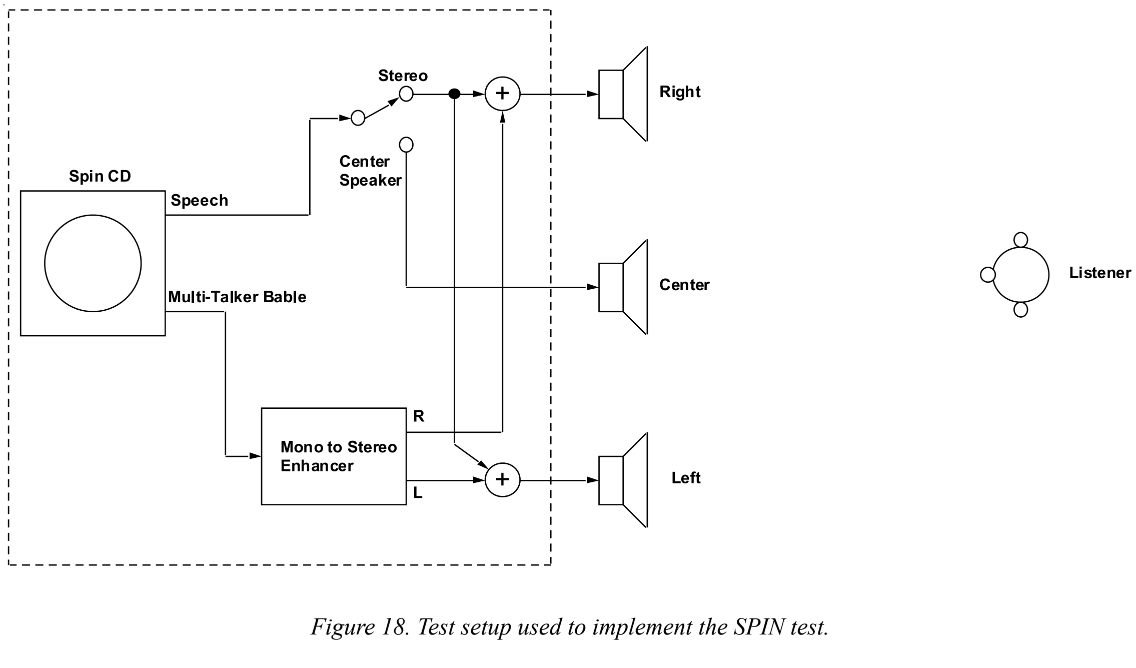


Figure 18. Test setup used to implement the SPIN test.

A representation of the test setup is shown in figure 18, the area of the diagram inside the dotted lines is implemented in software within a PC environment. The pseudo stereo multi-talker babble was generated from the authorised SPIN CD (Bilger R.C. Nuetzel J.M. Rabinowitz W.M. Rzeczkowski C, 1984) using a plugin implementing a pair of complementary comb filters. The signal flow for the filter is as shown in figure 19.

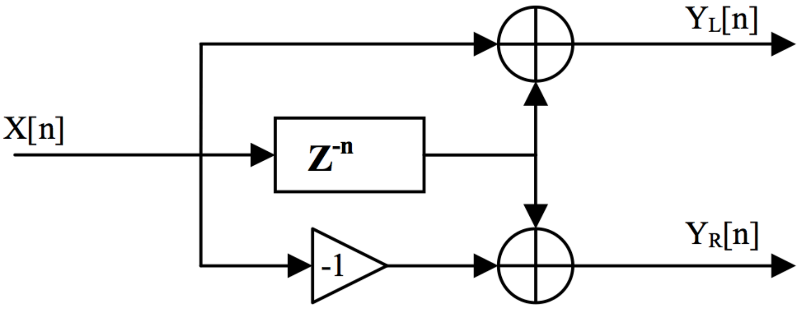


Figure 19. Signal flow of plug-in used to generate pseudo-talker babble

The frequency responses of the filters outputs are shown in figure 20. The filter outputs act as complimentary comb filters and can be added back together to get an approximation of the input.

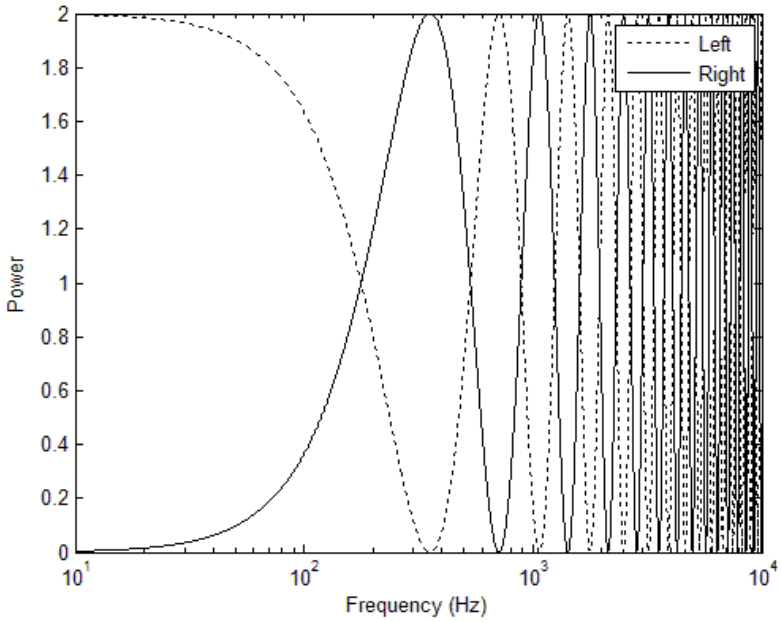


Figure 20. frequency response of mono to stereo enhancer plugin

5.2.5. Calibration

Each test condition was carefully calibrated so that each condition had not only the correct signal to babble ratio, but also the correct overall level. The tests took place in a listening room conforming to the ITU-R BS.1116-1 (ITU, 1997); three Genelec 1029 reference monitors were placed in loudspeaker positions according to ITU-R BS.775-1 (ITU, 2006) and connected to audio interface outputs. White noise was used to calibrate the relative level of each speaker using A-weighted Leq levels. The total level for each test condition was normalised so that each condition and each test was at the same overall SPL.

5.2.6. Test Procedure

A series of pilot tests were carried out prior to the main batch of tests in order to determine the appropriate signal to babble ratios required to obtain useful results. The test procedure was identical for both pilot test and main tests. The procedure was explained to each subject who was seated in the ideal listening position according to ITU-R BS.1116-1. Participants were played several examples of SPIN sentences in babble and asked to repeat the last word as a practice prior to commencing the test. The subject was then given an answer sheet to record the keywords recognised from SPIN sentences and the test was started. Each form, consisting of 50 sentences containing keywords, was played with no pauses, unless the subject raised his / her hand to indicate they were not ready to proceed to the next sentence. At this point, the test was paused until the subject was ready to continue.

5.3. Pilot Study

Previous work by Kalikow et al (1977) has shown ceiling effects where listeners get all keywords correct where the threshold is set too low, or all incorrect if it is set too high. The pilot study was designed to determine an appropriate signal to noise ratio that could be used in the main listening tests in order to avoid these ceiling effects. The pilot study tested a range of signal to babble ratios between +10dB and -10dB. Four participants with no hearing impairment participated in the pilot study, using six forms; central stereo image and central loudspeaker conditions were each tested at six signal to babble ratios. There were two test sessions for each subject. The combinations of subject, signal to babble ratio, audio process and form number were created for each test so that everything was tested a uniform number of times. The test order was designed to ensure that a person was only tested on each form once and that cognate form pairs were only ever tested across separate sessions. 12 tests were carried out in total for each condition as indicated in table 15.

Audio Process	No. Tests (per person)	No. Tests (Total)
2 Channel	3	12
3 Channel	3	12

Table 15. Tests carried out for each condition in pilot study

5.3.1. Pilot Study Results

The pilot study results are presented in figure 21 which shows the number of keywords correctly identified for the seven signal to babble ratios assessed in these pilot tests. A 2nd order polynomial trend line is used to indicate the trend of the data. The ceiling effects noted by Kalikow where participants get all of the keywords correct or all incorrect are clearly shown at the upper end of the trend line.

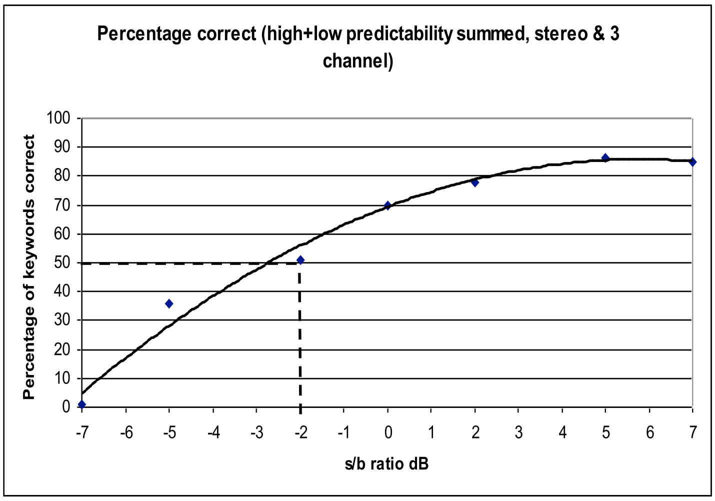


Figure 21. Percentage of correctly identified keywords for each signal to babble ratio assessed. Dashed lines indicate the -2dB ratio at which 50% of keywords are correctly identified and which was used for the main tests.

Figure 22 shows how the contextual information affects the listeners' ability to identify keywords. As can be seen the number of correct identifications for high predictability sentences is higher for corresponding signal to babble ratios.

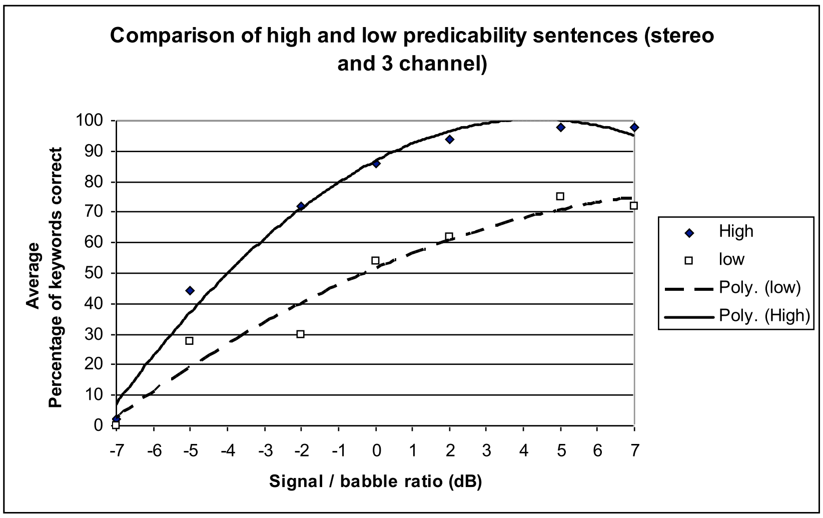


Figure 22. Number of identified keywords for high and low predictability sentences.

Note that the ratio of -2dB (indicated in figure 21) is on the slope of each trend line and so unlikely to result in participants getting all keywords correct or all incorrect.

5.3.2. Summary of Pilot Study Results

The results of the pilot study indicate that participants obtained an average of around 50 % total correct at a -2dB signal to babble ratio. The separated high/low predictability graph (figure 22) shows that a -2dB ratio should avoid ceiling effects for both high and low predictability sentences. This value was therefore used in the main batch of tests.

5.4. Main Tests

The main listening tests assessed speech intelligibility in multi-talker babble comparing directly the use of a virtual stereo image source and a real source. 20 normal hearing participants participated in the tests, which were carried out over two separate test sessions. The test was carried out at the signal to babble ratio of -2dB, which the pilot study had suggested was the ideal level to avoid ceiling effects for both high and low predictability sentences, and at an overall SPL of 67.9dB. Each subject was tested over two test sessions and each condition was tested twice; four different forms were used. Each of the four forms was tested 20 times, half from a central stereo image and half from a real source. The order of playback and audio process combination was changed for each subject. No cognate form pairs were used.

5.4.1. Main Test Results & Data Analysis

5.4.1.1. All Sentences (high and low predictability)

A two was repeated measures ANOVA was carried out to analyse the results.
Both number of channels (2 or 3) and keyword predictability (high or low) was shown to have an effect on the number of keywords recognised at $p<0.05$.

It is clear from the trend lines for high and low predictability sentences in figure 22 that the ability of participants to identify keywords is influenced, and improved, by contextual information and so it was expected that overall results considering the conditions together would reflect the wide variation in keywords correctly identified. Table 16 shows the average number of keywords correctly identified out of 25 along with their standard deviation for both two-channel and three-channel reproduction and for both high and low predictability sentences. Although useful in that the overall mean values indicated improved recognition for three-channel reproduction, the true difference in conditions is not represented owing to wide variation in the ability to recognise keywords between participants.

	2 Chnl	3 Chnl	Difference
mean	13.15	14.175	1.025
sd	5.480	5.791	

Table 16. Mean key words recognised for each condition

When all sentences are considered together, the 3-channel condition was seen to give a small but significant improvement in keyword recognition of 1.025 words at a confidence level of greater than 95%.

5.4.1.2. High and Low Predictability Sentences Considered Separately

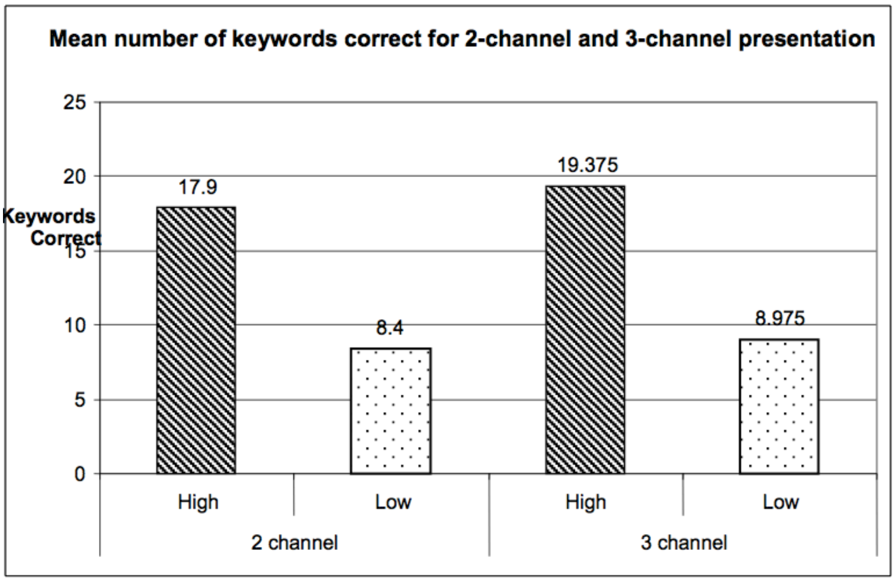


Figure 23. High & Low Predictability Sentences Considered Separately

Figure 23 shows the average number of words correctly identified out of 25 for high and low predictability sentences in both two-channel and three-channel listening conditions. The difference in keyword recognition between high and low predictability sentences indicated that there may have been benefit in considering these results separately. Table

16 shows the mean number of keywords recognised and standard deviation for high predictability sentences. The difference between the two channel and 3 channels conditions was analysed separately using a paired t-Test.

High Predictability Sentence Results

	2 Chnl	3 Chnl	Difference
mean	17.9	19.375	1.475
sd	2.942	2.579	

Table 17. Mean key words recognised for each condition and standard deviation (high predictability sentences only)

A paired t-Test indicated that although there is a relatively small improvement of 1.475 keywords correctly identified out of 25 (5.9% of all high predictability keywords) this result is statistically significant ($p<0.05$).

Table 18 shows the mean number of low predictability keywords identified correctly along with standard deviation. It is interesting that there appears to be considerably less improvement in keyword identification where the keyword is presented in a non-contextual setting as is the case for the low predictability sentences.

Low Predictability Sentence Results

	2 Chnl	3 Chnl	Difference
mean	8.4	8.975	0.575
sd	2.426	2.412	

Table 18 Mean key words recognised for each condition (low predictability sentences only)

When high predictability sentence results are considered separately from low predictability sentences it becomes clear that the confidence for high predictability sentences is considerably higher ($p \approx 0.014$) than for low predictability ones ($p \approx 0.102$). In each case, the mean number of keywords recognised is higher for the 3 channel listening condition. Significance in the paired t-test for high and low predictability sentences considered separately is probably diminished in both cases owing to the smaller number of tests in each group compared to the overall number and only the high predictability sentences give significant results.

5.5. Discussion

5.5.1. Measured Transfer Function

The results indicate that there is a difference between two-channel (phantom centre) and three-channel (with centre loudspeaker) presentation. The use of a real, as opposed to a phantom, source is shown to give a small but statistically significant intelligibility improvement in these test conditions, based on number of keywords recognised. One hypothesis for this is that acoustical crosstalk resulting from the stereo image, that has been noted by Holman (Holman.T., 1996) and others, has had a significantly detrimental effect and that this effect was not only on the effort required to understand words, but also on speech intelligibility.

Analysis of the test results indicates an improved average number of keywords recognised for the high predictability sentences compared to low predictability which was expected and indeed is utilised in Kalikow's original test. However the difference in the statistical significance between the two sets of results was not expected. Referring back to the pilot test results (figure 22) it is clear that the graph of keywords correct against speech to babble ratio shows a much steeper gradient for the high predictability sentences. If the effect of comb filtering caused by crosstalk has a similar influence on the results as decreasing the signal to babble ratio then a lower significance should perhaps have been expected.

The magnitude of the effect caused by crosstalk was then measured using a B&K dummy head as follows. White noise was played through a single loudspeaker located 2.27m away in front of the dummy head. The same signal was then also played through loudspeakers located at 30° either side of the axis, again 2.27m away. The white noise was played for 60 seconds under each condition and the frequency response measured at the left ear for both two channel and single channel conditions. From these measured frequency responses the mono to phantom mono transfer function was calculated. Figure 24 (A and B) shows the transfer function, or difference in frequency spectra, between single channel and two channel playback up to 10 kHz. This is shown using both linear (24A) and logarithmic (24B) frequency scales.

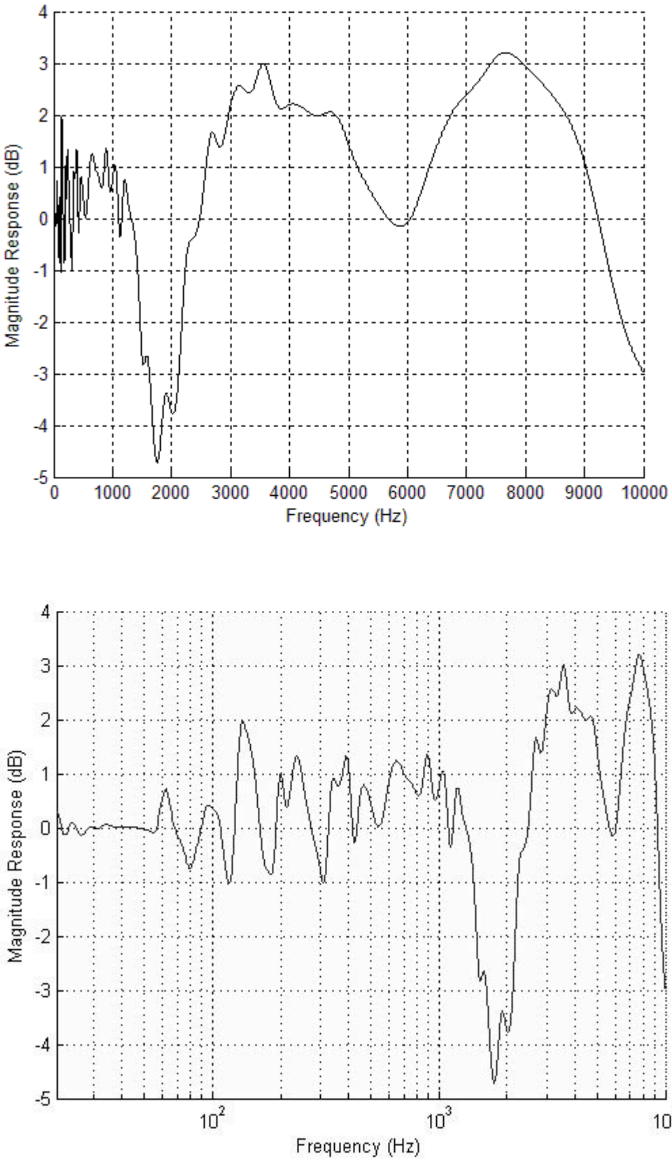


Figure 24 A and B Measured mono to phantom mono transfer function shown with linear (A) and logarithmic (B) scale; this represents the difference in frequency spectra caused by acoustical crosstalk. Note the dip at around 1.5 kHz to 3 kHz

The dips at around 2 kHz, 8 kHz and 11 kHz (not shown) indicate the presence of a cancellation and reinforcement pattern typical of comb filter effects. Early work by

Fant, described in 2004 (Fant, 2004), describes the frequency content of speech utilising a visual representation of what Fant called a 'speech banana' (Fant, 1959) to show the relative power of aspects of speech (figure 25).

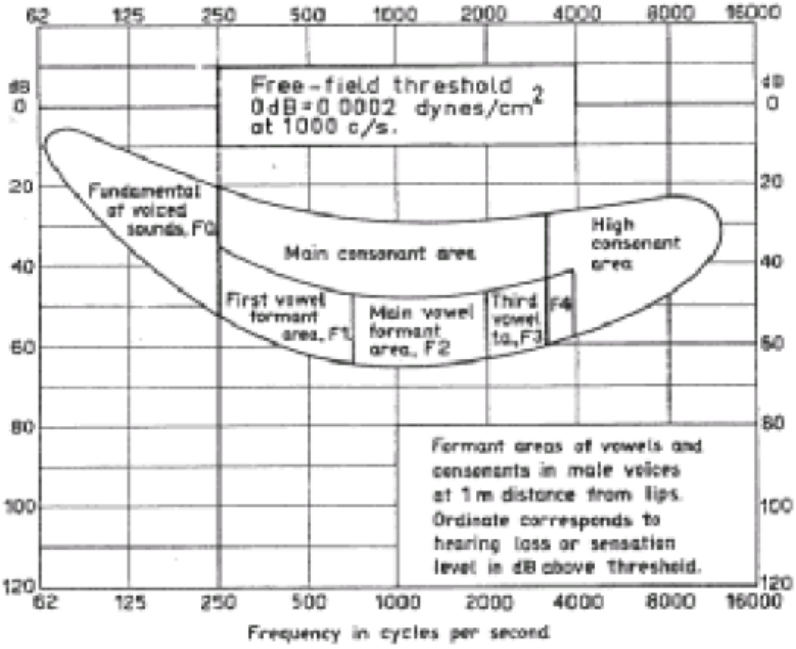


Figure 25. Fant's "Speech Banana"(Fant, 1959)

Another representation of this is shown in figure 39, which more clearly indicates typical sound intensity of various speech sounds.

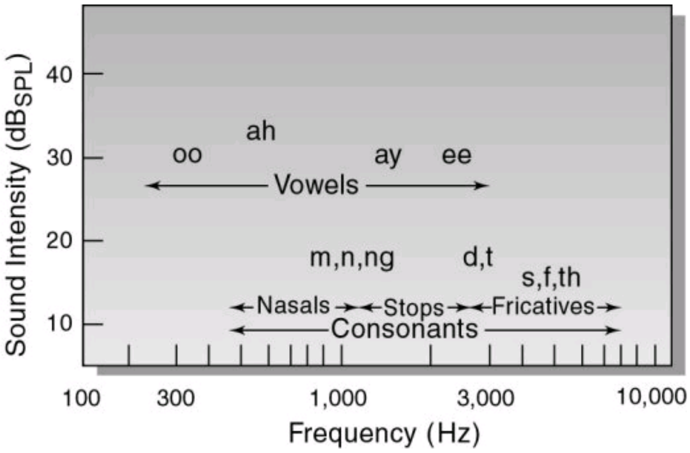


Figure 26. Sound Intensity of Speech Frequencies (Appalacian State University)

Figure 26 shows that, as is generally recognised, many of the frequencies most critical to speech reside in the region above 1 kHz. Superimposing this data onto the derived transfer function for a phantom mono source compared to a real mono source, as shown in figure 27, goes some way toward explaining the loss of intelligibility caused by the phantom centre image measured in the research.

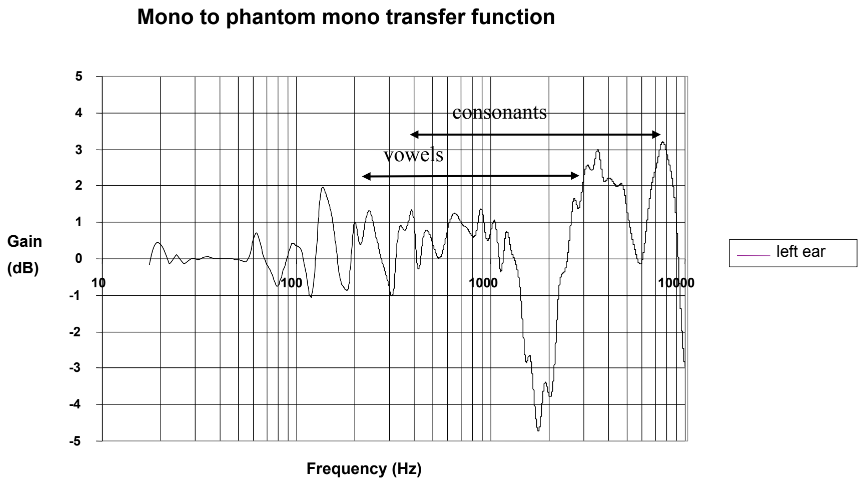


Figure 27. Speech Frequencies superimposed on mono / phantom mono transfer function plotted on a logarithmic frequency scale.

5.5.2. Independent Calculated Crosstalk Effect

Crosstalk is caused by the same signal being received from 2 sources, one arriving later than the other and it is the duration of this delay and the degree of head shading that determines the effect of this crosstalk. By calculating the difference in signal path for each loudspeaker signal to the left ear, as shown in figure 28, a measure of this crosstalk was calculated independent of room parameters and reproduction equipment. The additional distance that sound from the furthest loudspeaker must travel to the ear, and so the actual time delay incurred, was calculated based on the dummy head's position (2.27m from the loudspeakers) as follows.

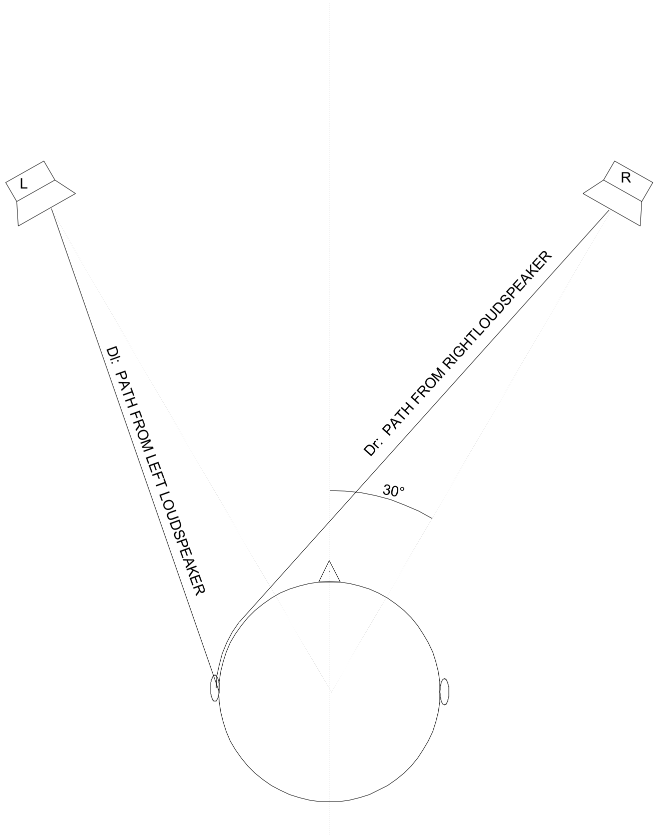

L
R
Dl: PATH FROM LEFT LOUDSPEAKER
Dr: PATH FROM RIGHTLOUDSPEAKER
30°

Figure 28. Path from left and right loudspeakers

5.5.2.1. Left/Right Loudspeaker Delay

The length of the signal path from each loudspeaker to the left ear was calculated using known distances from loudspeaker to subject, the radius of the dummy head and the known angle of incidence for each loudspeaker (as specified in ITU-R BS.775). A simple spherical head model was used and the calculations for this are presented here for each loudspeaker in order to calculate a theoretical value for the time delay and hence the theoretical crosstalk transfer function.

Left Loudspeaker Signal Path

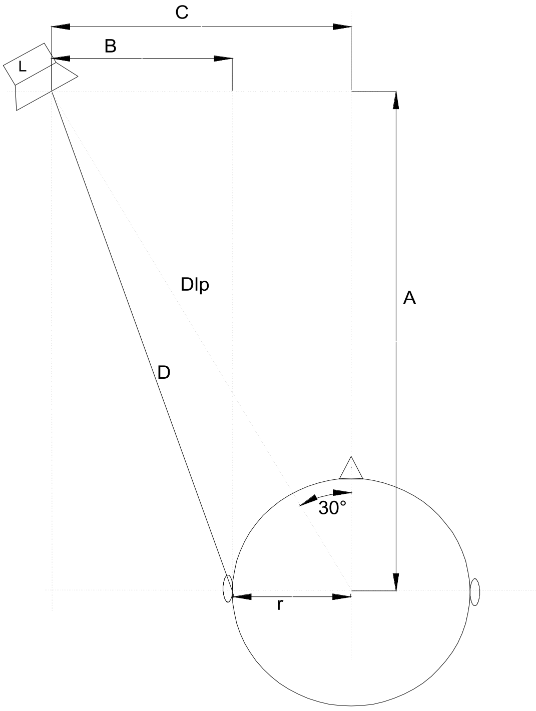


Figure 29. Signal path for left loudspeaker to left ear

The following calculations are with reference to figure 29.

r = 76mm (for B&K dummy head)

Dlp = 2270mm

Distance from left loudspeaker to left ear

$$C = Dlp \times \sin\theta = 2270 \times 0.5 = 1135$$

$$B = C - r = 1135 - 76 = 1059$$

$$A = \frac{C}{\tan\theta} = \frac{1135}{0.577} = 1966$$

$$D = \sqrt{B^2 + A^2} = \sqrt{1059^2 + 1966^2} = 2233mm$$

Right loudspeaker signal path

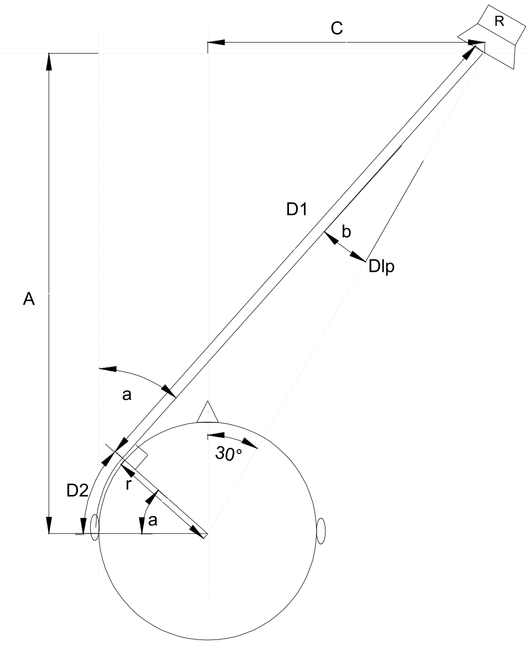


Figure 30. Distance from right loudspeaker to left ear

The following calculations are with reference to figure 30.

Total distance = D1+D2

$$D1 = \sqrt{Dlp^2 - r^2} = \sqrt{5147124} = 2269mm$$

$$\sin b = \frac{r}{Dlp} = \frac{76}{2270} = 0.0335$$
$$b = 1.919°$$

Angle of incidence $a = 30 + b = 31.919°$

Length of arc describing distance around head (using spherical head model) where r is in metres and θ is in radians. For θ in degrees

$D2 = \theta r$

$D2 = \theta \frac{\pi}{180} r = 0.042\text{m} = 42\text{mm}$

Total distance D = D1+D2 = 2269+42 = 2311mm

Left/Right Loudspeaker Path Difference

Left path = 2231mm

Right path = 2311mm

Difference = 80mm = 0.080m

This figure was used to calculate the time difference of the signal from each of the loudspeakers as follows:

Left path = 2231mm

Right path = 2311mm

Difference = 80mm = 0.080m

Speed of sound c = 344ms^{-1}

Delay owing to signal path $t = \frac{0.080}{344} = 0.000233s$

Independent Crosstalk Transfer Function

Inter-aural crosstalk creates a comb filtering effect that causes alternating dips and peaks in the transfer function. The first and major anomaly is a dip in the transfer function when the two acoustic signals arrive out of phase. This is calculated as follows:

First cancellation occurs at:

$\frac{1}{2t} = 2250\text{ Hz}$

Table 19 shows the calculated cancellation and reinforcement frequencies for the calculated value of delay up to 13.5 kHz. Note that head shadowing reduces the cancellation effects at the higher frequencies

Cancellation **1/2t, 3/2t, 5/2t**	**Reinforcement** **2/2t, 4/2t, 6/2t**
2250	5500
6750	9000
11250	13500

Table 19. Calculated Reinforcement and Cancellation Frequencies for calculated crosstalk

The transfer function for crosstalk caused by the calculated delay was plotted using Matlab and superimposed on the measured frequency response observed at the left ear (figure 31).

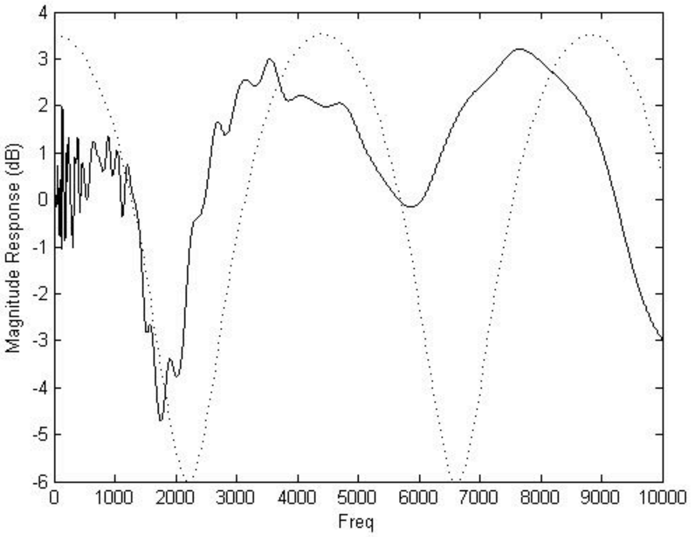


Figure 31. Measured Interaural crosstalk effect on transfer function using a B&K dummy head (solid line) compared to calculations based on a spherical head model (dashed line)

The graph clearly shows the effect of comb filtering resulting from crosstalk. The predicted first cancellation frequency of 2250Hz is seen to be approximately 500 Hz offset from the measured cancellation. This offset is likely to be a result of the delay being calculated using a simple spherical head model rather than a more sophisticated head related transfer function. From experimentation it was shown that the delay corresponding to the actual measured transfer function was approximately 0.00024s, rather than the 0.000233 calculated using the spherical head model.

5.5.2.2. Effect of Listening Position

It is important at this point to realise that the effect of this acoustical crosstalk is highly dependent on the position of the listener with respect to the loudspeaker positions. The calculation documented here is for the *ideal listening position* as defined in ITU-R BS775-1 Moving the listener to another position is certain to change which frequencies are attenuated and by what degree. In order to gain some insight into this variation further measurements were carried out using the dummy head.

Measurements were carried out in the same way as in previous measurements with white noise being played out of both loudspeakers. Loudspeakers were again at positions defined in ITU-R BS775-1 and measurements taken at the worst case listening positions defined in the same document as shown.

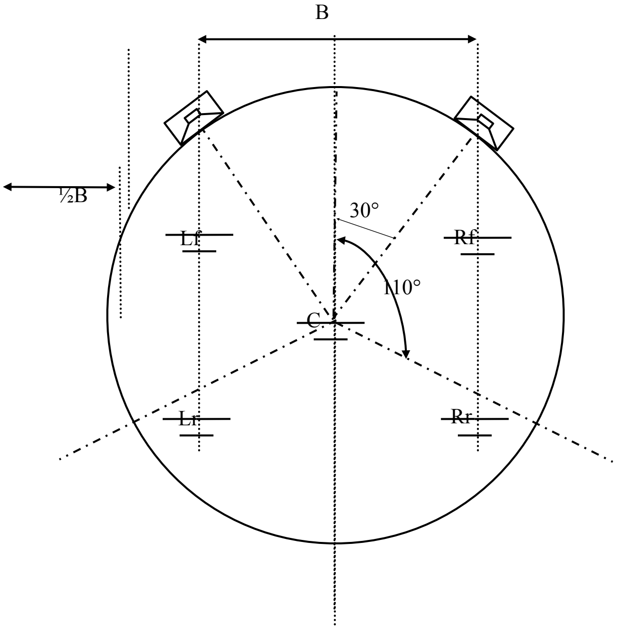


Figure 32. Ideal (C) and "worst case" (Lf, Rf, Rl, Rr) listening positions adapted from ITU-R BS 775-1 (ITU, 1994a).

Transfer Functions Measured with Dummy Head

The transfer functions illustrated here show the difference in amplitude measured for a real centre loudspeaker source and for a phantom centre source presented using two loudspeakers up to 16 kHz, for the left-front and left-rear positions illustrated in the previous figure (figure 32).

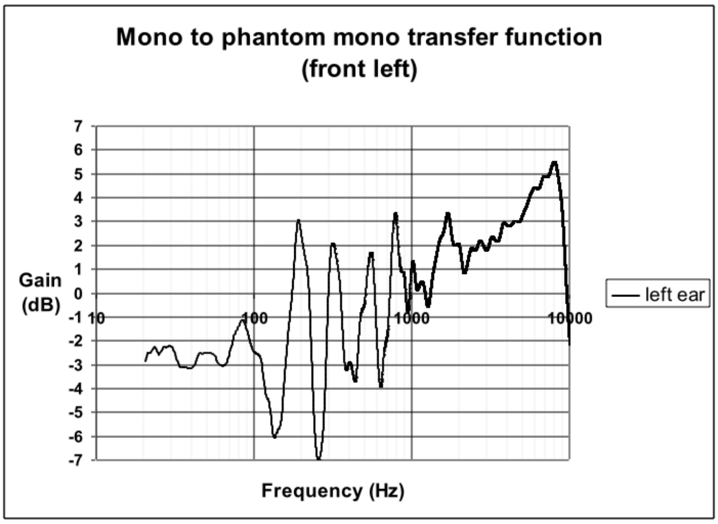


Figure 33. Front left position (Lf)

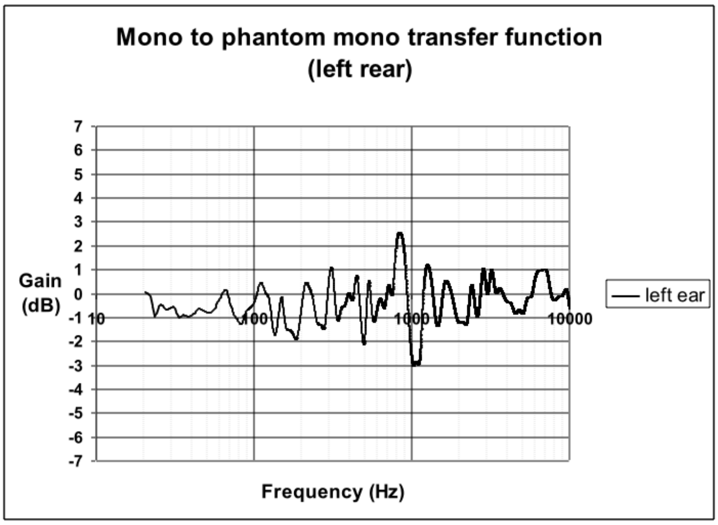


Figure 34 Rear left position (Lr)

The location of the dummy head for these measurements, at the worst case positions specified in ITU-R BS 775-1 alters the delay considerably between the same sound arriving at the ear from each loudspeaker and therefore has the effect of offsetting the entire comb filter response. This offset means that the cancellation frequencies are then moved beyond the range of speech and therefore will probably have little or no effect on intelligibility although the fluctuations prior to the initial cancellation frequency may have an impact. The variations indicated by these graphs probably correspond to the fluctuations prior to the first cancellation frequency in figure 31.

5.6. Conclusions

This research demonstrated a measurable improvement in intelligibility (as opposed to perceived clarity) as a result of using a real loudspeaker source to carry speech when compared with a central phantom stereo image. This improvement can be quantified in terms of the increase in number of keywords identified in the three channel condition using a central loudspeaker. By using a central loudspeaker to reproduce speech the total percentage of keywords correctly identified is improved on average by 4.1% across both contextual and non-contextually set keywords at a significance level of >95%. The reason the 2 channel system results in less intelligibility than the 3 channel system is consistent with an acoustical crosstalk effect causing cancellation and therefore a dip in the frequency response at frequencies important in speech.

This finding indicates that there are significant intelligibility gains associated with using a central loudspeaker such as is utilised in 5.1 surround sound broadcast systems although this benefit is only apparent for surround audio mixes where speech is panned centrally.

6. Principal Components Analysis

6.1. Introduction

One possibility for making TV sound more intelligible is to develop a method for separating out the speech element from the broadcast 5.1 audio stream. Various methods have been proposed for this, one of which, Principal Components Analysis (PCA), was proposed by Zielinski et al (Zielinski et al., 2005) in response to early work on the Clean Audio project (Shirley and Kendrick, 2004) documented in chapter 3 of this thesis. Principal component analysis is a statistical technique widely used to reduce the dimensionality of complex data sets for analysis or for data compression. It achieves reduced dimensionality by retaining lower order components which contribute most to its variance, on the assumption that these are the most important, and discarding higher order ones. The proposition presented is that the audio presented in a 5.1 multichannel mix is essentially a complex data set with many variables to which we wish to find a single component, in this instance, speech.

Zielinski argued that the Clean Audio solution was inappropriate as it assumed all speech would only be in the centre channel. The work presented an artificial situation with centre and one rear channel swapped so that speech emanated only from one of the rear surround channels in order to illustrate perceived shortcomings of the solution presented in chapter 3 of this thesis and proposed by the UKCAF. The processing proposed by Zielinski et al utilised a ‘speech filter’ to pre-process audio channels prior to carrying out PCA. This ‘weighted’ PCA analysis was carried out on a short section of audio, carefully chosen with speech present to allow the algorithm to work, and the resultant unmixing matrix was then applied to the entirety of the programme material with some success. This method allowed the PCA process to pick out the channel containing speech as the principal component of the material and act appropriately on other channels, by either muting or attenuating the non-speech channels.

In order to more fully ascertain any benefits to this approach and to assess PCA as fit for the purpose of producing ‘clean audio’ for broadcast an investigation was carried out where various synthesised audio mixes were processed using PCA. In this way a clear understanding of the technique, and of its limitations was gained.

Several 5.1 audio soundtracks mixed to reflect current broadcast and film making practice were then processed using PCA, using both weighted and unweighted techniques, in order to ascertain how the technique would work with ‘real world’ broadcast material.

In this chapter the principles of PCA are presented and the potential of PCA for providing a useful method of speech separation is evaluated in the context of broadcast 5.1 surround audio.

6.2. Principles of PCA

There are many examples of applications for which principal component analysis can be implemented. It has been used in a number of applications where it is necessary to reduce the dimensionality of large data sets, one example is in image compression where it can be used to remove attributes from the image that will not be noticed (Kurita and Otsu, 1993). It has also been used in image analysis, for example in face recognition applications it is a useful step in identifying key differences between images (Zhao et al., 1998).

In this section the process of identifying principal components from a data set is carried out to illustrate the steps required to implement PCA. The data set in this case was a two-channel stereo audio file. This example was used in order that data can be easily represented in two-dimensional graphs and so that the principles of PCA and those on which the solution proposed by Zielinski, Mayne et al (Zielinski et al., 2005) is based could be clearly understood. This work is then expanded to cover data sets of more dimensions including 5.1 audio.

The data in this example consists of a stereo audio file; two channels of 100 samples each. Test material was generated as mono audio files and imported into a multichannel audio software package where they were mixed as follows.

6.2.1. Example 1

440 Hz sine wave at 0dB panned 30% left,
White noise at 0dB panned 30% right.

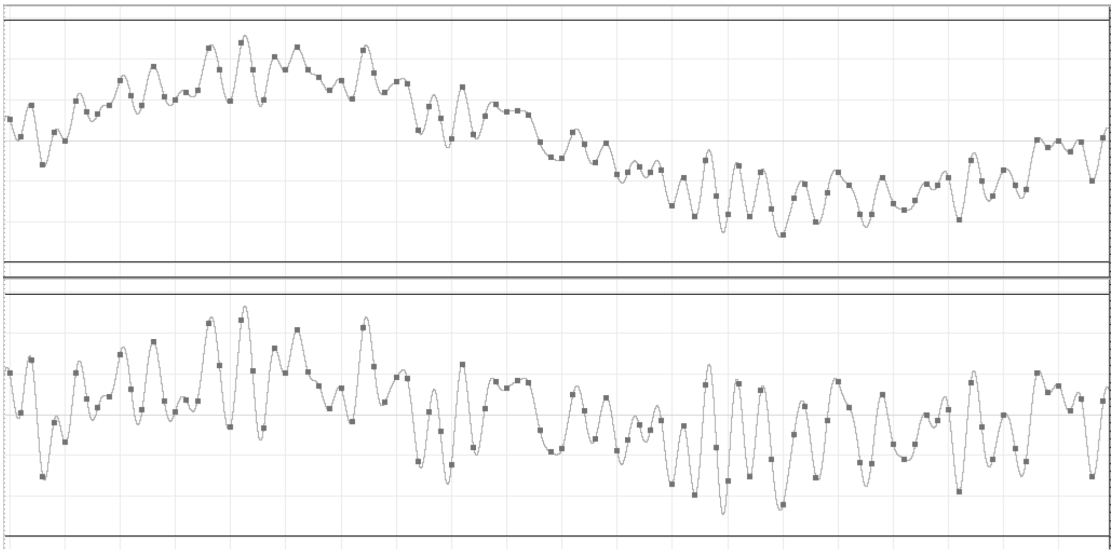


Figure 35. PCA Example 1: 440 Hz sine wave 30% left, white noise 30% right.

6.2.1.1. Subtract the Mean

The first step in PCA is to subtract the mean from each sample in the sample set; in the case of an audio file this is equivalent to removing any DC offset from the audio media. Although the data in this example consists of audio samples which are changing in time, the PCA process is unaware of this fact and simply looks at paired data sets, in this case of left-right sample values. Here the data is shown with left sample value plotted against right sample value (figure 36).

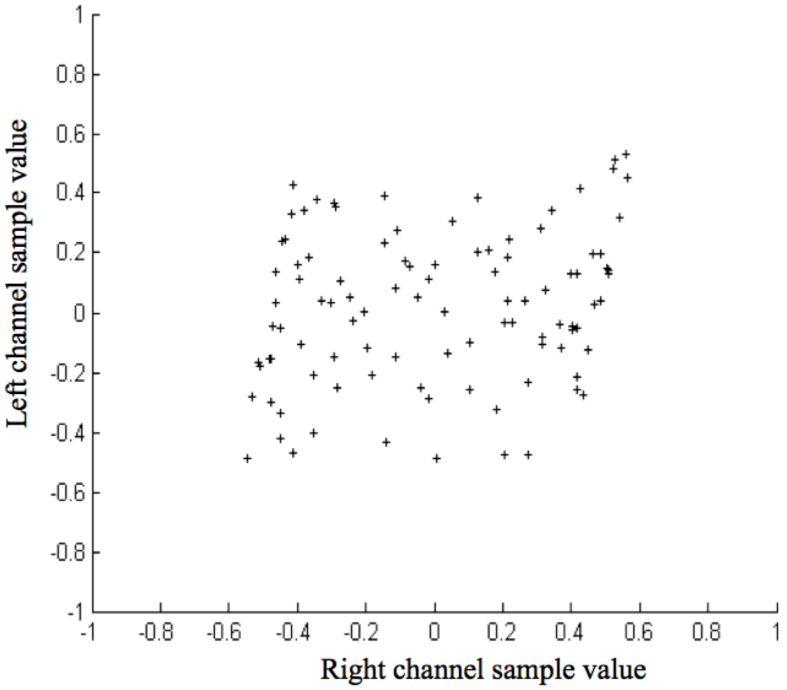


Figure 36. Sample values plotted left channel against right channel. A signal which was identical in both channels (panned centre in the case of an audio example) will show as a set of points following a straight line from point (-1,-1) to point (1,1)

It can be seen from the data when it is plotted in this way that there is some small degree of correlation between left and right which would indicate some commonality. For any centrally panned signal the plot would show a set of sample points following a straight line from point (-1,-1) to point (1,1) indicating perfect correlation between left and right channels.

6.2.1.2. Covariance Matrix

The covariance of two data sets is a measure of how the data sets vary with respect to each other. It is given by the following equation (Smith, 2002):

$$\operatorname{cov}(X,Y) = \frac{\sum_{i=1}^{n}(X_i - \overline{X})(Y_i - \overline{Y})}{(n-1)}$$

Covariance is always between two data sets so where there are more than two sets of data and therefore more than a single corresponding covariance value it is necessary to use a covariance matrix to express the covariance between each variable simultaneously. In the case of this example there are only two data sets however PCA can be used for multi-dimensional data and is demonstrated for 5 data sets (actually 5 audio channels) later in this chapter. The covariance matrix is calculated as follows (Smith, 2002):

$$C^{n*n} = (C_{i,j}, C_{i,j} = \mathrm{cov}(D_i D_j))$$

Where C^{n*n} is an n*n matrix and D_x is the *x*th data dimension .
For a 3 dimensional data set this gives a matrix as follows:

$$C = \begin{pmatrix} \mathrm{cov}(x,x) & \mathrm{cov}(x,y) & \mathrm{cov}(x,z) \\ \mathrm{cov}(y,x) & \mathrm{cov}(y,y) & \mathrm{cov}(y,z) \\ \mathrm{cov}(z,x) & \mathrm{cov}(z,y) & \mathrm{cov}(z,z) \end{pmatrix}$$

Or for a 2 dimensional data set:

$$C = \begin{pmatrix} \mathrm{cov}(x,x) & \mathrm{cov}(x,y) \\ \mathrm{cov}(y,x) & \mathrm{cov}(y,y) \end{pmatrix}$$

For this two channel audio example, a two-dimensional data set, the covariance matrix was calculated to be as follows:

$$C = \begin{pmatrix} 0.1274 & 0.0202 \\ 0.0202 & 0.0646 \end{pmatrix}$$

6.2.1.3. Eigenvectors and Eigenvalues

The eigenvectors and associated eigenvalues of the covariance matrix are now calculated (in this case using Matlab) and are as follows.

$$\text{Eigenvectors: } \begin{pmatrix} 0.2914 & -0.9566 \\ -0.9566 & -0.2914 \end{pmatrix}$$

$$\text{Eigenvalues: } \begin{pmatrix} 0.0585 & 0 \\ 0 & 0.1309 \end{pmatrix}$$

At this point it is useful to superimpose the eigenvectors for this data onto the data plot shown earlier.

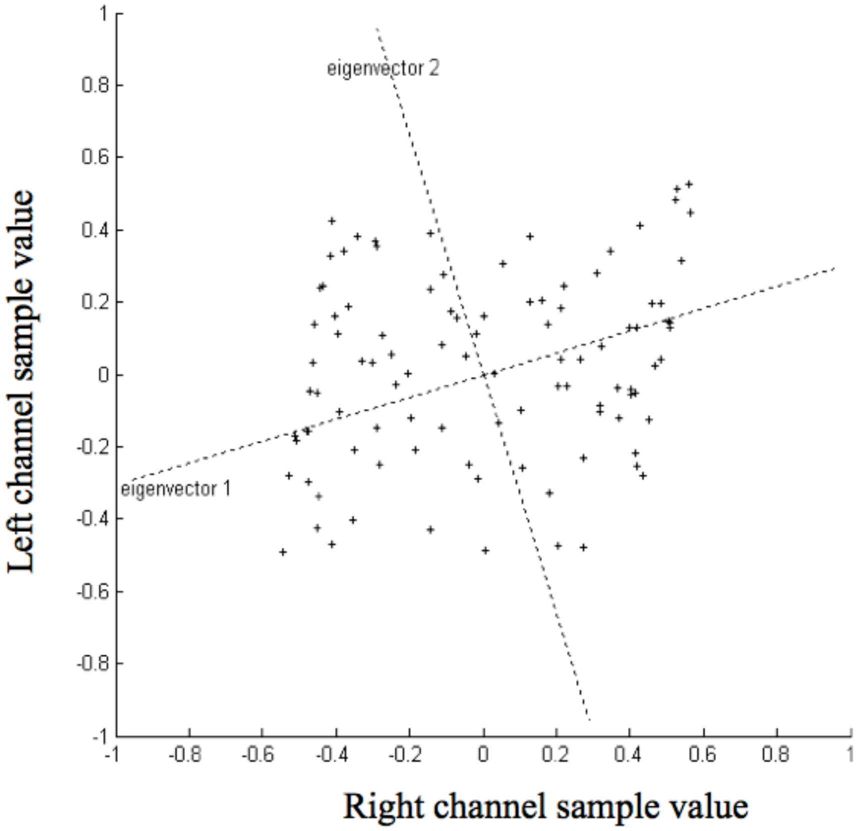


Figure 37. Data plot of original data with eigenvectors superimposed, note these are always orthogonal to each other.

It can be seen in figure 37 that eigenvector 1 and eigenvector 2 define the signal using two components. Eigenvector 1 looks like a 'line of best fit' (Smith, 2002) for the data whereas eigenvector 2 indicates how far the data deviates from this line. Between them the two eigenvectors define the combined data sets in two dimensions.

The eigenvalues calculated earlier are associated with these eigenvectors and are a measure of the importance of their paired eigenvector, or component. The component represented by the eigenvector that has the highest associated eigenvalue is known as the *principal component* and other components can be ordered in term of decreasing eigenvalue to indicate their relative significance to the combined data sets. When using PCA in image and other compression techniques some of these lower order components are discarded thus reducing the amount of data.

In this example the principal component turns out to be represented by eigenvector 1, with an eigenvalue of 0.1309 compared to 0.0585 for eigenvector 2.

Once these components are known we can form a *feature vector* that consists of a matrix with the eigenvectors that we intend to retain in their order of importance.

$$\text{Feature Vector} = (\text{eig}_1, \text{eig}_2, \text{eig}_3, \text{eig}_4 \ldots . \text{eig}_n) \text{ (Smith, 2002)}$$

If we keep both of our components from the example we have a feature vector of:

$$\text{Feature Vector} = \begin{pmatrix} -0.9566 & 0.2914 \\ -0.2914 & -0.9566 \end{pmatrix} \text{ (both components)}$$

If, on the other hand, we are only interested in a single component the feature vector is made up of only that eigenvector, in this case the principal component (that eigenvector which has the highest eigenvalue):

$$\text{Feature Vector} = \begin{pmatrix} -0.9566 \\ -0.2914 \end{pmatrix} \text{ (principal component only)}$$

At this point the new final data set can be derived with one or more of the lower order components left out by multiplying the transposed Feature vector with the transposed normalised data.

Final Data = (Feature Vector)^T x (Normalised Data) ^{T}

Figure 38 shows the derived data set with the feature vector above utilised and only the principal component remaining.

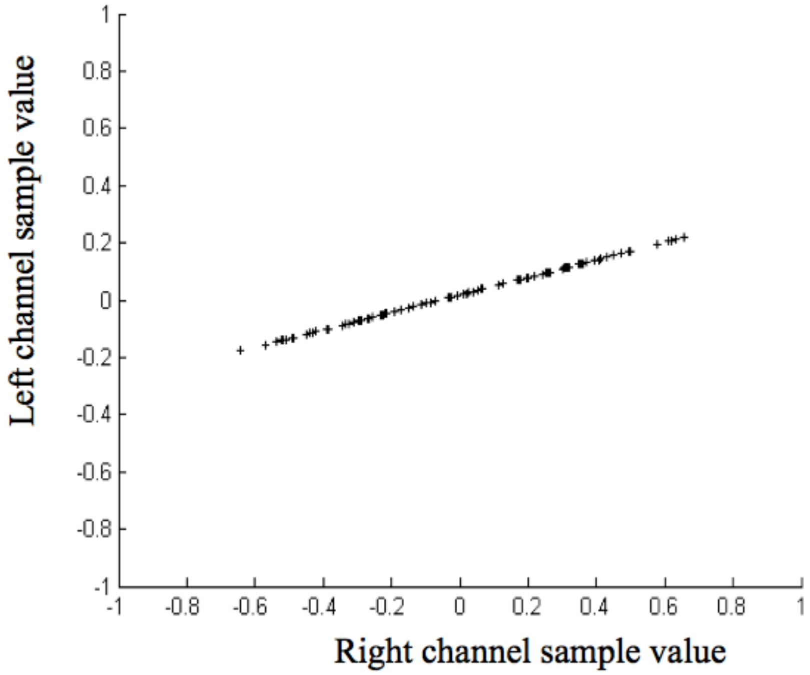


Figure 38. Derived data set using only the principal component in the feature vector.

The data set could equally well be derived using only the non-principal component as shown here in figure 39.

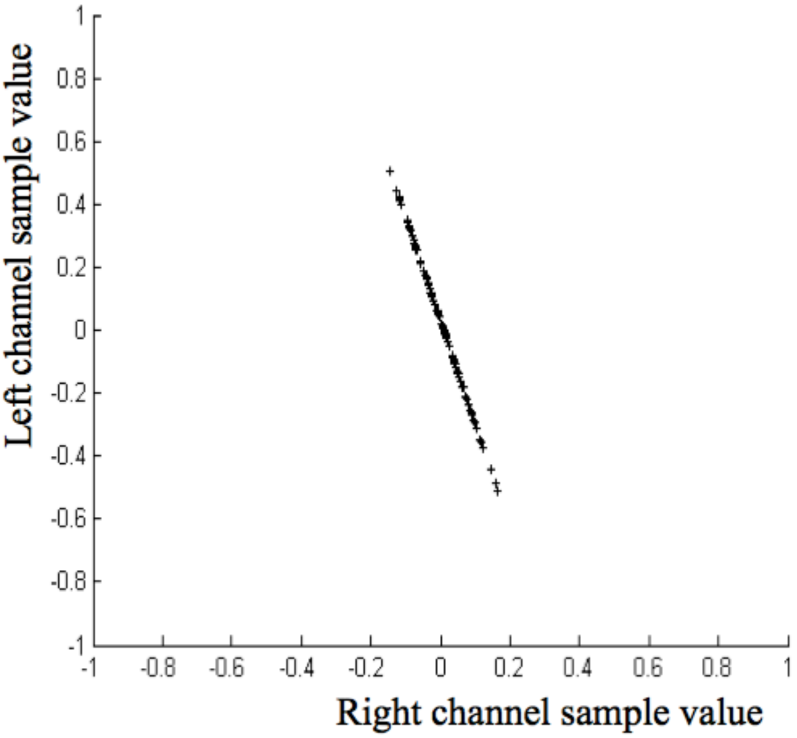


Figure 39. Derived data set using only the non-principal component in the feature vector.

Using both eigenvectors to derive the new, reconstituted data set gives the following data plot. The data is now expressed in terms of the principal components rather than in terms of x and y coordinates and is a rotated version of the original data shown in figure 37. In effect the eigenvectors are now the axes.

At this stage we can reconstruct the data using either, all, or a limited number of the eigenvectors we have defined. Figure 40 shows a new derived data set using both components with data now expressed in terms of eigenvectors.

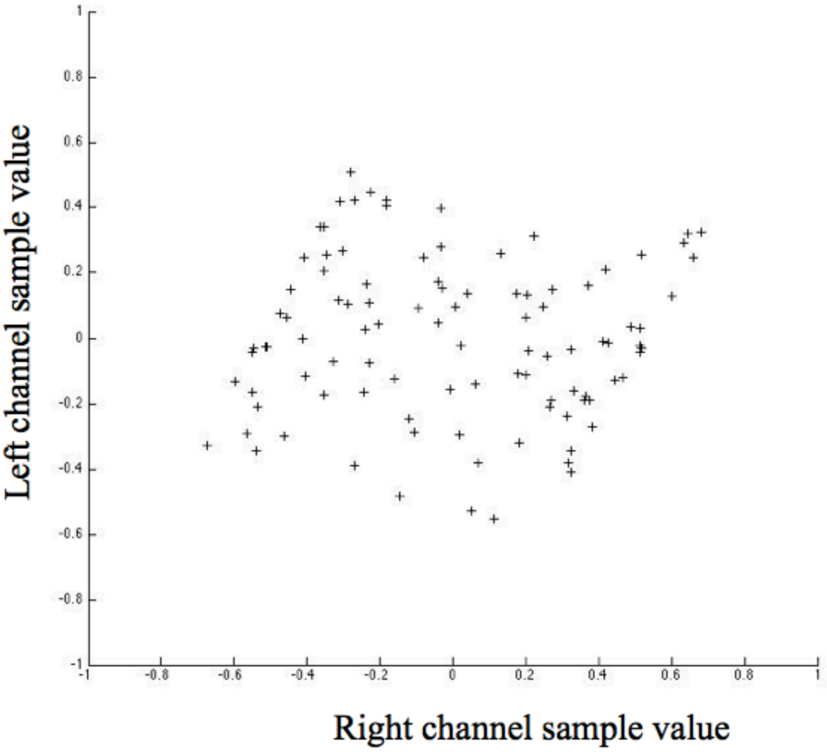


Figure 40. new derived data set using both eigenvectors to construct the feature vector. The reconstituted data is a rotated version of the original data with the data now expressed in terms of its eigenvectors rather than original x, y values. The data in this case was rendered as a two channel audio file containing both components.

6.3. Further Examples of Source Separation using PCA

Code was developed in Matlab to carry out PCA processing on audio files in order to understand more fully the potential for PCA to be used as a process for separating audio signals in a mixture. Several test separations were attempted and are documented here.

6.3.1. Example 2:

Source file: two channel stereo with white noise at -10dB, panned 40% left, Square wave at 0dB panned 60% right.

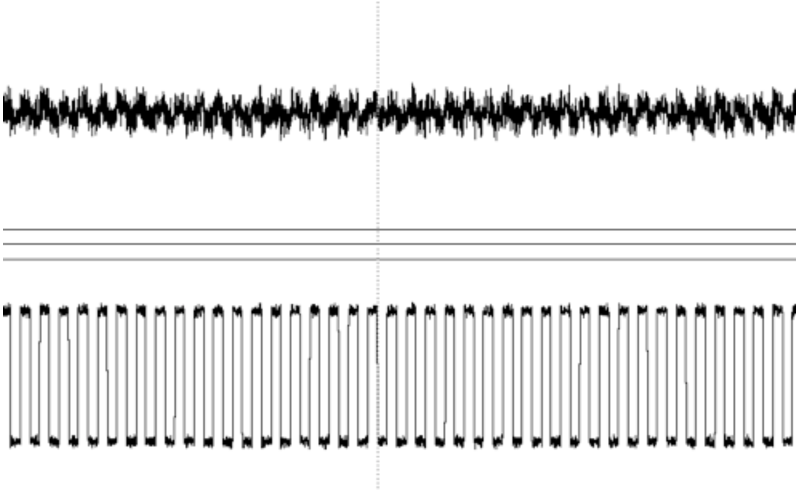


Figure 41. Input file for PCA processing

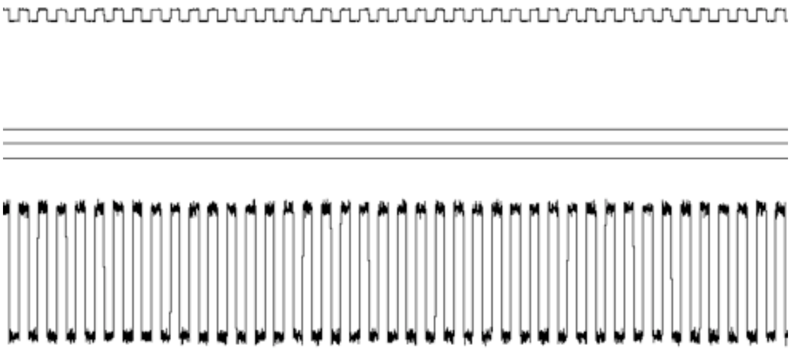


Figure 42. Output File after PCA processing with only principal component remaining.

The signal mixture can be clearly seen in the input file (figure 41) with particularly the left channel (upper channel) clearly dominated by noise. After carrying out PCA using the matlab file *PCA_single _vector_recon_mix.m* (Appendix I) and selecting only the principal component, the output file (figure 42) has noise largely removed from the left channel. It is still present in the right channel although subjectively the stereo output has much less noise in the signal. It is interesting to see how effective the process can be for removing noise from an audio signal, a close analogue to removing background noise from speech components in the broadcast domain.

6.3.2. Example 3:

Again the processing here has been carried out on two channel stereo audio files in order that the process can be easily visualised and the potential for PCA filtering evaluated more clearly.

Sin and Noise

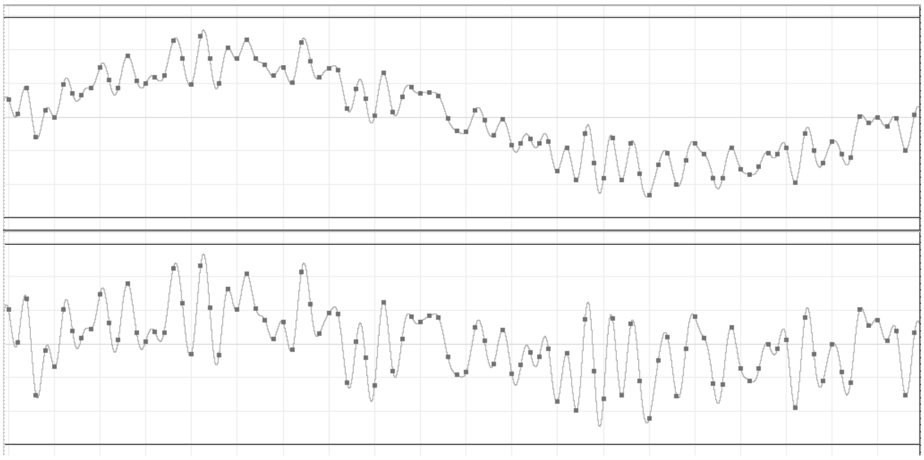


Figure 43. Input2.wav with 440 Hz sine panned 30% left, white noise panned 30% right

Original data was then restored using only a single eigenvector as follows:

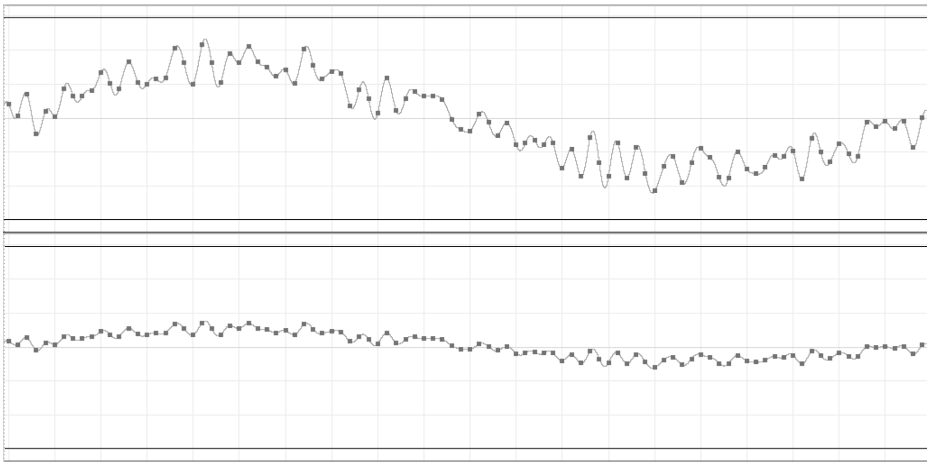


Figure 44. Output2wav - Principal Component Only

Again the PCA algorithm has managed a reasonable degree of separation, the principal component (shown in figure 44) contains mainly the sine wave component of the mix, panned to the left, the non-principal component has no visible sine wave component remaining (figure 45). Given that this is carried out manually with no 'weighting' of the algorithm the PCA method for speech separation looked promising on this evidence.

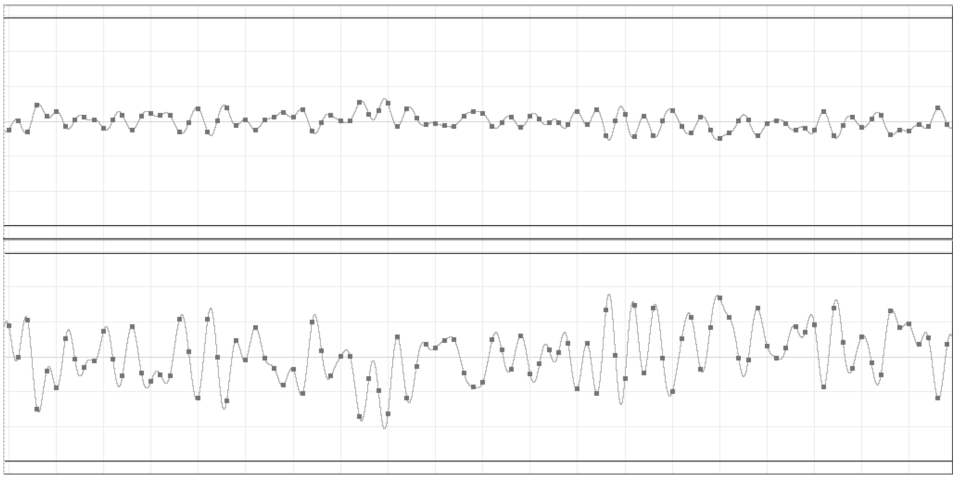


Figure 45. Output for non-principal component only

The Matlab code was then automated and adapted to allow processing of multichannel wav files in order that the process could be evaluated with some real-world examples of 5.1 surround audio media. Code was developed that read multi-channel audio files into Matlab for this purpose. Unfortunately the somewhat non-standard nature of the multichannel wav file format meant that none of the multichannel wav files capable of being output from Adobe Audition could be read using Matlab's *wavex* function. Because of this issue the code was adapted to work on multiple mono wav files that were extracted from DVD media for processing.

Each mono file was opened and input into the next matrix column in Matlab where similar code as that already developed could be used to carry out PCA. After processing the individual columns were read out into separate mono wav files using the *wavwrite* function utilising column addresses rather than matrix names in Matlab. The code used for this can be found in Appendix I.

6.4. *PCA Carried out on 5.1 Surround Sound*

6.4.1. Multichannel from 5 mono wav files - speech in centre

This example utilised a short audio excerpt from the film *The Matrix* (Wachowski and Wachowski, 1999). PCA was carried out on the 5 channels of the 5.1 channel soundtrack, only the low frequency effect channel being omitted. All but the principal component of the resulting matrix was deleted and the five individual wav files then reconstructed.

The speech in this clip is a female voice, the background is loud music in a nightclub scene. The mix in this case is typical of film material in that the speech is solely present in the centre channel, no music or atmosphere track is present in centre channel at all except for a very small amount of foley generated effect (footsteps, rustle of clothes), left, right left surround and right surround channels all contain music.

This initial experiment with 5.1 material tests the hypothesis that the principal component of the section of AV material used will be speech and also indicates how an unweighted PCA algorithm responds to some 'typical' film soundtrack material. The Matlab code used can be found in appendix I, *PCA_5_mono_wavs.m*.

The Audition screenshot below (figure 46) shows both the original (to the left) and the processed 5 channel mix (on the right).

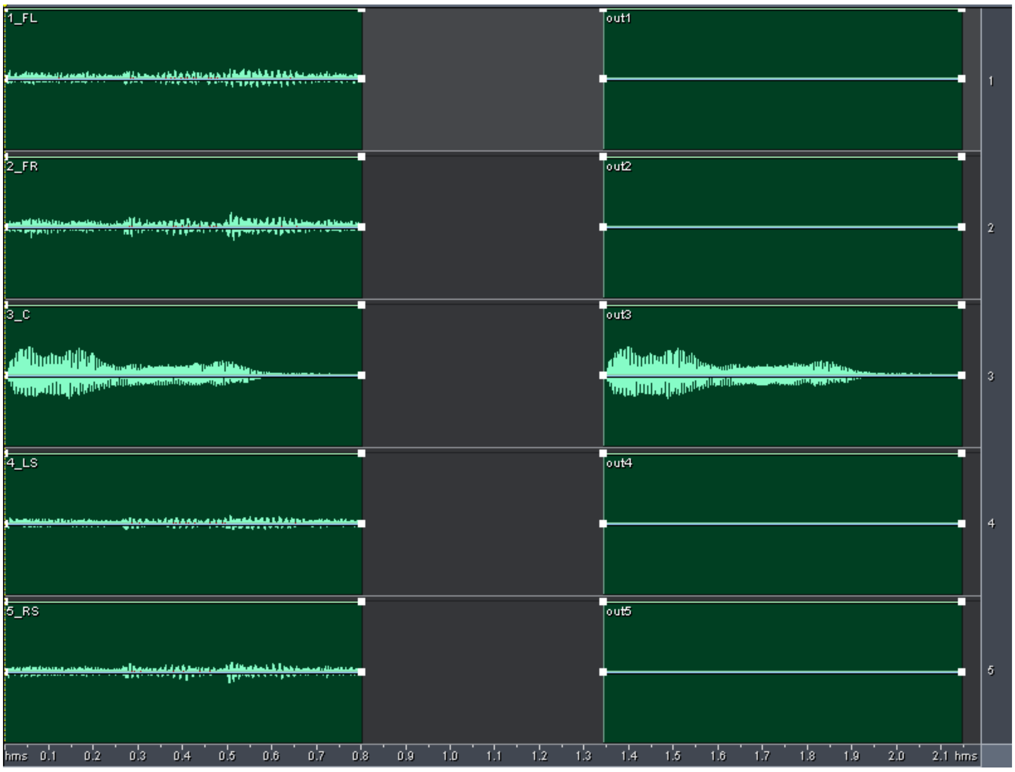


Figure 46. screenshot of both input and output of the PCA process carried out on 5 channel film audio

Looking at the input and output waves in figure 46 it can be seen that the PCA process has effectively muted all but the centre channel so removing all music and background atmosphere and effects from the clip and leaving only the speech. On the face of it this would seem to have accomplished what is required; it has removed background noise detrimental to speech. Also it would be quite feasible to alter the code in such a way that it merely attenuated the non-principal components by a factor instead of muting altogether, for example by the 6dB shown to be effective earlier in this thesis.

Given that the eigenvalues associated with each eigenvector indicate the order in which the components are critical to the overall signal it is useful to view a scree plot of these eigenvalues which indicates the relative importance of each component (figure 47).

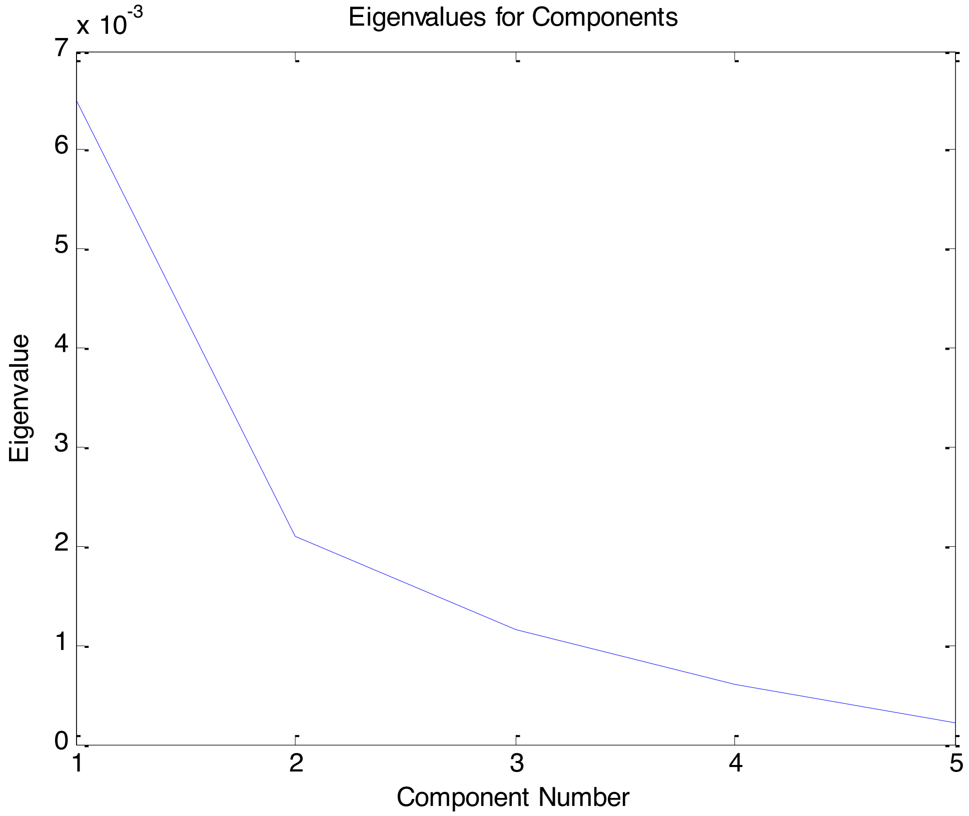


Figure 47. Scree plot of eigenvalues indicating relative importance of each component

Tables 20 and 21 show the eigenvectors and eigenvalues derived from the 5 channel audio content.

0.2069	-0.1748	0.9614	0.0488	-0.0040
-0.1925	0.6077	0.1899	-0.7467	0.0072
0.0002	-0.0023	0.0032	0.0085	1.0000
-0.6402	-0.6711	0.0346	-0.3723	0.0016
0.7143	-0.3870	-0.1962	-0.5490	0.0043

Table 20. Matrix of eigenvectors

0.0002	0	0	0	0
0	0.0006	0	0	0
0	0	0.0012	0	0
0	0	0	0.0021	0
0	0	0	0	0.0065

Table 21. Matrix of eigenvalues

6.4.2. Speech-biased PCA Algorithm: When Speech is Not Only in Centre Channel

Some 5.1 mixes do not adhere precisely to the Dolby recommendation that speech should be predominantly in the centre channel only and any solution which attempts to separate speech from competing background sound must therefore be assessed with regards to these other, less standard, mixes. Examples include some movie material analyzed during this research which had speech panned either between left and centre or between right and centre depending on who was talking e.g. (Altman, 2001)) and a considerable amount of BBC entertainment and also BBC and BSkyB sport programming that has speech spread across the front three channels of the 5.1 mix.

In the following experiments a weighted PCA algorithm was used in the same way as described by Zielinski (2005), the intention being to determine whether this solution is a credible solution for other real world broadcast material instead of only for material that *only* has speech, and speech only in the centre channel throughout the programme material and the artificial scenario constructed in Zielinski's work where a pair of channels were swapped over. The input for these experiments was in two parts; a reference 5.1 audio section taken from the media to be processed, carefully selected to contain speech, had a band pass filter applied to it with the same characteristics as described in the aforementioned paper, discarding those frequencies that contained no speech components. PCA was carried out on the filtered reference audio and the principal component identified from the mix with this positive bias towards 'speech frequencies'. The second complete unfiltered input had PCA carried out on it and all components except the principal component *based on eigenvectors and eigenvalues that had already been determined by the filtered reference* are attenuated or deleted. The PCA process is therefore biased, or weighted, as shown in figure 48. Matlab code for this processing can be found in appendix I,

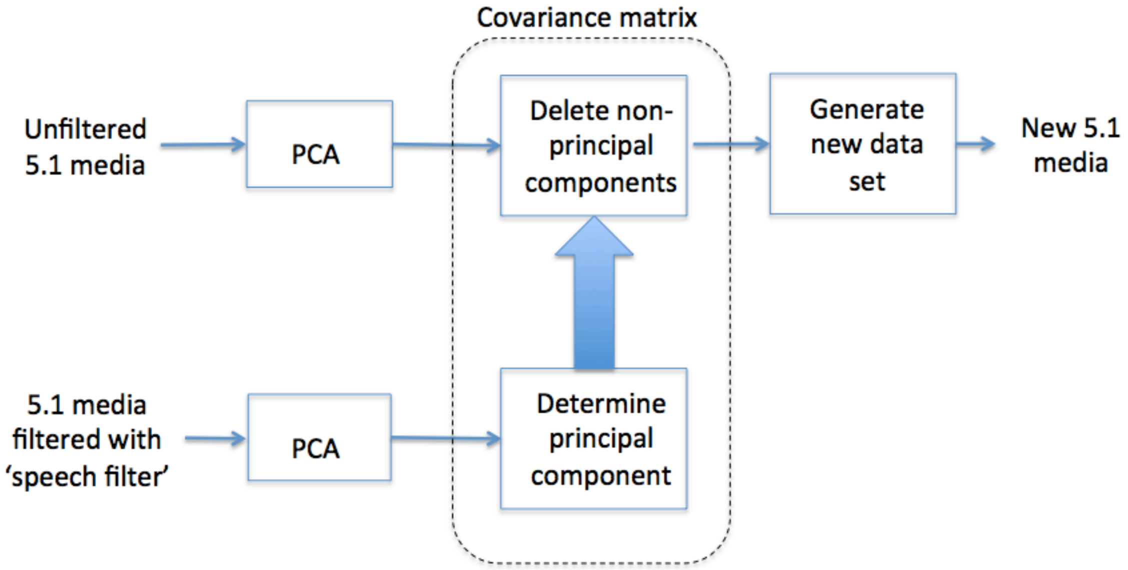

Covariance matrix
Unfiltered 5.1 media
PCA
Delete non-principal components
Generate new data set
New 5.1 media
5.1 media filtered with 'speech filter'
PCA
Determine principal component

Figure 48. System diagram showing a 'speech filtered' covariance matrix determining attenuation or deletion of components in unfiltered path

For the purpose of ascertaining the effectiveness of the process the attenuation factor has been set to delete all but the non-principal components however setting a different value (such as -6dB used in the Clean Audio Project) is a trivial matter of altering a single variable value.

The following two scenarios have been mixed to reflect examples found in broadcast practice where Dolby recommendations of speech to be in centre channel have not been strictly observed.

6.4.3. Speech-biased PCA Algorithm: Speech across left, centre and right channels

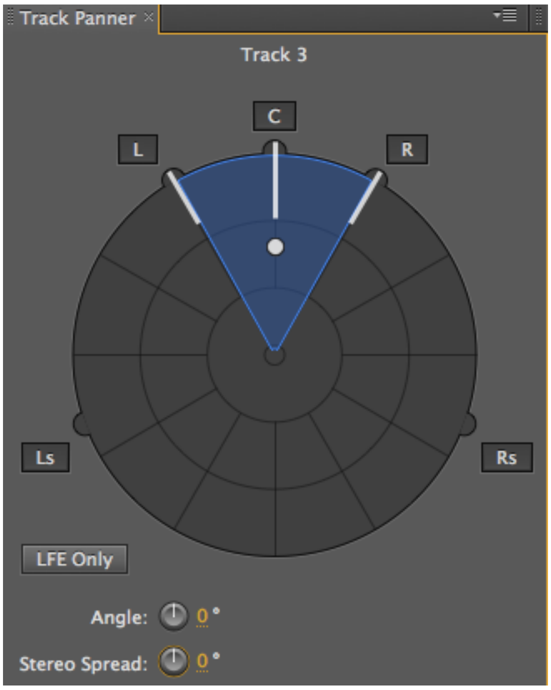


Figure 49. Panning reassignment for centre channel across left, centre and right of the 5.1 mix, other channels are mixed as usual

The input was mixed to reflect common BBC Entertainment and Sport, and Sky Sport practice of panning speech across the front three channels of the the 5.1 mix (shown in figure 49). The processed output waveforms had subjectively slightly less background music than the input but this was much less noticeable than when speech and music were in discrete channels in a 'standard' 5.1 mix. Some speech (panned to centre, left and right as in some BBC TV entertainment and sport programming) was also present in the rear surround channels of the output although much of the music had disappeared from the rear channels.

A screenshot of the input and output waveforms for all 5 channels can be seen in figure 50. Channel order is as follows: 1- left, 2 - right, 3 - centre, 4 - left surround, 5 - right surround.

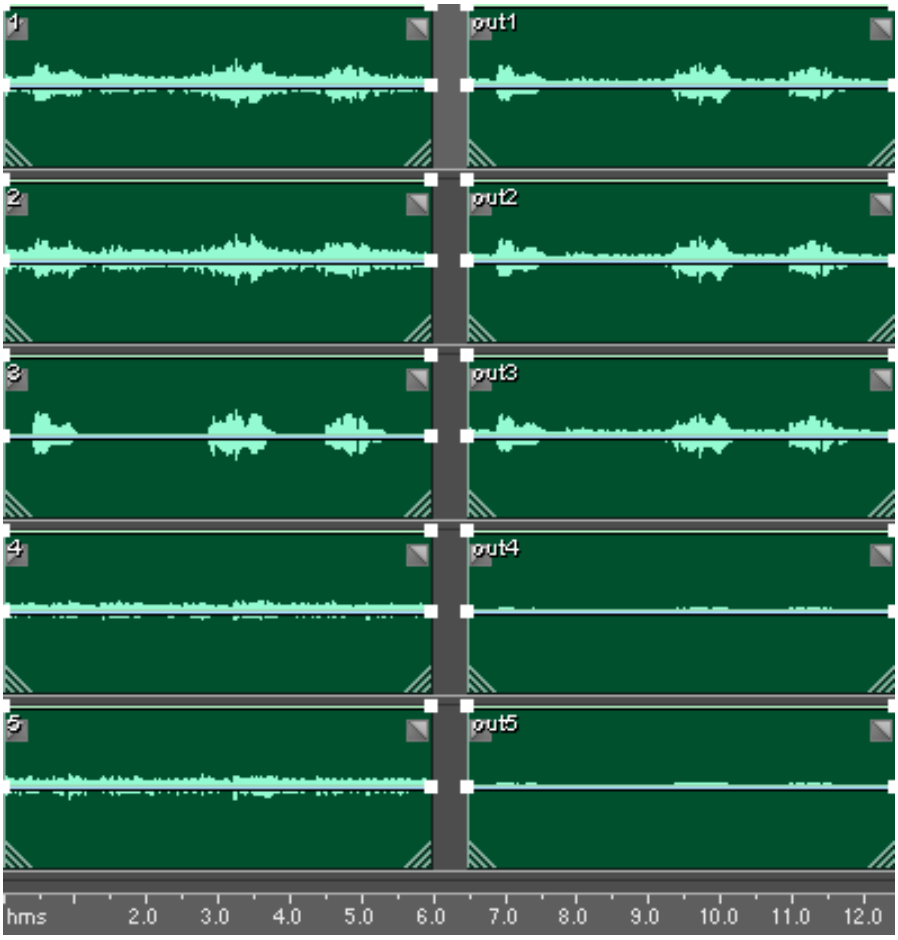


Figure 50. input and output waveforms shown in screenshot showing some attenuation of non-speech elements in left and right but added music in centre.

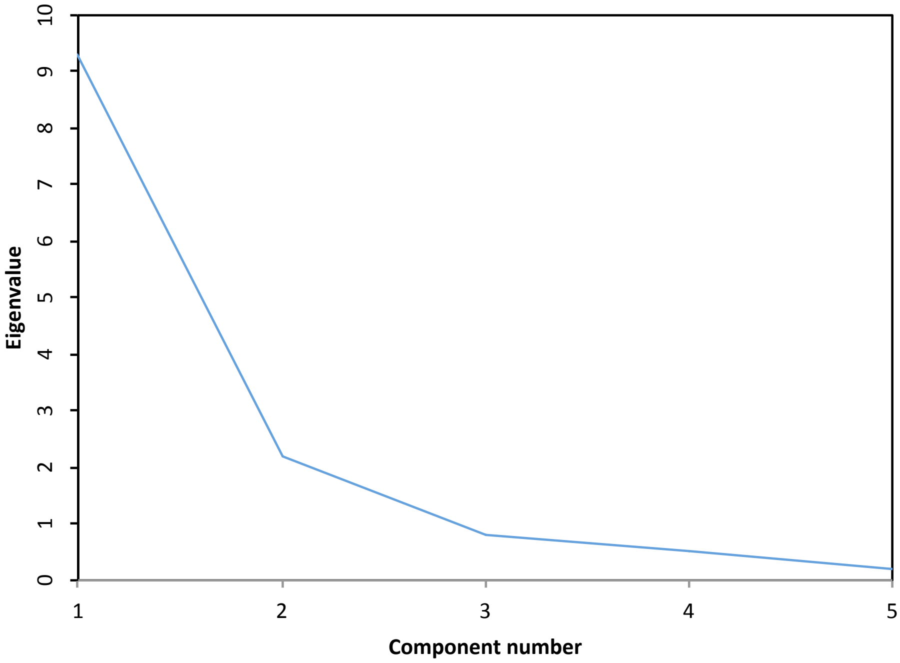


Figure 51. Scree plot of eigenvalues

6.4.3.1. Matrix of Eigenvalues

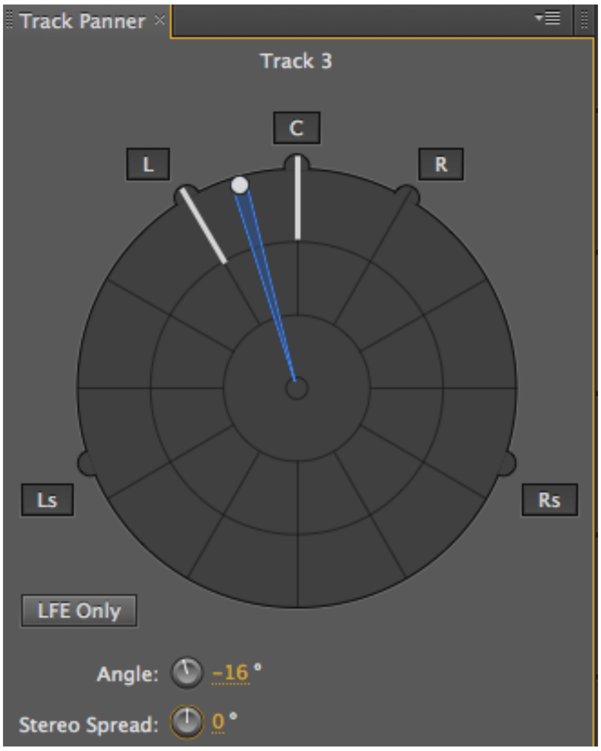
0.0002	0	0	0	0
0	0.0005	0	0	0
0	0	0.0008	0	0
0	0	0	0.0022	0
0	0	0	0	0.0093

Table 22. Matrix of Eigenvalues

Looking at the eigenvalues derived in the case where speech is spread across left, centre and right channels, the eigenvalue of the principal component was 0.0093 compared to an eigenvalue of 0.0065 for the principal component where speech is present in centre channel only. This indicates that the principal component was much more clearly defined in this instance. The output waveforms had subjectively less background music than the input waveforms overall but this was much less noticeable than when speech and music were in discrete channels in a ‘standard’ 5.1 mix. Most reduction of background (music in this case) has taken place by removal from the surround channels which had no speech element present. However some speech (panned to centre, left and right in the input file as in some TV entertainment and sport programming) was now present in the rear surround channels. Although the principal component is clearly defined the PCA process was much less effective at separating speech from competing sources than in the case where speech was in a separate channel. The principal component seems to have been a mix of speech and background music.

6.4.4. Speech-biased PCA Algorithm: Speech between Centre and Left (or right)

One example of a mix that does not adhere to Dolby guidelines is where speech is panned between centre and either left or right. Typically, although fairly rarely, this is used in movies for a close up scene with talkers to left and right of the cinema screen. Speech is still anchored to the cinema screen but the technique serves to give some separation to the voices.



Angle: -16 °

Stereo Spread: 0 °

Figure 52. Panning reassignment for centre channel between left and centre of the 5.1 mix, other channels are mixed as usual

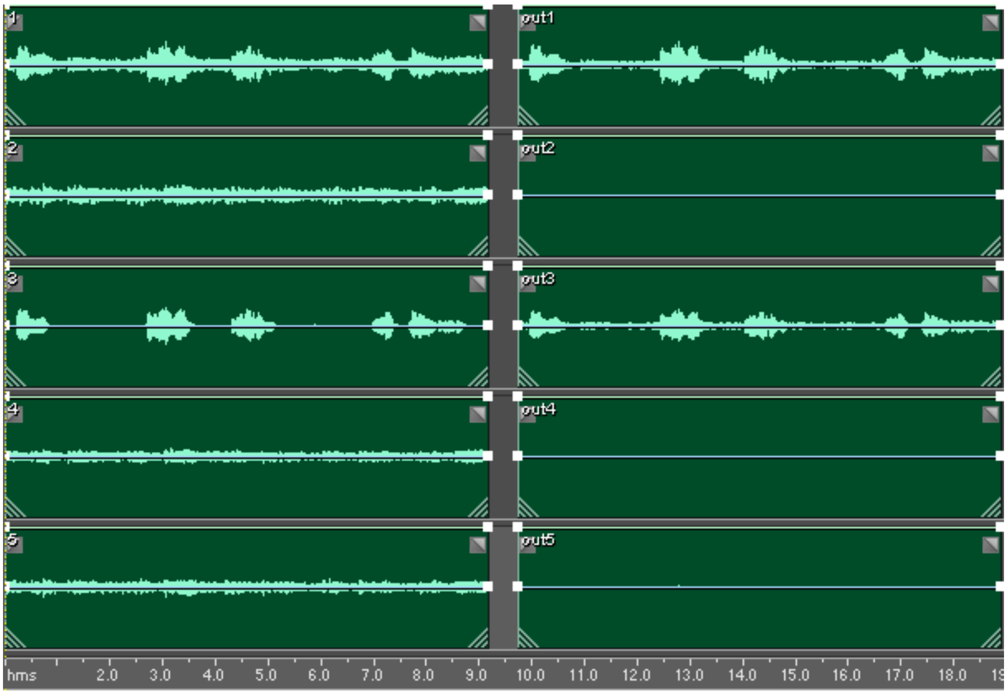


Figure 53. input and output waveforms shown in screenshot showing attenuation of non-speech elements in right and surround channels

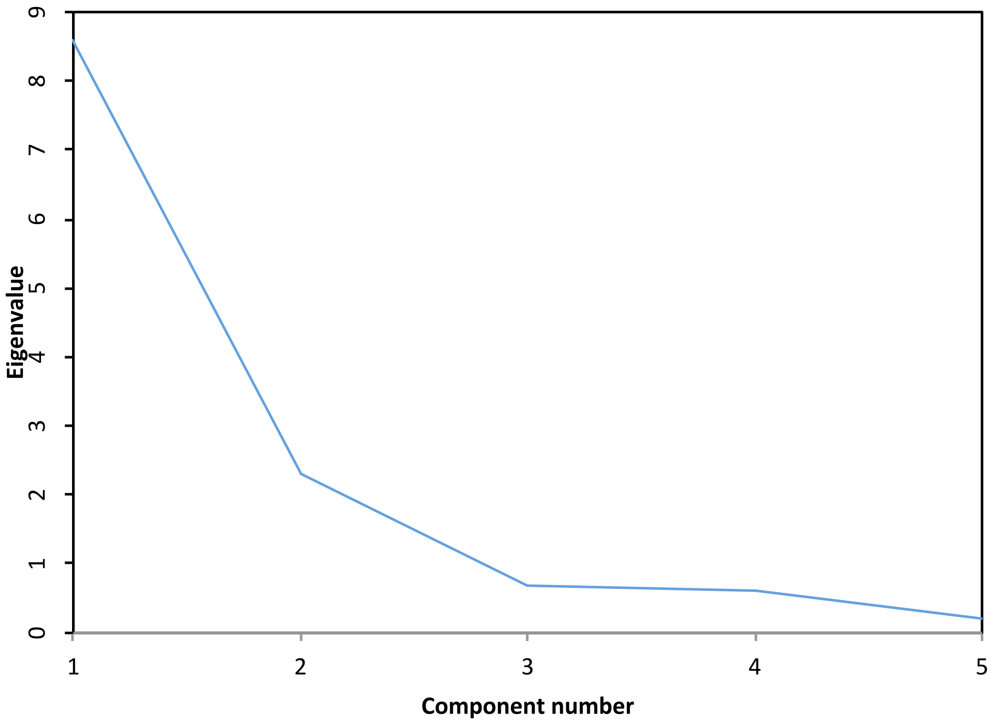


Figure 54. Scree plot of eigenvalues

0.0002	0	0	0	0
0	0.0006	0	0	0
0	0	0.0007	0	0
0	0	0	0.0023	0
0	0	0	0	0.0086

Table 23. Matrix of eigenvalues

The highest eigenvalue (indicating the principal component) was again high at 0.0086 although not as high as in the previous case when speech was across three channels. Subjectively the PCA process reduced the background music more effectively and removed music entirely from right, right surround and left surround. Some music was also added to the centre channel which previously contained only speech content.

6.4.5. PCA Algorithm with Dynamically Allocated Components

One shortcoming of the proposed PCA method is that after user intervention to identify a section of media with speech present it is assumed that speech remains panned in this position for the duration of the programme material. In order to dynamically re-evaluate components in the PCA system code was developed to operate the PCA process on short sections of audio. The code operated in a similar way to that described previously; two parallel paths were utilised as shown in figure 61 with one subject to a speech frequency bandpass filter and the calculated components being applied to the unfiltered audio path. An overlapping Hanning window envelope function was implemented into the Matlab code which split audio into 500ms sections with a 50% overlap. For each 500ms section of multichannel audio all components other than the principal component were muted and the audio reconstituted in order to assess the impact of a dynamically adaptive PCA process on multichannel audio. The input multichannel audio was the same section of *The Matrix* as was used in the first example. It is mixed according to Dolby guidelines with only speech and some foley in the centre channel.

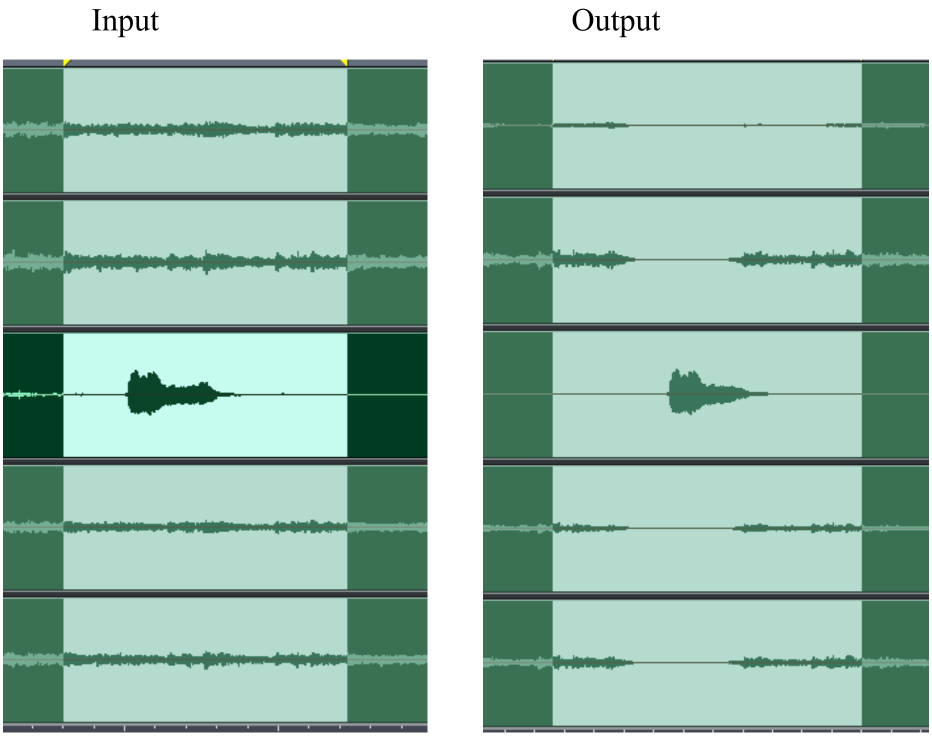


Figure 55. Input and output waveforms from a dynamically adapting PCA process on a 'standard' 5.1 surround mix. Note the gating like effect wherever speech is present in the multichannel content

It can be seen from the screen shot in figure 55 that the dynamic PCA had no effect on the multichannel audio until speech was present in the mix. For the period that speech was present (in centre channel in this example) it is identified as the principal component because of the weighted PCA algorithm and all other components are removed. Because of the 'standard' mix following Dolby guidelines in this case this mutes all channels but the centre channel containing dialogue.

For the cases of speech in other common locations documented in 7.1.4 and 7.1.5 the impact of dynamic analysis of components is predictable based on the examples documented here. For each period where speech was present the PCA process has the same effect as already documented, where no speech is present no effect has been observed. A screen shot example of dynamic PCA output for speech panned between right and centre is shown in figure 56 where it can be seen that the impact is identical to that for non-dynamic PCA but only for those sections of audio where speech is present.

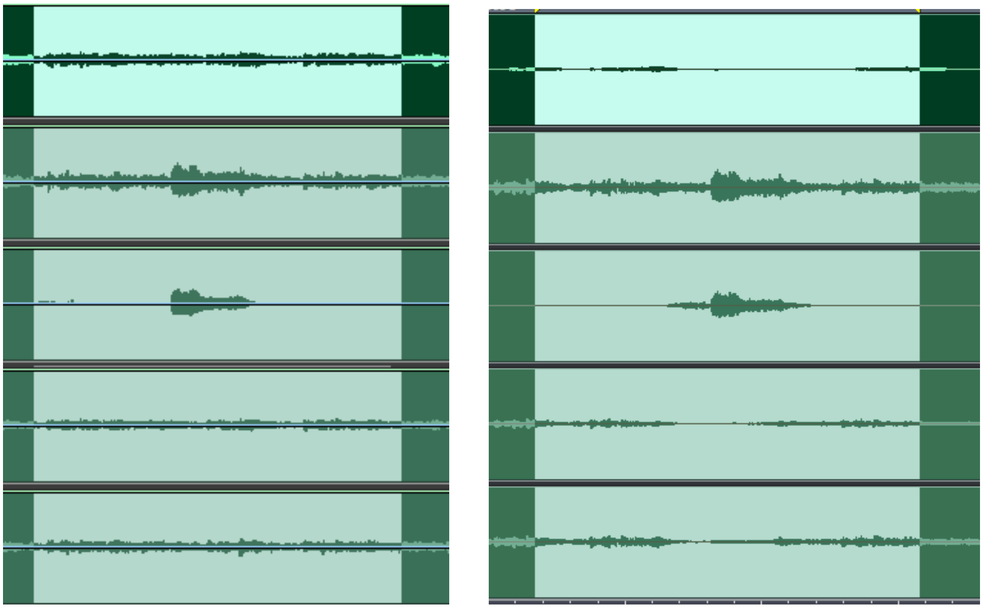


Figure 56. Input and output screenshots where speech is between right and centre channels before and after dynamic PCA processing

Again, as with non-dynamic PCA processing on similar mixes of speech between two channels, there was some reduction in background audio however the effect was much less noticeable than where speech was panned to a single channel.

Although clearly the process shown here would have a positive impact on intelligibility for ‘standard’ mixes following Dolby guidelines the audible effect of dynamic PCA for these mixes was distracting and unpleasant to listen to, sounding much like a ‘gating’ audio effect and so generates a mix that would be unsuitable for a television audience. It is not therefore a useful process for generating accessible audio at the STB. However it is proposed that it could have application at post production and pre-broadcast stages of the broadcast chain as a tool to identify the location of speech content in a 5.1 surround soundtrack. If run as a preprocessor prior to broadcast it could be utilised to automatically set or unset the bit already identified in Dolby metadata (encinfo) to indicate whether clean audio processing (as defined in this thesis and documented in (ETSI, 2009)) would be beneficial to intelligibility and so would be an appropriate treatment for the programme material if a HI output was selected. Where speech was not present in centre channel only no processing or remixing would be carried out.

6.5. Discussion

Some important considerations have to be taken into account before accepting principal component analysis as a useful process for speech detection and enhancement for broadcast as suggested by Zielinski in (Zielinski et al., 2005). Firstly some considerable human interaction needs to take place before the process as defined can be effectively carried out. In that research the choice of which section was used to generate the unmixing matrix was key to the success of the method for the media utilised in the research. Additionally the method assumes that the unmixing matrix generated by PCA of this section is applied to all of the 5.1 material, the assumption being that this will be appropriate for the entirety of the remaining media.

There are two main problems with this approach. Firstly the decision to base the experiments on a contrived 5.1 mix where left surround and centre channels had been swapped is flawed when looking for a solution that can be applied to real world broadcast material. It is incorrect to state, as the paper does, that the Clean Audio solution proposed earlier in this thesis assumes the speech will always be in centre channel. The Clean Audio work presented in chapter 3 rather proposes a situation where processing will only be applied *if* the speech is in centre channel *only*. A single bit in the AC-3 metadata would be set to 1 or 0 depending on whether a clean audio process was appropriate. This bit had been identified by Dolby at the time of the original research (encinfo) however the imminent release of Dolby Digital+ (E-AC-3) and potential issues for some legacy equipment made implementation unlikely. Secondly, for all of the media examples that were analysed during the research carried out in this thesis, wherever the speech was not present in centre channel, it was always present in more than one channel, usually centre and left, centre and right or centre, left and right. In some movie content analysed the speech was also dynamically changing panned position; for most of these movie examples the speech was *usually* in centre channel - the instances stated above, such as between centre and left or right, were for specific scenes and not consistent throughout the media. In these circumstances the weighted PCA solution proposed is at best unpredictable and for the fairly common TV broadcast scenario of mixes with speech across three channels it is largely ineffective as indicated by the experiments documented here.

The adaptation of the technique using dynamically changing PCA components, while avoiding the issue of changing mixes scene-by-scene, is also inappropriate to directly generate accessible audio at the set top box. Its gating effect is unpleasant and distracting to listen to and the variable attenuation caused by the aforementioned mix shifts between scenes make it unpredictable. It is possible however that a dynamic PCA method such as that applied here may be useful in generating metadata to be embedded in media prior to broadcast.

6.6. Conclusions

Using speech biased principal components analysis as proposed by Zielinski et al (2005) has been show to be effective only for mixes following Dolby guidelines and ineffective when assessed using other common mixing paradigms used in film and television. The technique relies heavily on consistency as to where speech is panned throughout the duration of the media content and requires user input wherever speech resides elsewhere in a 5.1 surround mix. An adaptation of the technique documented here utilising dynamic adaptation of PCA components is shown to be effective at picking out speech across a range of 5.1 mixes and may have application in automatically generating metadata which could be appended to untagged media content prior to broadcast indicating whether a ‘clean audio’ mix would be appropriate as a HI output. Further experimentation would be required to assess the performance of this use compared to other speech detection algorithms (Van Gerven and Xie, 1997). For example, a single tag could state whether speech was *consistently* in centre channel only or tags could be added at regular intervals indicating whether it was appropriate for large or small sections of the media content. Given that Dolby metadata is constantly received by the STB at the user end it would be feasible to set or unset an accessible audio bit for almost any duration of programme material and update this option on a regular basis.

7. Accessible Audio Implementations

The research outlined in this thesis has already impacted on the development of standards and guidelines for digital TV and these have been documented earlier in this thesis together with details of how the research findings have informed development of international guidelines for IPTV.

This chapter describes two experimental implementations of accessible audio for hearing impaired people that have resulted from the research documented in this thesis. The first was implemented by Dolby Labs (San Francisco) and was based on research to generate accessible TV audio following their involvement in the UKCAF. The second example was implemented by the author as part of the EU FP7 FascinatE project which developed a complete end-to-end future AV broadcast system including an object based implementation of clean audio. The first section therefore documents test design and listening tests supervised by the author under a Dolby Labs funded project at the Acoustics Research Centre at University of Salford which aimed to assess the potential for a process developed by Dolby Labs to improve TV audio for hard of hearing people. The second section of this chapter documents the FascinatE clean audio solution and the developments required in production techniques in order to apply this method to TV broadcast. These represent both a potential solution driven by industry for current broadcast systems (in the case of Dolby Labs) and a solution implemented for future broadcast systems (FascinatE's object based accessible audio solution).

7.1. Dolby 'Clear Audio' Implementation

7.1.1. Introduction

Following the original Clean Audio Project research and it's presentation at DVB in Geneva a clean audio implementation was developed by Dolby Labs. The process utilised a speech detection algorithm to identify if speech was present in centre channel and attenuated other channels as per the EBU guidelines which came from chapter 3 of this thesis. In addition it also applied multi-band compression techniques such as those found in digital hearing aids to the centre channel. The details of the process itself are not covered in this thesis however test design and subjective assessment of the processes are relevant and are documented here. A series of tests were implemented in collaboration with Dolby Labs in order to assess the effectiveness of audio processing developed with the aim of improving TV sound for hard of hearing people. The methodology involved subjective assessment of AV media; firstly user assessments to identify appropriate media clips and secondly subjective assessment of processing on these clips identified as being effectively equivalent in terms of speech clarity.

7.1.2. Methodology

Methodology was broadly similar to that adopted in previous chapters of this thesis with forced choice paired comparison tests chosen as the most appropriate means. An additional testing stage stage for media selection was added with the intention of ensuring that pairs of clip sections used in the tests were shown to be equivalent by a panel of test participants. Clips were presented over three loudspeakers setup according to the left, centre and right loudspeakers in ITU-R BS.775-1 in a listening room confirming to ITU-R BS.1116-2.

7.1.2.1. Listener selection

Hard of hearing and normal hearing participants were recruited by two methods; via hard of hearing user groups and via a pool of listeners who had previously taken part in listening tests. Hard of hearing listeners took part that watched TV either with or without hearing aids but without the use of subtitles or other accessibility aids. An analysis of listeners who took part in each part of the testing process was carried out to assess hearing acuity.

7.1.2.2. Selection of test material

The choice of a paired-comparison test paradigm required generating a set of suitable pairs of content. In previous tests this content had been chosen based on stated criteria however for these listening tests an additional stage of assessment was built in to attempt more stringent identification of suitable clips. These test pairs had to satisfy two criteria:

1. The two sections of each clip pair had to be rated as equivalent on a scale of the perceptual attribute to be tested (i.e., perceived ease of understanding) and
2. The difficulty of understanding the items should be rated ‘easy’ by non-hearing impaired listeners and progressively more ‘difficult’ relative to the hearing loss of the subject.

The second requirement aimed to eliminate material that was inherently difficult to understand for reasons other than hearing loss, for example because of poor mixes, poor enunciation, or of prior knowledge needed to comprehend the speech.

A selection of AV media with audio in 5.1 surround format was collected from broadcast and fixed media sources. These provided the basis for test material selection and were sampled from the genres movie/drama, news and documentary, animation, stage shows, and sport. Initially, the recorded material was edited into 82 clips each split into two sections. Each section’s duration was approximately 15 seconds. Care was taken to match the two members of the pair as closely as practical with regard to talker, speaking style, type and level of background sound, and dialogue spoken with the talker facing the camera.

Eighteen listeners viewed each of the 82 pairs and answered the question “In which section was the speech more difficult to understand.” Listeners were also asked to rate the effort required to understand what was said in the more-difficult section on a labeled 5 point scale. The rating scale was labelled as follows:

1. Complete relaxation possible, no effort required;
2. Attention necessary, no appreciable effort required;
3. Moderate effort required;
4. Considerable effort required;
5. No meaning understood with any feasible effort.

The data was analysed to identify subsets of pairs that satisfied both criteria stated previously. Pairs that were mismatched in terms of ease of understanding were identified and rejected. The data was analysed to identify any pairs that were mismatched in their difficulty with a confidence level of 95% using a binomial sign test for a single sample. 14 pairs of clip sections were rejected on this basis as being non-equivalent.

The second requirement for the clips was to ensure that difficulties were as the result of hearing loss rather than a problem with the clip mix or other factors inherent to the clip. Clips had to be easy to understand for non-hearing impaired people but cause difficulties in understanding for hearing impaired people based on their degree of hearing impairment. To this end the difficulty ratings assigned to clips by participants were plotted as a function of a hearing loss descriptor and ranked based on variance from a least squared regression line. The better half of all pairs was then retained.

The combined selection criteria resulted in 36 clip section pairs that passed both tests and these were retained as the final test material.

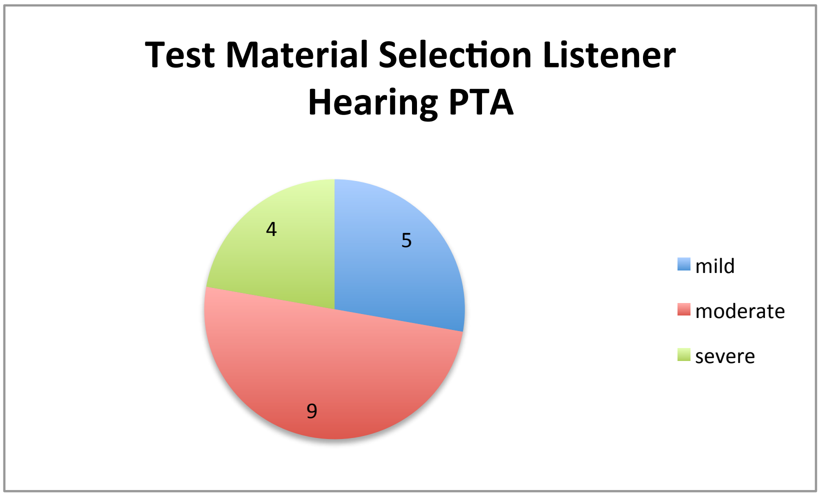


Figure 57. Listener hearing pure tone audiogram data for test material selection test

7.1.2.3. Listening Tests

Listening tests were carried out in a listening room complying with ITU-R BS.1116. 38 hearing impaired and 15 normal hearing participants participated in the tests. Each subject had an assessment of hearing acuity using standard audiogram techniques prior to tests being carried out (figure 58).

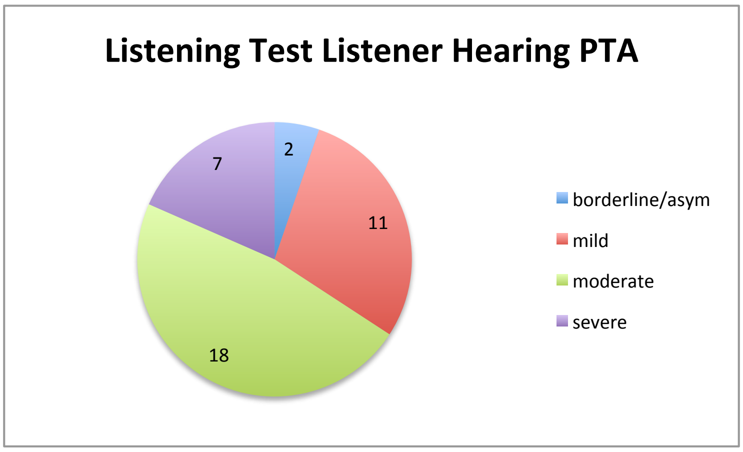


Figure 58. Listener hearing pure tone audiogram data for test material (38 participants)

Participants were asked to view two sections of AV material subject to different conditions and to choose their preferred section in terms of overall sound quality and speech clarity. They were also asked to rate how much better their preferred clip was for each choice.

Processes assessed were:

- Clean Audio with surround channels removed and left and right channel at -6dB as per ETSI TS101154 Annex E.4 (based on research documented in chapter 3 of this thesis)
- Variation 1 of Dolby's clear audio process.
- Variation 2 of Dolby's clear audio process.
- Unprocessed reference

The answer sheet used by participants can be found in Appendix D. Clip order was varied such that each condition was compared an equal number of times as first and as second section in order to avoid the recency effects noted in earlier research (Shirley and Kendrick, 2004) and documented in chapter 3 of this thesis.

Presentation of material was identical to that used in the previous research already mentioned: clip sections were played back as follows:

a. Black screen, title: Clip 1
b. Black screen, title: Section A
c. Section A with first condition to be assessed played
d. Black screen, title: Section B
e. Section B with other condition to be assessed played

It was considered useful after each set of tests to informally discuss with each subject their experience of the tests and of TV sound and their hearing difficulties. In several studies carried out as part of this PhD such unstructured or semi-structured informal interviews with participants have been useful in providing understanding of results obtained, and in several cases provided real insight into the experience of hearing impaired and older people in interacting with broadcast technology and television audio. During some of these interviews after early pilot tests it became clear that a small number of the participants were unclear about, or had misunderstood, the instructions for the test. Results to that point were considered unreliable and these were discarded. The test instructions were then redesigned based on information obtained from the participants who had taken part in the tests so far, and, from then on, a trial test was carried out with each subject with a researcher present to guide them through the procedure. A break was also arranged part way through each participant's tests and a second researcher queried the subject about how they were finding the test and asked what they were being asked to do. This functioned both as a natural break to avoid fatigue and also as a check that the subject understood instructions clearly. After these alterations had been made to the test procedure it was felt that tests could proceed with confidence, that participants would understand the test instructions clearly and any misunderstandings about test procedures would be identified quickly.

7.1.3. Experimental Results

Two factor mixed measures ANOVAs were carried out for speech clarity and for sound quality ratings. Conditions assessed were as follows:

UP	Unprocessed
EBU	Clean Audio with surround channels removed and left and right channel at -6dB as per ETSI TS101154 Annex E.4.
DLB	Dolby clear audio process, example 1.
DLB	Dolby clear audio process, example 2.

Overall Sound Quality

Main effects from the two factor ANOVA showed no statistically significant difference in sound quality ratings between hearing impaired and non-hearing impaired groups. The audio condition main effect however showed a significant effect on ratings of overall sound quality.

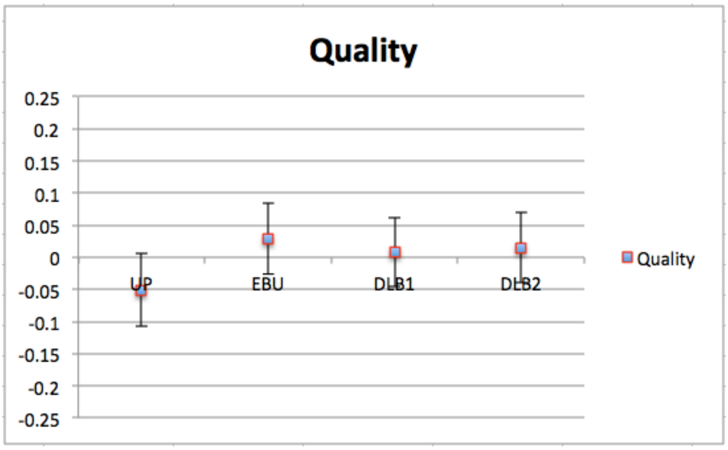
Speech Clarity

Looking at main effects there was a statistically significant interaction between the participants' hearing ability and audio condition on perceived speech clarity. Although there was no statistically significant difference in clarity between HI and non-HI groups for the UP, EBU and DLB1 conditions there was shown to be a statistically significant difference in clarity between HI and non-HI groups for the DLB2 condition.

One way repeated measure ANOVAs were carried out for speech clarity and for sound quality ratings for hearing impaired and non-hearing impaired groups separately.

Overall results from the repeated measures ANOVA for the hearing impaired subject group at a 95% confidence level indicated no significant differences between the means for either sound quality or for speech clarity. Because main results across both groups indicated some significant differences between the hearing impaired and non-hearing impaired groups the results are presented here for completeness (table 59 A&B).

Hearing Impaired Group (15 participants): Sound Quality Mean Rating



Hearing Impaired Group (15 participants): Speech Clarity Mean Rating

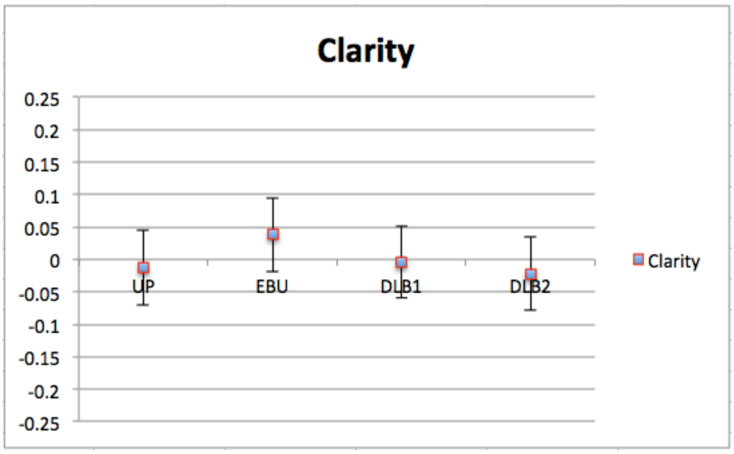
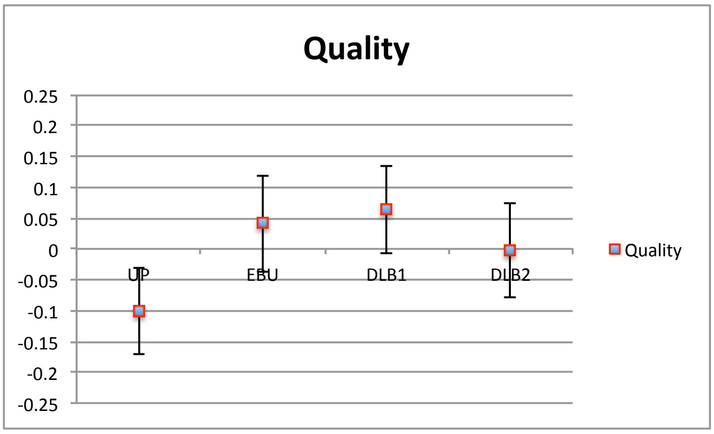


Figure 59 A & B. Plots showing values obtained for sound quality and dialogue clarity for the hearing impaired group. Error bars indicate 95% confidence interval.

As can be seen from tables 24 and 25 limited significance can also be drawn from these tests for the non-hearing impaired group. Sound quality ratings for the unprocessed condition (*UP*) were lower than for the *EBU* condition (based on chapter 3 recommendations and published in ETSI TS101154 Annex E.4) and also for *DLB1* (the least heavily processed of the two Dolby conditions). No significant difference in means was found when unprocessed was compared to *DLB2.* When considering mean ratings for speech clarity all conditions were rated significantly higher than *DLB2*, the more heavily processed of the Dolby processing conditions.

Non-Hearing Impaired Group (15 participants): Sound Quality Mean Rating



Non-Hearing Impaired Group: Speech Clarity Mean Rating

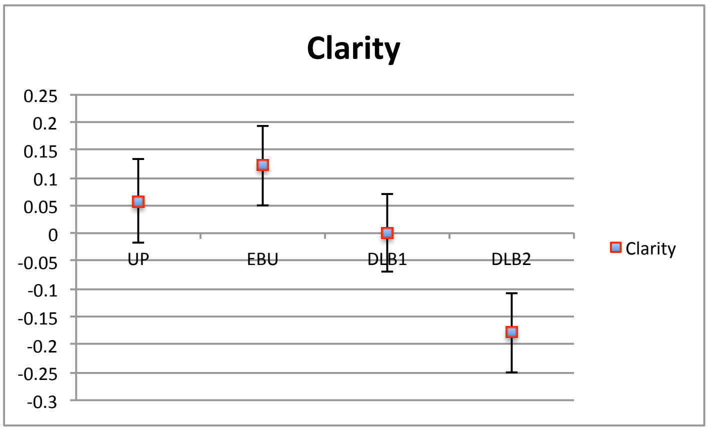


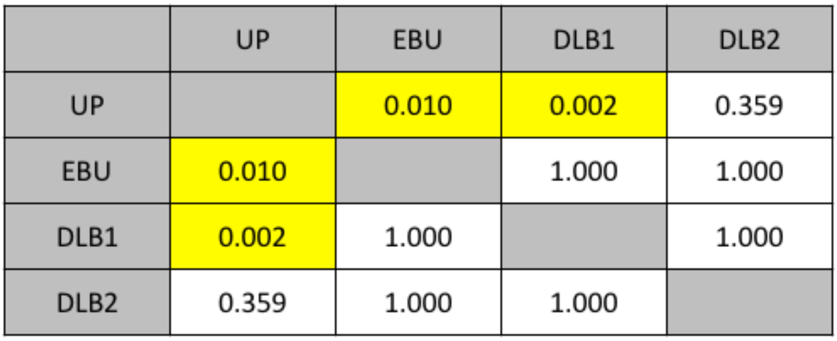
Figure 60 A & B. Plots showing values obtained for sound quality and dialogue clarity for the non-hearing impaired group. Error bars indicate 95% confidence interval.

7.1.4. Statistical Significance of Results

As already discussed no statistically significant results were obtained for hearing impaired participants. Statistically significant results for the non-hearing impaired group are presented here.

7.1.4.1. Non-Hearing Impaired Participants

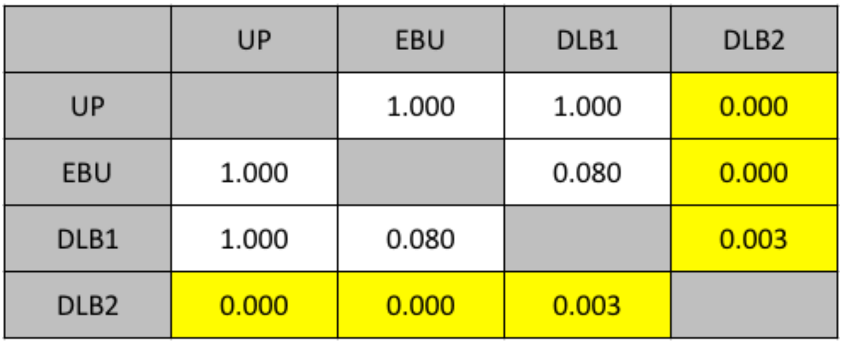
Sound Quality



	UP	EBU	DLB1	DLB2
UP		0.010	0.002	0.359
EBU	0.010		1.000	1.000
DLB1	0.002	1.000		1.000
DLB2	0.359	1.000	1.000	

Table 24. P-values for each pairwise comparison for sound quality ratings, highlighted values <0.05 indicate statistical significance.

Speech Clarity



	UP	EBU	DLB1	DLB2
UP		1.000	1.000	0.000
EBU	1.000		0.080	0.000
DLB1	1.000	0.080		0.003
DLB2	0.000	0.000	0.003	

Table 25. P-values for each pairwise comparison for speech clarity ratings, highlighted values <0.05 indicate statistical significance.

7.1.5. Conclusions

For the hearing impaired group no significance was found from ANOVA analysis. One potential contributing factor for the lack of strong evidence could be that the wide range of hearing impairments present in the subject group led to them responding very differently when rating the conditions. Although the range of hearing impairments is broadly comparable to previous listening tests, this is the first set of tests in this thesis where a frequency dependent process has been assessed. A contributing factor may be

that the process developed by Dolby carries out some multi-band compression techniques on the audio, similar to DSP carried out in hearing aids. This could have had unintended effects in two ways. The parameters of digital hearing aids are adjusted individually based on an individuals hearing impairment however for *DLB1* and *DLB2* conditions general settings were attempted based on an 'average' hearing impairment. It is likely that for some hearing impaired individuals this processing actually made speech clarity and quality worse. For the non-hearing impaired subject group this was indeed the case and the more extreme of the processed conditions was rated lower than all other conditions for speech clarity ($p<0.05$) by this group. Participants experienced the media 'as they would at home' so some of the hearing impaired group wore hearing aids during testing and the addition of an extra stage of processing could have had unpredictable consequences on ratings. There is an unfortunate side effect of the nature of the test method adopted here which was not apparent in previous tests because of the nature of the conditions assessed. Because every condition is tested against every other condition many times (AB, BA and every clip with every process to avoid recency and section non-equivalence effects), where there is a condition, or conditions, present that are rated very differently by participants in the same group it can create statistical noise that will reduce any significance in the results. It is thought that this is the main contributing factor here. Where there is potential for very different experiences between participants of the same condition (e.g. frequency based processing with participants using hearing aids) a different test method should perhaps have been used.
An additional factor not present in previous tests was the extensive pre-testing of clips to try and ensure section equivalence during testing. Given the range of hearing impairments present in participants who took part in these pre-screening tests, it is also possible that the sections were less useful and less equivalent than those picked manually by a non-hearing impaired researcher based on strict criteria for listening tests (as in chapters 3 and 4 of this thesis).

The clearest conclusion that can be drawn from the results of these tests is on listening test design; simplicity in test design may be key to reducing unknown variables. Specifically reducing the number of condition comparisons would reduce the impact of unpredictable ‘outlier processes’ that are perceived very differently by different participants in the same group and so reduce significance. More generally the additional steps introduced to try to make the tests more valid may also have introduced unexpected additional variables that impacted on results in unknown ways and created further statistical noise reducing the significance of results.

7.2. *Object Based Audio Solution: The FascinatE Project and Accessible Audio*

The FascinatE project (Joanneum Research et al., 2010) was an EU FP7 project that developed a complete end-to-end future broadcast system designed to be format agnostic and interactive based on user navigation of an ultra high definition panorama with accompanying 3D audio. As the project partner responsible for much of the audio part of the FascinatE project the author implemented accessible audio as part of the project deliverables. The FascinatE project outcomes give an interesting view of how accessible audio could be implemented in a future object based audio system alongside other interactivity. This section presents an overview of object based audio and it's purpose in the FascinatE project, describes the project implementation of accessible audio and describes the techniques developed that would need to be adopted by the broadcast industry in order for these methods to become a reality. The FascinatE project completed in July 2013. Some of this material is adapted from (Shirley, 2014).

7.2.1. Object Based Audio Review

In the FascinatE project object based audio was utilised as a means to provide a dynamically matching audio for interactive navigation through an AV scene. The project captured a very high resolution panorama of 7K resolution (approx 7K x 2K pixels) and allowed pan, tilt and zoom navigation of the panorama by the user. In order to provide matching audio for the user defined scene it was necessary to move away from a conventional channel based audio paradigm.

Instead an object based paradigm was used to capture the audio scene without reference to any specific target loudspeaker configuration. Instead of defining captured audio events as emanating from a given loudspeaker or from between two loudspeakers of a target loudspeaker layout as is the case with channel based audio, events were captured complete with 3D coordinate information specifying where in the audio scene the event had taken place. This is analogous to a gaming audio scenario and, in the same way as a first person game allows navigation around and between audio objects, the object based audio capture enabled users to pan around and zoom into the AV scene with audio events remaining in their correct locations. It was possible in the FascinatE system to zoom across a scene and past audio objects which would then move behind the user's viewpoint thus realising a realistic audio scene to match the chosen visual viewpoint.

Other non-spatial uses of object based audio have been proposed; BBC research has implemented object based audio in a test radio broadcast which used audio objects to tailor the radio programme depending on the listeners geographical location (Forrester and Churnside, 2012). Object based audio allowed specific audio events that made up the programme such as sound effects, actors' voices and music to be customised based on geographical location and the date and time of access to the radio programme. The programme was delivered over IP and used the HTML5 standard to carry out all audio processing and mixing at the user end. Another use of audio objects proposed by the BBC was to be able to change the duration of a programme by adjusting the spaces between audio events, without any need for time stretching or other process that may be detrimental to audio quality or intelligibility. An implementation of object based audio by Fraunhofer is described in a paper not yet published at the time of writing and involves using parametric data to unmix signals from a transmitted stereo, or 5,1, mix.

7.2.2. Object Based Accessible Audio in the FascinatE Project

By combining the production and acquisition techniques developed in the FascinatE project, and applying some of the customisation principals of perceptive media a system was developed that could provide an accessible audio output without any need to address the issue of separating speech components from the background 'noise' that has been the subject of much of this thesis. By maintaining a separation of all audio components and objects that make up a programme throughout the broadcast chain it was possible to enable mixing of every aspect of broadcast audio at the user end based on user preferences including a requirement for accessible audio for hard of hearing people.

For a produced programme such as that described in the Forrester and Churnside's work on perceptive radio it is relatively trivial to enable a hearing impaired mix to be made via simple user command; sound effects and music could be lowered relative to dialogue for example, and an implementation of 6dB or any other attenuation of non-speech content (even one customised based on user requirements) would be straightforward and could be readily adapted for accessibility purposes. The FascinatE project however focused on broadcast of live events and this created considerable challenges for the provision of object based accessible audio. The project covered a range of event genres as part of test shoots and the example given here, that of live sports broadcast, is of most relevance to the concept of clean, or accessible audio.

7.2.3. Accessible Audio for Live Sports Coverage

Consideration of providing accessible audio for sports coverage introduces an interesting question as to what audio should be considered useful and retained, and what should be considered as background noise that may be considered detrimental for comprehension of the programme material. The FascinatE project used the example of live football coverage as one of its test scenarios and this provides a good illustration of techniques developed that would be of equal relevance to other live event genres. Clearly, as in other programme genres discussed in this thesis, speech (in this case commentary) is an important component to understanding the narrative of and meaning of events during the football game. One could gain a clear understanding of what is happening on the football pitch by listening to the commentary channel alone however, for example, the sound of a referees whistle also provides meaning. Sound design techniques such as hyperreal and exaggerated ball kick sounds that have become commonplace over the last 20 years indicate that all of these on-pitch sounds are considered important to the experience of enjoying a televised football game. Indeed the exaggerated on-pitch sounds introduced to live sport by Sky have been adopted in computer gaming and have become synonymous with live sports coverage. There is a parallel here with diegetic and non-diegetic sounds in film. Diegetic sounds are usually defined as “sounds whose source is visible on the screen or whose source is implied to be present by the action of the film” (Carlsson). In improving the TV experience of hard of hearing people it may be that diegetic sounds that are critical to the narrative of the programme should be treated differently to the background ‘noise’ whose reduction has been the focus of this thesis. To this end three categories of sounds were considered here; speech content whose comprehension is critical, background noise that has been shown to be detrimental to both clarity and to perceived overall sound quality, and other non-speech sounds that are considered important to comprehension and/or enjoyment of the material. In approaching an object based audio broadcast these should each be capable of being processed or mixed independently either directly by the user, or based on predetermined user preferences at the set top box. In the example of live football broadcast these categories consisted of speech in the form of live commentary, crowd noise that could be considered as detrimental to comprehension, and on-pitch sounds such as ball kicks and the referee's whistle blows that may be important for comprehension and also for perceived overall sound quality. In current TV broadcasts these discrete audio object categories are not available at any point in the broadcast production chain.

In order to provide these three sound sources as independent and controllable entities some considerable development had to take place in the acquisition and production techniques used to capture a complex sound scene such as that found at a live sports event. Currently the key objectives for audio in football coverage are twofold; picking up sounds on the pitch as clearly as possible during the game, and utilising the 5.1 surround sound capability to give the viewer a sense of immersion and of 'being there'. These objectives are achieved by use of two separate microphone setups common to Premiere League coverage and also coverage of World Cup and other international football.[7]

For on-pitch sounds the ball kicks and whistle blows are happening some distance from any possible microphone position so shotgun microphones are used to pick them up. Twelve shotgun microphones are positioned around the football pitch facing towards the action. If all of the microphones are live in the mix at any given time the background noise from the crowd swamps the sounds from the pitch making them inaudible. In order to prevent this from happening, microphones are mixed live so that only the microphone, or microphones, closest to the ball location is in the mix at any given time.

[7] Although this setup is common for football, capture techniques vary between live events.

This requires a skilled sound engineer to follow the action on the pitch on the mixing console in the outside broadcast truck and ensure that only the appropriate microphones are active in the mix. As the broadcast is live the engineer must predict where the ball is going to be next but also has to be aware of what likely camera angles will be chosen by the producer. At any given moment during the event between one and three microphones will be active in the mix. All of these microphones are panned centrally, either to a central loudspeaker, or more often to a phantom centre between left and right in order to avoid any potential issues from the downmixing process.

The crowd sound for live football coverage is considered key to building the atmosphere for the television viewer and is usually picked up by a single Soundfield microphone suspended from the gantry above the crowd. The Soundfield microphone consists of four near-coincident microphone capsules arranged as a tetrahedron. The four outputs are encoded into a B-format ambisonic (Gerzon, 1980) signal by a microphone controller on the gantry. The B-format signals from the Soundfield microphone define the sound in three dimensions at the microphone location and these can, if desired, be decoded for periphonic (with height) reproduction. The four B-format signals (W, X, Y and Z) are sent to the OB truck as AES 3-id (AES, 2009) signals on unbalanced BNC cables. For television broadcast the Z (height) component is ignored and the B-format signals are decoded into a 5.1 feed at the OB truck. This 5.1 crowd noise channel is mixed into surround and left and right channels of the 5.1 programme audio both to give a more immersive experience for viewers and also to cover up any audible artefacts from mixing between the pitch-side microphones. Although the pitch-side shotgun microphones pick up many of the sounds on the pitch that are of interest the audio feeds from these also contain large amounts of crowd noise. Trying to separate these on-pitch sounds by reducing the mix of the Soundfield microphone dedicated for crowd noise leads to unpleasant and disorientating effects as microphones in different areas of the stadium are faded in and out. Therefore in order to provide an object based accessible solution such as that described there was a need to develop a method of separating out on-pitch sounds effectively from crowd noise.

Method: New Production Techniques for Audio Acquisition

The development documented in this section was carried out as part of Work Package 2 of the FascinatE project (Joanneum Research et al., 2010) by the author and a Research Assistant on the project (Rob Oldfield) and some parts of the work are published in (Shirley et al., 2014)(Oldfield et al., 2012) and (Kropp et al., 2011). Research was carried out by the author, software development for audio object extraction was carried out out by the Research Assistant under the direction and supervision of the author, test captures of live events were carried out in collaboration with other members of the FascinatE project consortium and with the cooperation of SISLive, Chelsea Football Club and the BBC.

In order to extract on-pitch sounds from the audio scene as audio objects techniques were developed that were designed to cause minimum change to the current workflows of sound engineers. The methodologies adopted utilised the same microphones that are already used and were designed to provide a largely automated system for the broadcast production team. Considerable research was carried out into existing workflows, interviews were held with sound engineers from SISLive and Sky and site visits carried out to outside broadcasts to elicit a detailed understanding of the processes currently carried out and of the problems associated with providing robust separation of sounds that could both be used for spatially positioning the resultant audio object (in the FascinatE project) and that would be potentially useful for comprehension and enjoyment of hard of hearing people in more traditional broadcast scenarios.

Microphone feeds from every microphone used were captured on site including all pitch-side microphones and a Soundfield microphone and an Eigenmike (Barnhill et al., 2006) capturing crowd noise. These were stored on hard disc for later processing along with a separate BBC commentary feed. Audio object templates were developed for each class of sound that was required to be captured as a discrete object, in this case templates for ball kicks and whistle blows based on envelope and spectral content were created. Software was developed that monitored every pitch-side microphone, comparing it to the stored audio object template. When a matching sound event was detected in a microphone feed all other microphone feeds were scanned for matching characteristics to identify all microphones that had some element of the event present in its feed. For every pair of microphone feeds that picked up the event a hyperbola, along which the event must have occurred, was calculated based on the difference in time of arrival of the sound at the two microphones. Where more than two microphone pairs had captured the sound event the intersections of the calculated hyperbolas gave an accurate coordinate location for the sound event. In the FascinatE project, which utilised object based audio for spatial positioning of audio objects in surround and 3D reproduction systems, these coordinate locations were used to automatically spatially position the sound dependent on a user defined viewpoint, by a *virtual director* system (Weiss and Kaiser, 2012) or on a viewpoint defined by the production team. In the case of audio objects for accessible audio the critical factor is different; the audio objects would still be panned centrally, as in current broadcast, and the key outcome is to identify the event, and extract it from the acoustic scene in isolation from crowd noise and other sound that may be detrimental to clarity.

In order to accomplish this the microphone feed containing the loudest example of the sound event was identified based on the derived location of the sound event. Once this microphone was identified an envelope was applied to the microphone feed based on the temporal characteristics of the detected sound event. In this way relevant pitch-side microphones were only ever active in the produced mix for the duration of the sound event. The short section of crowd noise picked up by that microphone was then effectively masked by the sound event itself. A flow diagram for the object extraction process can be seen in figure 61. The resultant audio object, together with its paired coordinate metadata was coded into the broadcast audio stream for decoding at the rendering device.

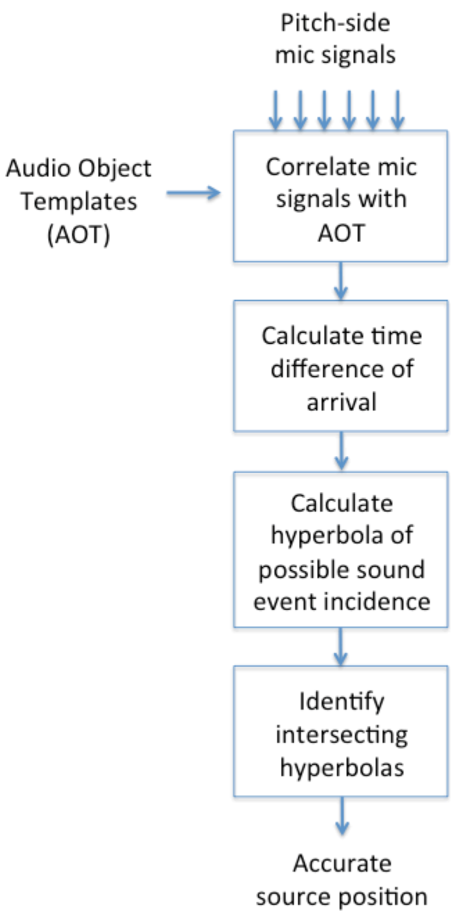

Pitch-side mic signals
Audio Object Templates (AOT)
Correlate mic signals with AOT
Calculate time difference of arrival
Calculate hyperbola of possible sound event incidence
Identify intersecting hyperbolas
Accurate source position

Figure 61. Flow diagram illustrating audio object extraction developed in the FascinatE project.

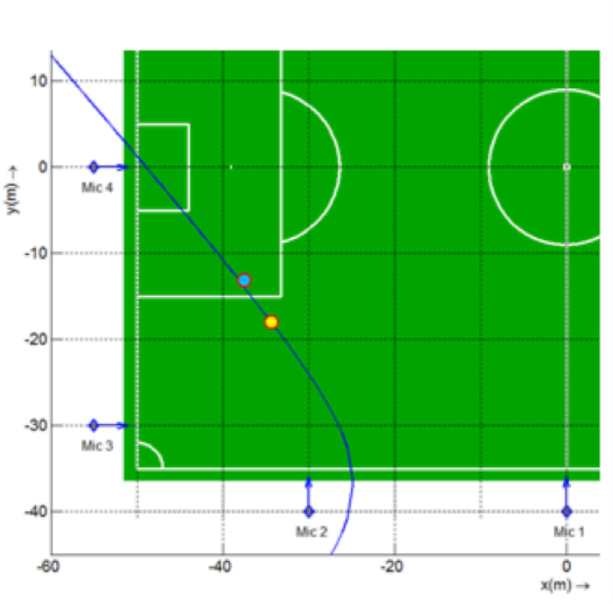


Figure 62. Hyperbola indicating possible locations of sound event based on time difference of arrival at 2 microphone positions.

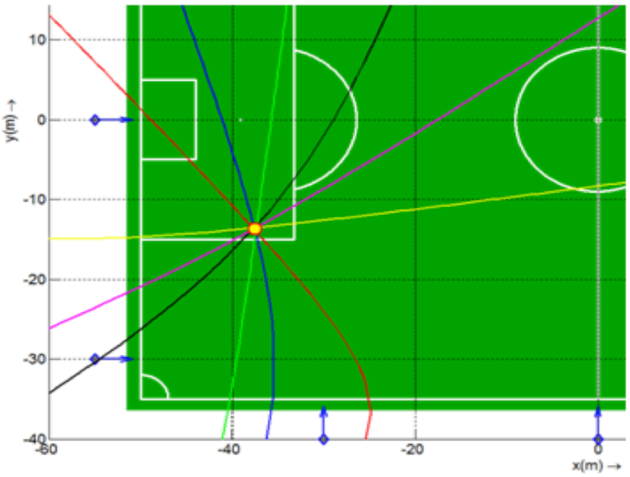


Figure 63. Intersecting hyperbolas indicating derived location of sound event based on time difference of arrival at multiple microphone positions

The user of the system was presented with an interface enabling selection of pre-determined reproduction mixes the input of which was three discrete streams of audio.

1. Clean BBC commentary feed with no background noise taken directly from the commentary microphone.
2. Crowd noise from the Eigenmike located on the gantry at the camera position
3. On-pitch sounds extracted using the audio object extraction techniques described here.

The three streams were combined and transmitted over IP to the audio part of the FascinatE Render Node (FRN). The FRN consisted of the audio composer (developed by University of Salford) and the audio presenter (developed by Technicolor). The audio composer was responsible for decoding the various audio streams and metadata received, arranging them spatially and mixing them based on user input and on scripts generated as part of the production. User input included the chosen user defined scene including pan, tilt and zoom information necessary to spatially position the on-pitch audio appropriately for the chosen camera view, and also user-choices of relative levels of foreground (commentary and on-pitch) and background (crowd) sound. Script information was generated in two ways, at both production and user end.

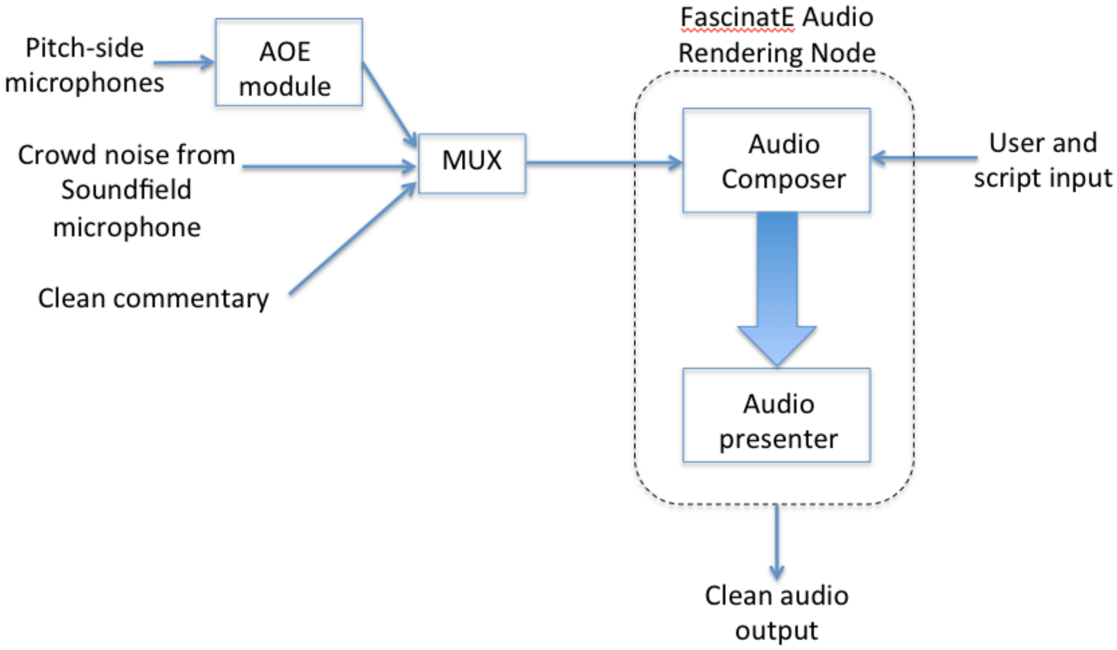
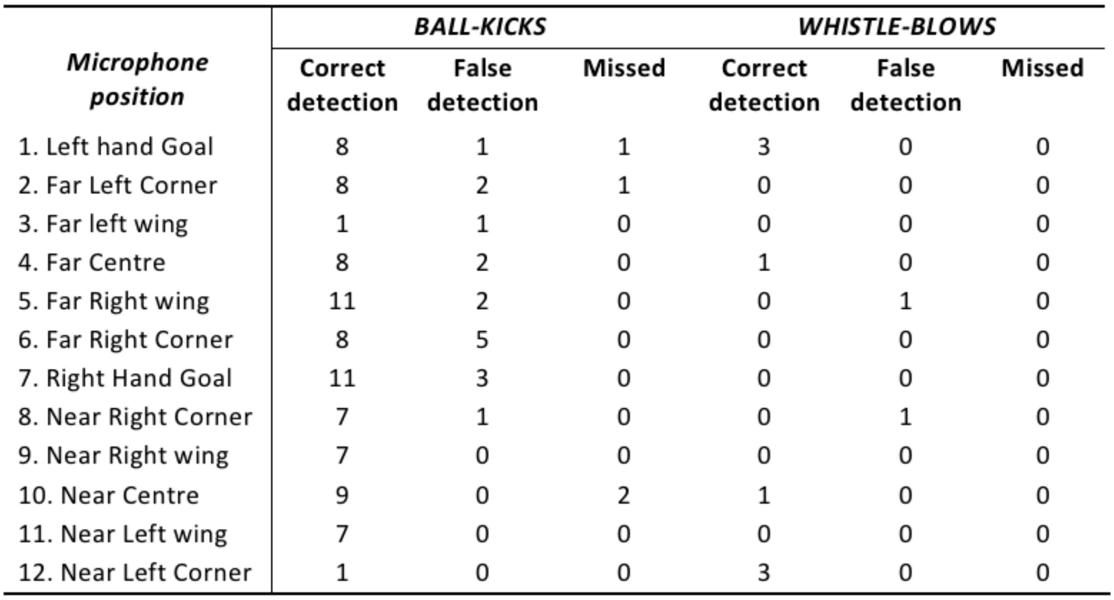


Figure 64. Clean Audio in FascinatE: Extracted audio objects, crowd noise and clean commentary multiplexed for transmission, decoded and reproduced by the FascinatE audio render node

At the production end scripting data was generated both automatically and manually including automatically chosen regions of interest determined by the virtual director (developed for FascinatE by the Joanneum Research Institute) or by producers. At the user end user preferences in the rendering device would provide further scripting input and this included choices based on user access requirements such as preferred speech level relative to background noise.

Results

The effectiveness of the techniques developed here in successfully detecting and extracting sound events compared to current broadcast methods was assessed by comparing events detected and extracted with the number of events present in BBC broadcast of the football match. These results are shown in table 26.



Microphone position	BALL-KICKS			WHISTLE-BLOWS		
	Correct detection	False detection	Missed	Correct detection	False detection	Missed
1. Left hand Goal	8	1	1	3	0	0
2. Far Left Corner	8	2	1	0	0	0
3. Far left wing	1	1	0	0	0	0
4. Far Centre	8	2	0	1	0	0
5. Far Right wing	11	2	0	0	1	0
6. Far Right Corner	8	5	0	0	0	0
7. Right Hand Goal	11	3	0	0	0	0
8. Near Right Corner	7	1	0	0	1	0
9. Near Right wing	7	0	0	0	0	0
10. Near Centre	9	0	2	1	0	0
11. Near Left wing	7	0	0	0	0	0
12. Near Left Corner	1	0	0	3	0	0

Table 26. Comparison of number of ball kicks and whistle blows detected and extracted compared to those audible in BBC broadcast of the same football match (Oldfield et al., 2012)

7.2.3.1. Conclusions

An object based approach to clean audio, combined with methods to isolate sounds that are important to the narrative and meaning of a broadcast has the potential to enable users to have complete control of the relative levels of all aspects of audio from TV broadcast. Any of the solutions previously discussed, such as non-speech channel attenuation, dynamic range control and other processing could then be carried out on only the speech, or other important content of the programme, in isolation depending on user preferences and on the nature and genre of the programme material. Although the FascinatE implementation described here was for live event broadcast, object based methods could be applied to any genre - live sports probably being the most challenging because of the difficulties in extracting audio objects.

The main limitation of the system described was that of latency. In the FascinatE project there was sufficient video latency as a result of the stitching together of 6 camera feeds into a 7K panorama so that all processing could take place and the resultant stream of audio objects, together with positional metadata could be streamed alongside the panoramic video stream to the end user. There are substantial challenges associated with adapting the technique for real time implementation in today’s broadcast systems, namely a much reduced video processing latency of around 20ms.

This technique for provision of accessible audio was demonstrated using recorded microphone feeds at the FascinatE project’s final demonstration at the University of Salford’s campus at MediaCityUK on 30th May 2013. During the demonstration visitors were able to alter the balance between foreground and background audio where the foreground audio consisted of commentary and on-pitch sounds and the background consisted of crowd noise.

8. Conclusions and Further Work

8.1. Conclusions

This thesis set out to investigate and to identify potential solutions to the problems that hearing impaired people face in understanding speech on television. Specifically the research set out to look at what solutions could be derived from the roll out of digital TV and the parallel increase in broadcast of 5.1 surround sound with its associated metadata. The initial work documented in chapter 3 looked into the possibilities introduced by the common mixing paradigm of utilising the centre channel of 5.1 primarily for speech and assessed the impact of reducing channels that did not carry speech content in the mix for both hearing impaired people and also non-hearing impaired people sharing a TV with them. This initial study indicated significant benefits to perceived speech clarity, overall sound quality and enjoyment for hearing impaired people for material that utilised the centre channel for speech. The outputs from this were presented to international broadcast standards bodies by the author and led to the formation of the UKCAF which presented recommendations to other international standards organisations. These recommendations were published in the ETSI standard for Digital Video Broadcast, referenced as recommendations in ITU standards for IPTV and have been adopted as mandatory requirements by some national broadcast bodies in Europe. The recommendations are based on the capability of STBs to generate a ‘hearing impaired’ mix from broadcast 5.1 media and discussions with STB manufacturers as part of UKCAF activity confirmed that modern devices were certainly capable of the limited processing that would be required to remix existing audio channels.

An additional benefit to deriving a hearing impaired mix at the STB rather than at the production end of the broadcast chain is that there is the potential for a HI output based on, and customised for, an individual user's preferences and needs. Methods were identified in collaboration with Dolby Labs to implement broadcast metadata in order to identify when the recommended 'clean audio' mix would be appropriate so that the accessible audio mix could be generated at the STB without the overhead of additional broadcast bandwidth for a dedicated hearing impaired audio channel. The *encinfo* bit in Dolby digital metadata was identified by Dolby as being available and unlikely to cause issues for legacy equipment, always a key requirement for any change in the usage of metadata. Further investigations would be required from Dolby in order to confirm this however and as yet these have not taken place partly owing to a refocusing on more extensive processing techniques at the STB and the need for extensive testing with legacy devices.

Although the identified solution was effective for hearing impaired people the problem remained that people with a hearing impairment were the least likely to invest in surround sound reproduction equipment so the people who could benefit most would potentially be those excluded from its benefits. The impact of a clean audio mix for people with stereo reproduction equipment listening to audio downmixed from 5.1 was investigated in chapter 4 and the recommended method for three loudspeaker reproduction was found to be compatible with down mixed stereo and to still provide clarity benefits for hearing impaired people and to improve the overall sound quality and enjoyment of media for this group.

Although a downmixed version of the clean audio condition was shown to be still useful in providing increased speech clarity the rating of downmixed Lt/Rt stereo was consistently worse than for all conditions utilising the three front loudspeakers of a 5.1 system. Poor ratings for default down mixed stereo in these tests have been investigated using subjective testing based on keyword recognition to assess intelligibility in chapter 5. Significantly poorer word recognition was found to occur where speech was presented as a phantom centre image rather than over a discrete centre loudspeaker. Further investigation showed that this was the result of acoustical crosstalk which was shown to have a significant negative impact on intelligibility of material based on keyword recognition. The impact of crosstalk was investigated at several listening positions and the resulting transfer functions showed great variability depending on listener position but substantial dips in frequency response at frequencies important to speech at the ideal listening position.

Further work investigated the potential use of principal component analysis to separate speech from competing noise at the STB. The technique, proposed by a paper in response to the Clean Audio Project findings, was replicated here in order to assess its performance when applied to a number of fairly common surround sound mixing practices. Although the process was found to be unsuitable for the proposed purpose additional development work has been carried out that identified a modified technique that could be effective as a pre-processing tool to identify the panned location of speech content in broadcast media. Chapter 6 of this thesis proposes a scenario where the modified speech biased PCA algorithm would be applied to broadcast material after post production and used to set, or unset, the *encinfo* bit of the Dolby broadcast metadata. Where speech was in centre channel only *encinfo* would be set, the STB would recognise the media as clean audio compatible and would attenuate non-speech channels as required based on the viewers user preferences if a Hearing Impaired (HI) output was selected by the user. Where speech was found in other channels (most typically in left, centre and right, or between centre and either left or right) or where no speech was present (for example in music programming) *encinfo* would be unset. The STB would then recognise the media as incompatible with, or inappropriate for, clean audio processing and would not apply channel attenuation regardless of whether an HI output was selected or not. Thus only a combination of *encinfo* set and HI output selected would lead to a clean audio output from the STB.

In chapter 7 two further implementations of the research documented in this thesis, in addition to adoption by international broadcast standards organisations, are discussed. The first of these, an experimental implementation of research recommendations plus additional processing from Dolby Labs, was assessed by the author using novel methods for media selection. Results from this study were inconclusive, it is concluded from the experience of these experiments that additional testing for media selection introduced additional variables that had an adverse effect on test results and that in test design, simpler is better.

The second future implementation of clean audio was developed as part of the author's work on the EU FP7 project, FascinatE. The FascinatE project developed a complete end-to-end future broadcast system which utilised object based audio broadcast. An object based approach allowed clean audio outputs to be readily provided at the viewer end of the broadcast chain and for the content of the audio output to be customised based on user preferences. The potential for object based audio in generating clean audio has been discussed in the context of *produced* media and an implementation developed for live broadcast. In the context of clean audio the parallel was drawn between diegetic and non-diegetic sounds in cinema, three sound categories were identified; speech content whose comprehension is critical, background noise that is detrimental to clarity, sound quality and enjoyment, and other non-speech sounds that can be considered important to comprehension and/or enjoyment and sound quality. In the object based audio system developed for live broadcast each of these categories were dealt with independently at the reproduction side with both speech, and non-speech factors considered important to comprehension mixed separately to other background sound or noise that was considered detrimental. The complete system, including clean audio components, was demonstrated in May 2013 in a live public event attended by broadcasters, producers, technologists and researchers.

During the course of this research guidelines for people with hearing impairments have been published by Ofcom, broadcasting standards have been published and are beginning to be implemented across Europe. It would be interesting to consider what impact the research documented here has had on the experience of hearing impaired people, if there is a 'Shirley effect'. Unfortunately data to base any such conclusion on is scarce. Action on Hearing Loss commission an annual questionnaire to its members however questions about the impact of background noise on television sound have only appeared in 2005 (RNID, 2005) and 2008 (RNID, 2008) surveys. Only one large scale study has been carried out in the UK (by the BBC and Voice of the Listener and Viewer (Menneer, 2011)) and no follow up is planned. Although the outcomes from the research presented here has now been embedded into broadcast standards it takes several years for such standards to progress from draft to final version. For this reason it is not yet possible to attribute real-world success in improving the experience of hearing impaired people at this time.

8.2. Further Work

Further work has been identified that could extend this research area as follows:

Research is needed over a period of several years as clean audio implementations are rolled out in order to assess the impact of these recommendations, guidelines and standards. Some limited studies have been carried out but typically as one off surveys. Work is needed to continue these surveys and to collate data indicating the number of complaints received by broadcasters each year so that possible correlations between standards implementations and other clean audio solutions with complaints can be more clearly understood.

Additional research will be required in order to further assess the impact of clean audio recommendations for non-hearing impaired people who may share a TV with hearing impaired relatives. The limited number of non-hearing impaired test participants in research carried out in chapter 3 meant that limited significance was drawn from the impact of the recommendations documented here for the perceived overall sound quality and enjoyment of this group. Additional testing with a larger subject group could help to identify a compromise position that maximised clarity, quality and enjoyment for hearing impaired people whilst providing the least negative impact for non-hearing impaired people.

Further work is required to implement broadcast metadata as suggested in this thesis. This would require collaboration; firstly with the broadcast industry in identifying any legacy issues that may result from appropriation of existing metadata, and secondly with set top box manufacturers in programming switchable clean audio functionality into end user devices.

Further work is needed to assess PCA as a pre-broadcast tool which could identify the location of speech in 5.1 TV programming and set or unset a *clean audio* bit in the metadata to indicate when a clean audio mix would be appropriate for the programme content. This should include comparisons with existing speech recognition algorithms for a range of 5.1 broadcast mixes.

Currently object-based audio is only utilised commercially for AV media in cinema applications in the Dolby Atmos system (Robinson et al., 2012) which was commercially released to the public in 2012. A parallel development for TV broadcast would have to be implemented in order for object based clean audio to be realised in a broadcast domain. Thus far this is the subject of some commercial research however it has not yet been implemented other than in a research context.

8.3. The Future for Accessible TV Audio

As requirements for accessible TV audio are rolled out into broadcast standards documents and as broadcasters and the broadcast technology industry identifies implementation methods a range of clean audio techniques could be utilised. Implementation of the metadata tagged clean audio process identified here is currently recommended as an appropriate minimum requirement for clean audio in ETSI requirements (ETSI, 2009) and referenced in several more standards documents. This means that it is likely to be the earliest adopted solution to the problems identified here. The increased use of second screen devices and IPTV makes transmission of a parallel audio stream delivered over IP feasible however the problem still remains as to how IP delivered clean audio is produced. Production of a separate mix for hearing impaired viewers may be the least likely outcome because of additional expense to the broadcast industry and an approach such as that adopted in chapter 3 could be more efficiently delivered via the aforementioned metadata solution. Object based audio is the solution that would provide the most flexible and effective system however a new approach to broadcast audio must be adopted for this to occur. This now seems more likely; already significant technological research is underway with broadcasters, manufacturers and universities all looking at object based audio as a solution to spatial and interactive audio broadcast systems generally as well as for accessible audio and this seems very likely to continue.

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APPENDIX A Dolby Digital Metadata Parameters

(Parameters in italics are part of the Extended Bit Stream Information)

Program Configuration

Program Description Text

Dialogue Level

Channel Mode

LFE Channel

Bitstream Mode

Line Mode Compression

RF Mode Compression

RF Overmodulation Protection

Centre Downmix Level

Surround Downmix Level

Dolby Surround Mode

Audio Production Information

Mix Level

Room Type

Copyright Bit

Original Bitstream

Preferred Stereo Downmix

Lt/Rt Centre Downmix Level

Lt/Rt Surround Downmix Level

Lo/Ro Centre Downmix Level

Lo/Ro Surround Downmix Level

Dolby Surround EX Mode

A/D Converter Type

DC Filter

Lowpass Filter

LFE Lowpass Filter

Surround 3 dB Attenuation

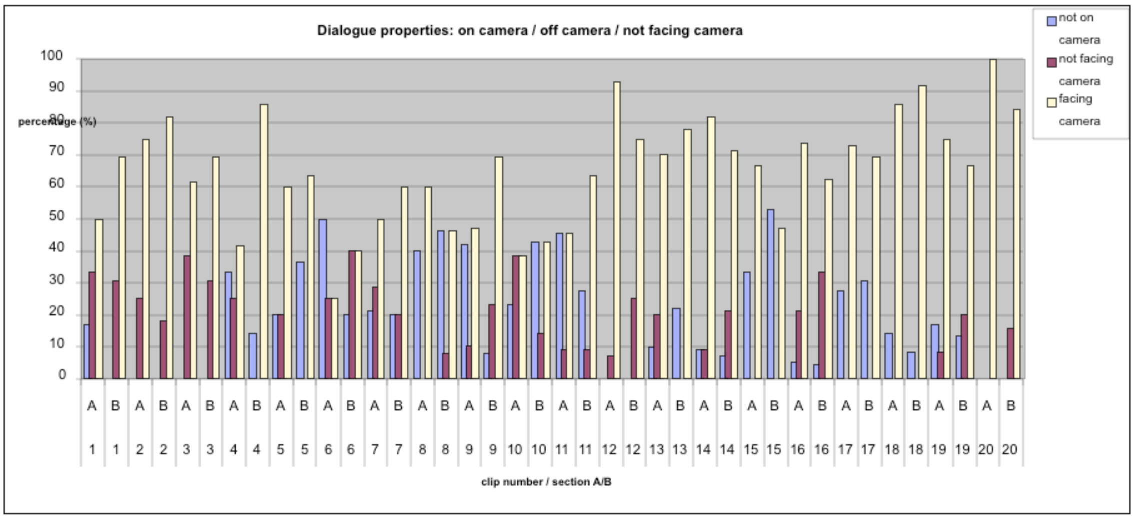
Surround Phase Shift

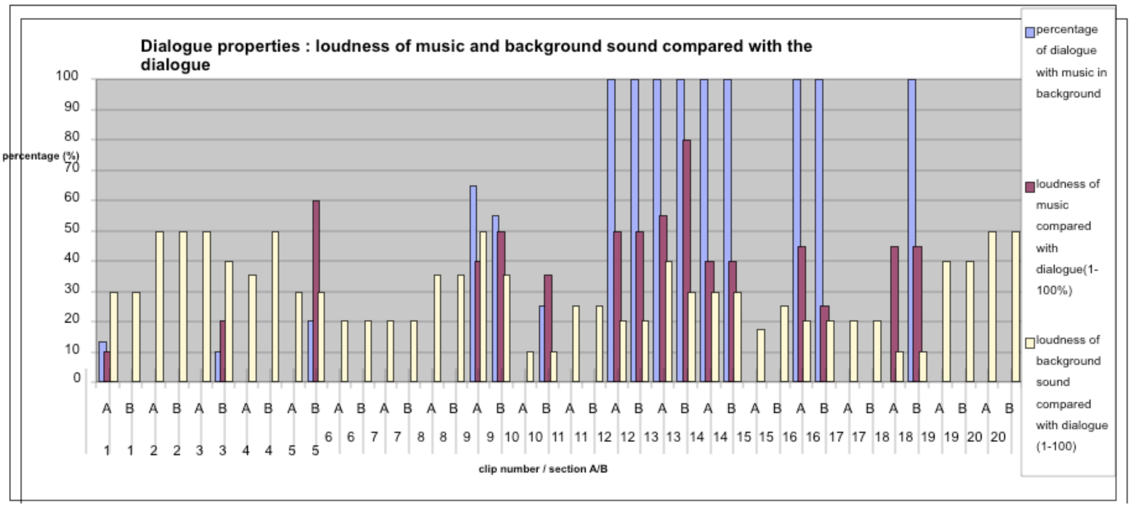
Reproduced from *Dolby Metadata Guide vol 2* published by Dolby Labs (Dolby Labs, 2003).

APPENDIX B Example Test Material

Clip number	Film	Start	End
1	Chocolat	1:10:25	1:11:50
2	Air force one	00:40:04	00:41:09
3	Air force one	01:13:28	01:14:50
4	Air force one	01:21:47	01:22:50
5	Gosford park	00:16:04	00:17:21
6	Gosford park	00:19:22	00:20:29
7	Gosford park	01:32:51	01:33:55
8	Gosford park	01:45:45	01:47:14
9	Gosford park	00:34:10	00:35:31
10	Gosford park	00:22:12	00:23:23
11	Gosford park	01:26:20	01:27:21
12	Gladiator	01:53:19	01:54:37
13	Gladiator	01:26:50	01:28:03
14	LA confidential	00:21:44	00:22:58
15	Devils advocate	00:34:40	00:30:10
16	Negotiator	01:00:51	01:02:12
17	Negotiator	01:30:27	01:31:32
18	Chocolat	00:40:30	00:41:47
19	Chocolat	00:52:11	00:53:24
20	Green Mile	00:07:35	00:08:58

APPENDIX C Analysis of Test Material





APPENDIX D Subject Questionnaire for Clean Audio Testing

OFCOM Clean Audio Research

Introduction

We want to find out about the sound quality of video clips. We are doing this to help manufacturers improve the sound for viewers who are hard of hearing.

Personal Details

In this section you will be asked a series of questions about yourself, all of this information will be treated in the strictest confidence. No one outside the research team at Salford University will see your personal details.

Name ____

Address ____

____

____

Tel: __________________

Email: __________________________

Age: (Please tick one box) **Under 30** □ **30-44** □ **45-59** □ **60-74** □ **75+** □

Nature of hearing impairment (write ***none*** if have no hearing impairment):

____

____

____

Gender: **M** □ **F** □

There are a series of questions on the following pages. Each question asks you to compare 2 sections of a video clip.

After watching each clip please tick the box to indicate your preference and rate the difference between the sections by making a mark on the line as shown. There are no right or wrong answers.

Please tick one box and make a mark on the line as in the example below.

Example

Which section do you think had the best sound quality.

Section A □ **Section B** ☑

How much better was your choice?

Slightly Better |←――――――――――/―――――→| Much Better

Clip x

a) Which section do you think had the best overall sound quality?

Section A □ **Section B** □

How much better was your choice?

Slightly Better |←————————→| Much Better

b) Which section did you enjoy the most?

Section A □ **Section B** □

How much more did you enjoy it?

Slightly More |←————————→| Much More

c) In which section was the speech clearer?

Section A □ **Section B** □

How much clearer was your choice?

Slightly Clearer |←————————→| Much Clearer

Do you have any comments or suggestions?

This concludes this questionnaire, thank you very much for taking the time to help us in our research; your help is very much appreciated.

APPENDIX E Consent Form and Information Sheet

We would be very grateful if you could help us in an important research project about the quality of sound on television.

The University of Salford Acoustics Research Centre is carrying out a series of tests on how to improve the quality of sound from television. Of course you are under no obligation and do not have to participate, but it would be extremely valuable if you could assist us in this study.

The work is sponsored by the OFCOM (formerly the Independent Television Commission (ITC)) and your responses to the tests will help us develop better sound for television in the future.

We need your consent to:

- Retain some background information on you (name, age, gender, contact details);
- Carry out a hearing test and retain an audiogram showing your hearing ability.
- Carry out a series of tests where you listen to speech and other TV programme content and we ask a series of questions intended to assess how well you have heard and enjoyed the recordings. We need your permission to retain the results.

All information will be kept confidential. The work will be used to help improve the quality of TV sound. No individuals will be identified in the results of the research. As some of this data is held on computer, some is covered by the data protection act, and you will be able to see a copy of it on request.

Both hearing and hard of hearing people are required for the tests though we are particularly interested in contacting hard of hearing people who may wish to participate. If you would be willing to participate or if you know of anyone else who may be interested my contact details are as follows:

Ben Shirley
Lecturer
Acoustics Research Centre
University of Salford
0161 2954524
b.g.shirley@salford.ac.uk
www.acoustics.salford.ac.uk

Consent Form

I have read and understood the information sheet and this consent form. I understand that participation is voluntary and I am free to withdraw at any time.

Name: ________________________________

Signature: _____________________________

Date: _________________________________

APPENDIX F Testing script

1) Thank you for taking part in our test today. The test will consist of a series of clips from films.

2) Each clip will be split up into two sections, section A and section B.

3) You will then be shown a clip and after the clip has finished (at the end of section B) asked to answer three questions about that clip.

4) *These questions will be;*

   a) Which section had the better overall sound quality? Tick the appropriate box and put a mark on the line stating how much better you thought the sound quality was.

   b) Which section did you enjoy the most? Tick the appropriate box and put a mark on the line indicating how much more you enjoyed the better clip.

   c) In which section was the speech clearer? Tick the appropriate box and put a mark on the line indicating how much clearer.

6) At the end of each clip the video will be paused until you have finished filling in the three questions for that clip. When you have finished filling in the questions please let me know by saying finished and the next clip will be played.

7) If you would like to see the clip again that is fine just ask me if you could see the clip again and I will play both sections again.

8) I will keep silent during the test so you are not distracted.

APPENDIX G Classification of Hearing Loss

Hearing loss was categorised using a pure tone audiogram with the hearing level threshold levels averaged at the following frequencies:

250, 500, 1000, 2000 and 4000 Hz

Audiometric descriptor of loss	*dB HL*
None	< 20
Mild	20 – 40
Moderate	41 – 70
Severe	71 – 95
Profound	> 95

Reproduced from;
The British journal of Audiology, 1988, 22, 123, ***Descriptors for pure-tone audiograms***

To determine whether a subject has asymmetric hearing loss, the definition used was that there is a difference of 30dB or more at two or more frequencies. This is the definition as per the TTSA (Technicians, Therapists and Scientists in Audiology) guidelines for direct referrals for Hearing Aids.

APPENDIX H Data Analysis Methods Example

In the analysis of the data the use of the scales was examined for each subject, and a measurement was made in mm along the length of the scale. The maximum (x_{max}) and minimum (x_{min}) values recorded on each scale for each subject were used to normalise the rest of the data using the following formula:

$$normalised\ value_{subject\ i} = \frac{(x - x_{\min})}{(x_{\max} - x_{\min})}$$

An example of pre-normalised and post-normalised data for one subject on one scale is shown in table 4.

Clip number	Section Preferred	Process selected	Process not selected	Scale	Normalised scale
1	B	Centre	LR	81	0.89
2	B	LR6dB	LR	57	0.52
3	B	DRC	LR	57	0.52
4	B	DRC6dB	LR	62	0.59
5	B	LR6dB	Centre	80	0.88
6	B	DRC	Centre	42	0.28
7	A	Centre	DRC6dB	32	0.13
8	A	LR6dB	DRC	24	0.00
9	A	LR6dB	DRC6dB	75	0.80
10	B	LR	Centre	32	0.13
11	A	LR6dB	LR	66	0.66
12	B	LR	DRC	24	0.00
13	A	DRC6DB	LR	75	0.80
14	B	Centre	LR6dB	88	1.00
15	B	Centre	DRC	28	0.06
16	B	Centre	DRC6dB	34	0.16
17	B	LR6dB	DRC	32	0.13
18	B	LR6dB	DRC6dB	79	0.86
19	B	DRC6dB	DRC	81	0.89
20	B	DRC	DRC6dB	77	0.83
			Min	24	
			Max	88	

An example of pre-normalised and post-normalised data for one subject on the sound quality scale

APPENDIX I MatLab Code for Principal Component Analysis

PCA_single _vector_recon_mix.m

```
%   PCA, Principal Component only
%   This reconstructs spin1.wav from only it's principal component

samples = wavread('spin1.wav')      % get data
ave = mean(samples)                 % mean of l & r columns
[m,n] = size(samples)               % how many samps?
meanmat = repmat(ave,m,1)               % expand averages matrix so can
                                    % subtract from samp values
normsamps = samples - meanmat
covariance = cov(normsamps)                 % find covariance matrix
[vect,val] = eig(covariance)                % get eigenvectors and eigenvalues
                                    % for covariance matrix
% d = eig(cov)                      % not using this but puts values
                                    % into same format as tutorial
% need to delete less important vectors at this point
% eg. Which is largest n vectors? Delete rest!
vect(:,1) = []
%   Generate new data set
FinalData = vect' * normsamps'
RowOrigData = vect * FinalData
RowOrigData = RowOrigData + meanmat'    % add means in again

wavwrite(RowOrigData', 44100, 'Recon.wav')  % write reconstructed wav
```

Matlab code used for accessing 5 channel audio as multiple mono wav files, in this example all components other than the principal component are deleted before reconstruction.

PCA_5_mono_wavs.m

```
samples = wavread('1_FL.wav')
samples(:,2) = wavread('2_FR.wav')
samples(:,3) = wavread('3_C.wav')
samples(:,4) = wavread('4_LS.wav')
samples(:,5) = wavread('5_RS.wav')
ave = mean(samples);                % mean of channel columns
[m,n] = size(samples)               % how many samps?
%   m = numsamps and n = numchannels
meanmat = repmat(ave,m,1);          % expand averages matrix so can
% subtract from samp values
normsamps = samples - meanmat;
covariance = cov(normsamps);        % find covariance matrix
[vect,val] = eig(covariance)        % get eigenvectors and
                                    %     eigenvalues
% for covariance matrix
% need to delete less important vectors at this point.
% eg. Which is largest n vectors? Delete rest!
[C,I] = max(val)
[x,y] = max(C)
%   y now contains the column number of the HIGHEST value ( of the
%    principal component)
%   The rest should then be deleted
if y == 1
    vect(:,5) = []
    vect(:,4) = []
    vect(:,3) = []
    vect(:,2) = []
elseif y == 2
    vect(:,5) = []
    vect(:,4) = []
    vect(:,3) = []
    vect(:,1) = []
elseif y == 3
    vect(:,5) = []
    vect(:,4) = []
    vect(:,2) = []
    vect(:,1) = []
elseif y == 4
    vect(:,5) = []
    vect(:,3) = []
    vect(:,2) = []
    vect(:,1) = []
elseif y == 5
    vect(:,4) = []
    vect(:,3) = []
    vect(:,2) = []
    vect(:,1) = []
end
%   Generate new data set
FinalData = vect' * normsamps';
RowOrigData = vect * FinalData;
RowOrigData = RowOrigData + meanmat';    % add means in again
ReconData = RowOrigData'
wavwrite(ReconData(:,1),48000,'out1.wav')
wavwrite(ReconData(:,2),48000,'out2.wav')
wavwrite(ReconData(:,3),48000,'out3.wav')
wavwrite(ReconData(:,4),48000,'out4.wav')
wavwrite(ReconData(:,5),48000,'out5.wav')
```

Matlab code used for processing 5 channel audio from multiple mono wav files using a weighted PCA unmixing matrix based on a speech frequency filter, in this example all components other than the principal component are deleted before reconstruction.

PCA_5_mono1_ref.m

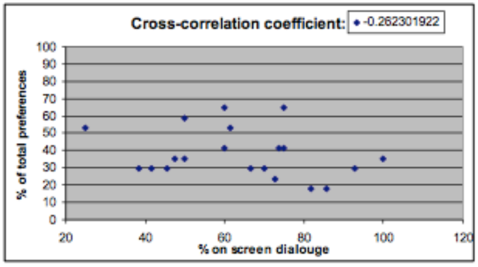
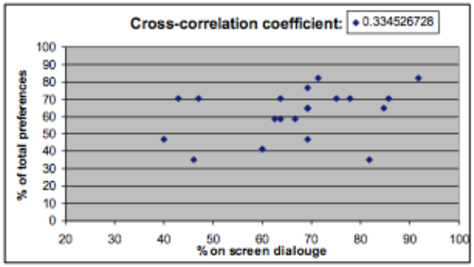
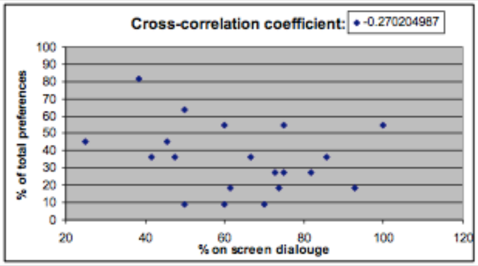
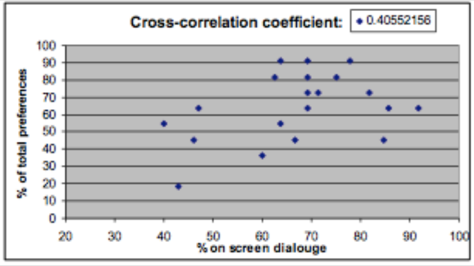
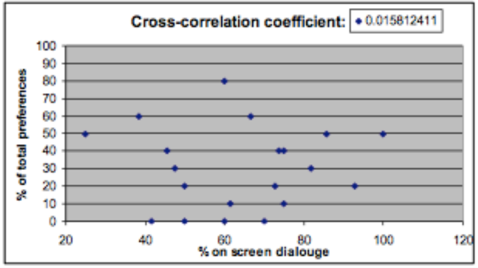
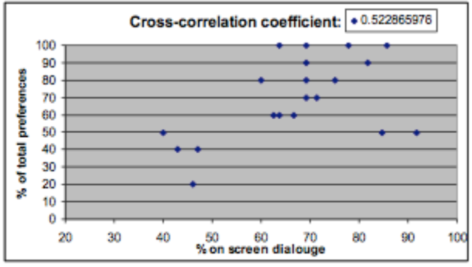
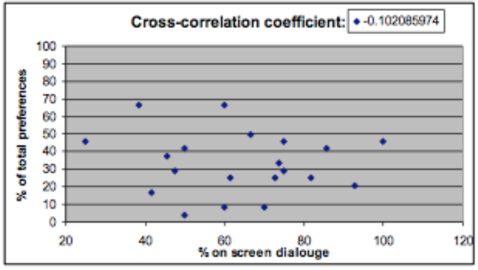
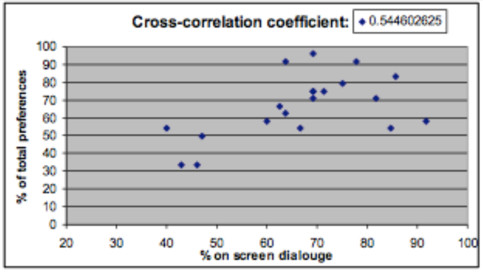
```
%-- 16/03/06 10:05 --%
% PCA_5_mono1_ref.m %
% carries out weighted PCA on 5 channels of audio %
% based on speech filter %
% reference for PCA path
refsamples = wavread('ref1.wav')
refsamples(:,2) = wavread('ref2.wav')
refsamples(:,3) = wavread('ref3.wav')
refsamples(:,4) = wavread('ref4.wav')
refsamples(:,5) = wavread('ref5.wav')

% samples to carry out PCA on
samples = wavread('1.wav')
samples(:,2) = wavread('2.wav')
samples(:,3) = wavread('3.wav')
samples(:,4) = wavread('4.wav')
samples(:,5) = wavread('5.wav')

%   get filter
BPF = speechFltr;

% PCA for reference parallel path
refave = mean(refsamples);                     % mean of channel
columns
[o,p] = size(refsamples)                    % how many samps?
%   o = numsamps and p = numchannels
refmeanmat = repmat(refave,o,1);               % expand averages matrix
so can
% subtract from samp values
refnormsamps = refsamples - refmeanmat;
BPrefnormsamps = filter(BPF, refnormsamps);
refcovariance = cov(BPrefnormsamps);             % find covariance
matrix
[refvect,refval] = eig(refcovariance)            % get eigenvectors
and %eigenvalues
% for covariance matrix
[D,J] = max(refval)
[a,b] = max(D)

% PCA for actual samples based on unmixing matrix from filtered
samples
ave = mean(samples);                     % mean of channel columns
[m,n] = size(samples)                    % how many samps?
%   m = numsamps and n = numchannels
meanmat = repmat(ave,m,1);               % expand averages matrix so
can
% subtract from samp values
normsamps = samples - meanmat;
covariance = cov(normsamps);             % find covariance matrix
[vect,val] = eig(covariance)             % get eigenvectors and
eigenvalues
% for covariance matrix
% need to delete less important vectors at this point.
% eg. Which is largest n vectors? Delete rest!
[C,I] = max(val)
```

```
[x,y] = max(C)
%   y now contains the column number of the HIGHEST value ( of the %
principal component)
%   The rest should then be deleted
if b == 1
vect(:,5) = []
vect(:,4) = []
vect(:,3) = []
vect(:,2) = []
elseif b == 2
vect(:,5) = []
vect(:,4) = []
vect(:,3) = []
vect(:,1) = []
elseif b == 3
vect(:,5) = []
vect(:,4) = []
vect(:,2) = []
vect(:,1) = []
elseif b == 4
vect(:,5) = []
vect(:,3) = []
vect(:,2) = []
vect(:,1) = []
elseif b == 5
vect(:,4) = []
vect(:,3) = []
vect(:,2) = []
vect(:,1) = []
end
%   Generate new data set
FinalData = vect' * normsamps';
RowOrigData = vect * FinalData;
RowOrigData = RowOrigData + meanmat';     % add means in again
ReconData = RowOrigData'
wavwrite(ReconData(:,1),48000,'out1.wav')
wavwrite(ReconData(:,2),48000,'out2.wav')
wavwrite(ReconData(:,3),48000,'out3.wav')
wavwrite(ReconData(:,4),48000,'out4.wav')
wavwrite(ReconData(:,5),48000,'out5.wav')
```

Matlab code implementing a Hanning window envelope to allow dynamic PCA. Windows are set at 500ms with 50% overlap to ensure unity gain. Variable ‘*atten*’ sets factor for gain/attenuation of non-principal components, atten = 0 removing or muting all non-principal conponents.

```
%-- 23/03/06 11:55 --%
%============================================================
%           PCA ON 500MS BLOCKS OF SAMPLES
%           FROM RAW PCM AUDIO FILES
%           48khZ, 16 BIT
%                               Ben Shirley
%   DELETES OUTPUT FILES FIRST BEFORE RUNNING
%============================================================
sampsPerWindow = 24000;  % define for ease of alteration
%   - 500ms block = 24000 @ 48KHz
atten = 0;              %  attenuation factor (<1 please!)
                        %   0 = mute, 1 = no atten
%   open pcm files for reading
fid1=fopen('1.pcm','r');
fid2=fopen('2.pcm','r');
fid3=fopen('3.pcm','r');
fid4=fopen('4.pcm','r');
fid5=fopen('5.pcm','r');

%   open output files then close them to delete existing data
fido1 = fopen('out1.pcm', 'w');
fido2 = fopen('out2.pcm', 'w');
fido3 = fopen('out3.pcm', 'w');
fido4 = fopen('out4.pcm', 'w');
fido5 = fopen('out5.pcm', 'w');
fclose(fido1);
fclose(fido2);
fclose(fido3);
fclose(fido4);
fclose(fido5);

%   open output files for writing
fido1 = fopen('out1.pcm', 'r+');
fido2 = fopen('out2.pcm', 'r+');
fido3 = fopen('out3.pcm', 'r+');
fido4 = fopen('out4.pcm', 'r+');
fido5 = fopen('out5.pcm', 'r+');

%   get size of file based on channel 1
status = fseek(fid1, 0, 'eof');
sizbytes = ftell(fid1);              %   number of bytes
numsamps = sizbytes/2;    %   number of 16 bit samples
numread = 0;
firsttime = 1;
numtimes = 1;
status = fseek(fid1, 0, 'bof');
%   =========HERE'S THE MAIN CODE=============
%   TODO change so not based on numread - some of these

while numread < numsamps(1,1)
readposn = ftell(fid1);              %   number of samps into file
    [samples, count1] = fread(fid1,sampsPerWindow,'short');
    [samples(:,2), count2] = fread(fid2,sampsPerWindow,'short');
    [samples(:,3), count3] = fread(fid3,sampsPerWindow,'short');
    [samples(:,4), count4] = fread(fid4,sampsPerWindow,'short');
    [samples(:,5), count5] = fread(fid5,sampsPerWindow,'short');
    numread = numread + count1;   %   Total samples read
readposn = ftell(fid1);              %   number of samps into file

    %   if count < full window size, fill extra with 0s
    if count1 < sampsPerWindow
        samples(count1:sampsPerWindow, 1) = 0;
        samples(count2:sampsPerWindow, 2) = 0;
        samples(count3:sampsPerWindow, 3) = 0;
        samples(count4:sampsPerWindow, 4) = 0;
        samples(count5:sampsPerWindow, 5) = 0;
    end


    %============Here's the PCA code===========
    ave = mean(samples);                    % mean of channel columns
    [m,n] = size(samples);                   % how many samps?
    %   m = numsamps and n = numchannels
    meanmat = repmat(ave,m,1);              % expand averages matrix
so can
    % subtract from samp values to normalise
    normsamps = samples - meanmat;
    covariance = cov(normsamps);            % find covariance matrix
    [vect,val] = eig(covariance);            % get eigenvectors and
eigenvalues
    % for covariance matrix
    % need to attenuate or delete less important vectors at this
point.
    % eg. Which is largest n vectors? Attenuate rest!
    [C,I] = max(val);
    [x,y] = max(C);
    PCarray(numtimes,1) = y;    %   store principal component number
for THIS window - debugging
    %   y now contains the column number of the HIGHEST value ( of the
%    principal component)
    %   The rest should then be attenuated.
    if y == 1
        vect(:,5) = vect(:,5) * atten;
        vect(:,4) = vect(:,4) * atten;
        vect(:,3) = vect(:,3) * atten;
        vect(:,2) = vect(:,2) * atten;
    elseif y == 2
        vect(:,5) = vect(:,5) * atten;
        vect(:,4) = vect(:,4) * atten;
        vect(:,3) = vect(:,3) * atten;
        vect(:,1) = vect(:,1) * atten;
    elseif y == 3
        vect(:,5) = vect(:,5) * atten;
        vect(:,4) = vect(:,4) * atten;
        vect(:,2) = vect(:,2) * atten;
        vect(:,1) = vect(:,1) * atten;
    elseif y == 4
        vect(:,5) = vect(:,5) * atten;
        vect(:,3) = vect(:,3) * atten;
        vect(:,2) = vect(:,2) * atten;
        vect(:,1) = vect(:,1) * atten;
    elseif y == 5
        vect(:,4) = vect(:,4) * atten;
        vect(:,3) = vect(:,3) * atten;
        vect(:,2) = vect(:,2) * atten;
        vect(:,1) = vect(:,1) * atten;
    else
        vect(:,5) = vect(:,5) * atten;
        vect(:,4) = vect(:,4) * atten;
        vect(:,3) = vect(:,3) * atten;
        vect(:,2) = vect(:,2) * atten;
        vect(:,1) = vect(:,1) * atten;
    end

    %   Generate new data set
    FinalData = vect' * normsamps';
    RowOrigData = vect * FinalData;
    RowOrigData = RowOrigData + meanmat';      % add means in again
    ReconData = RowOrigData';
%================END OF PCA CODE================

    %   Windowing function....
    w = hann(size(ReconData,1));
    ReconData(:,1) = ReconData(:,1) .* w;
    ReconData(:,2) = ReconData(:,2) .* w;
    ReconData(:,3) = ReconData(:,3) .* w;
    ReconData(:,4) = ReconData(:,4) .* w;
    ReconData(:,5) = ReconData(:,5) .* w;

    %    seek to a bit before the last end of write so can do window
overlap
    if firsttime == 0
        fseek(fido1, - (sampsPerWindow), 'eof');   %   (*2 for samps
not bytes then /4 for last 1/4 Of last window written)
        fseek(fido2, - (sampsPerWindow), 'eof');
        fseek(fido3, - (sampsPerWindow), 'eof');
        fseek(fido4, - (sampsPerWindow), 'eof');
        fseek(fido5, - (sampsPerWindow), 'eof');
readposnop = ftell(fido1);                %   number of samps into file

        %   read end of last window into temp array
        [temp1, tmpcount] = fread(fido1, sampsPerWindow/2, 'short');
%   should this be /2 as well??? check tmpcount!!!
        temp2 = fread(fido2, sampsPerWindow, 'short');     %   or does
the 'short' argument negate this?
        temp3 = fread(fido3, sampsPerWindow, 'short');
        temp4 = fread(fido4, sampsPerWindow, 'short');
        temp5 = fread(fido5, sampsPerWindow, 'short');
readposnop = ftell(fido1);                %   number of samps into file
        %   temp1 to same length as ReconData
        %   Fill rest of array with 0s (*2 because this is
        %   BYTES not SAMPLES) - 16 bit files only
        temp1(sampsPerWindow/2:sampsPerWindow, 1) = 0; %
        temp2(sampsPerWindow/2:sampsPerWindow, 1) = 0;
        temp3(sampsPerWindow/2:sampsPerWindow, 1) = 0;
        temp4(sampsPerWindow/2:sampsPerWindow, 1) = 0;
        temp5(sampsPerWindow/2:sampsPerWindow, 1) = 0;
        ReconData(:,1) = ReconData(:,1)+ temp1;
        ReconData(:,2) = ReconData(:,2)+ temp2;
        ReconData(:,3) = ReconData(:,3)+ temp3;
        ReconData(:,4) = ReconData(:,4)+ temp4;
        ReconData(:,5) = ReconData(:,5)+ temp5;
        position = ftell(fido1);
        fseek(fido1, -(sampsPerWindow), 'eof');
        fseek(fido2, -(sampsPerWindow), 'eof');
        fseek(fido3, -(sampsPerWindow), 'eof');
        fseek(fido4, -(sampsPerWindow), 'eof');
        fseek(fido5, -(sampsPerWindow), 'eof');
readposnop = ftell(fido1);              %   number of samps into file

    end
    firsttime = 0;
    %   add to windowed array (overlap)
    %   write to output files
    fwrite(fido1, ReconData(:,1), 'short');
    fwrite(fido2, ReconData(:,2), 'short');
    fwrite(fido3, ReconData(:,3), 'short');
    fwrite(fido4, ReconData(:,4), 'short');
    fwrite(fido5, ReconData(:,5), 'short');
readposnop = ftell(fido1);              %   number of samps into file

    %   fseek back in input file for next overlapping window
    fseek(fid1, -(sampsPerWindow), 'cof');
    fseek(fid2, -(sampsPerWindow), 'cof');
    fseek(fid3, -(sampsPerWindow), 'cof');
    fseek(fid4, -(sampsPerWindow), 'cof');
    fseek(fid5, -(sampsPerWindow), 'cof');
    numread = ftell(fid1);              %   number of samps into file
    numtimes = numtimes + 1;
end
fclose('all')
```

APPENDIX J Correlation of speech clarity ratings against the percentage of on-screen dialogue for Clean Audio project

	Section A	Section B
Non-Impaired (17)	 Cross-correlation coefficient: -0.262301922 % of total preferences % on screen dialouge	 Cross-correlation coefficient: 0.334526728 % of total preferences % on screen dialouge
Mild (11)	 Cross-correlation coefficient: -0.270204987 % of total preferences % on screen dialouge	 Cross-correlation coefficient: 0.40552156 % of total preferences % on screen dialouge
Moderate (10)	 Cross-correlation coefficient: 0.015812411 % of total preferences % on screen dialouge	 Cross-correlation coefficient: 0.522865976 % of total preferences % on screen dialouge
Impaired (24)	 Cross-correlation coefficient: -0.102085974 % of total preferences % on screen dialouge	 Cross-correlation coefficient: 0.544602625 % of total preferences % on screen dialouge

APPENDIX K Liaison statement from the UK Clean Audio Forum to ITU Focus Group on IPTV

INTERNATIONAL TELECOMMUNICATION UNION **TELECOMMUNICATION STANDARDIZATION SECTOR** STUDY PERIOD 2005-2008	**FOCUS GROUP ON IPTV** **FG IPTV-IL-0039** **English only**
WG(s): 6	3rd FG IPTV meeting: Mountain View, USA, 22-26 January 2007

INCOMING LIAISON STATEMENT

Source:	UK Clean Audio Forum
Title:	Liaison Statement from UK Clean Audio Forum to ITU FG IPTV

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Introduction

An estimated 50 million people [1] in Europe find speech (dialogue and narrative) in television programmes difficult to follow. Reducing the background sound (music, effects and ambient sounds) can make the speech clearer thus providing 'Clean Audio'. Research at the University of Salford in the UK [2] has found that Clean Audio improves the clarity, quality and enjoyment for hard-of-hearing people and their families.

People with good hearing are able easily to more discern the speech from background, but this ability declines naturally with age. For people with even minor hearing loss or reduced cognitive ability, a high level of background audio information often leads to reduction in audibility or intelligibility of the speech; this often manifests itself before hearing loss would be clinically identified by traditional audiometric measures. This can sometimes lead to the viewer turning up the volume, but doing so will not necessarily improve intelligibility and may even exacerbate the problem.

Clean Audio

The UK Clean Audio Forum has defined Clean Audio as:

> "The provision of television sound in a manner that achieves clarity of speech (programme narrative and dialogue) for the maximum number of viewers, especially those with hearing loss."
>
> Various measures might be taken to achieve this.
>
> - *taking particular care in production and/or*
> - *providing specific mechanisms for delivery which allow the user to select and/or control a mix of speech and background sound (music, effects and ambient sounds) to suit his or her taste/capabilities, for example*
>   - *by transmitting metadata to generate a separate alternative mix in the receiver* [3] *or*
>   - *by providing a separate alternative mix*

The Forum is studying what practicable steps can be taken to offer such improvements and would welcome liaison with other bodies to share a wider understanding of the challenges and opportunities.

[1] Derived from Davis, A. *Hearing in Adults*, 1995, Whurr Publishers, London.

[2] Sponsored by the UK Communications Regulator OFCOM

[3] An example approach based on processing multi-channel audio in the television receiver is illustrated in annex A.

Annex A

An example approach to Clean Audio based on processing multi-channel audio in the television receiver

The introduction of multi-channel audio presents new opportunities to offer choice to to the viewer. In particular, the widespread availability of high-definition television programme material, delivered to the home from a variety of sources, will lead to multi-channel audio becoming more generally available in consumer equipment.

One potential approach to providing Clean Audio is therefore to process multi-channel audio in the television receiver so as to offer "cleaner" speech than is usually contained in conventionally delivered television sound. On selection, this could simply be played out through the existing loudspeakers in standard television sets.
The commonly employed approach to surround sound production for film has the front centre channel used to carry speech. As a direct result there is already a large (and so far untapped) archive of programme material suitable for delivering Clean Audio.

UK research has found that, for such material, a good combination of enjoyment and clarity of speech is obtained when the speech can be delivered in the front centre channel and the audio level of left and right channels is reduced by a modest amount.

To maximise the potential market, it is foreseen that a common approach should be adopted for the mechanisms used to provide assistive Clean Audio, whether for packaged media, or for broadcast or IP delivery.

The following commercial requirements are intended to cover the interests of end users, service providers and network operators.

1 General

- Technical specifications shall define optional (rather than mandatory) mechanisms.
- The mechanisms defined in those specifications shall not prejudice the enjoyment of television sound for those who wish to listen to television sound produced and delivered conventionally.
- The mechanisms in those specifications shall not prejudice the delivery of other assistive services such as Audio Description.
- The specifications for delivering Clean Audio and associated data shall be simple and coding-technology neutral, i.e. shall be capable of being employed in conjunction with any multichannel audio coding technology that is supported by DVB.
- The specifications shall be suitable for use on any platform that is employed for delivering multichannel audio, including packaged media, broadcast and IP delivery.
- End users should be able easily to select Clean Audio using a simple interface and (optionally) to control the mix of speech and background sound.

2 Signalling and metadata

- Technical specifications shall enable the receiver to identify the incoming multichannel audio as offering a Clean Audio service [4].

[4] For example signalling to support identification of the following possible Clean Audio implementations :
(a) as a separate audio stream
(b) the conventional television sound-track authored as Clean Audio
(c) Clean Audio derived dynamically using transmitted metadata
(d) Clean Audio derived automatically without use of metadata
(e) programme material not suitable for Clean Audio.

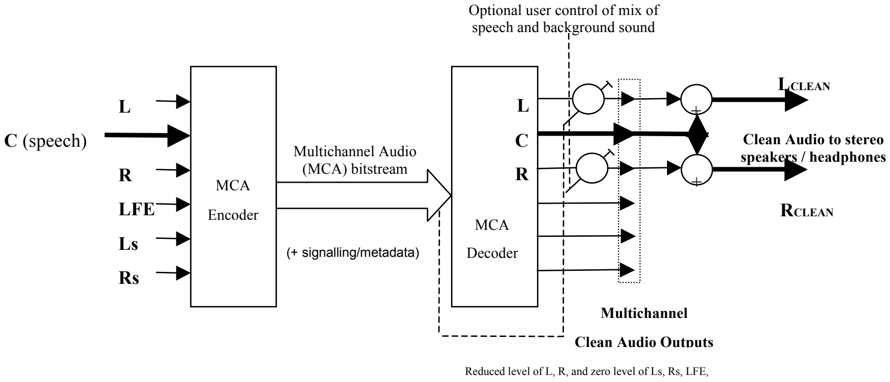
- Those specifications shall enable electronic programme guides to optionally include an indication of which programmes are available with Clean Audio.

3 Quality of Service

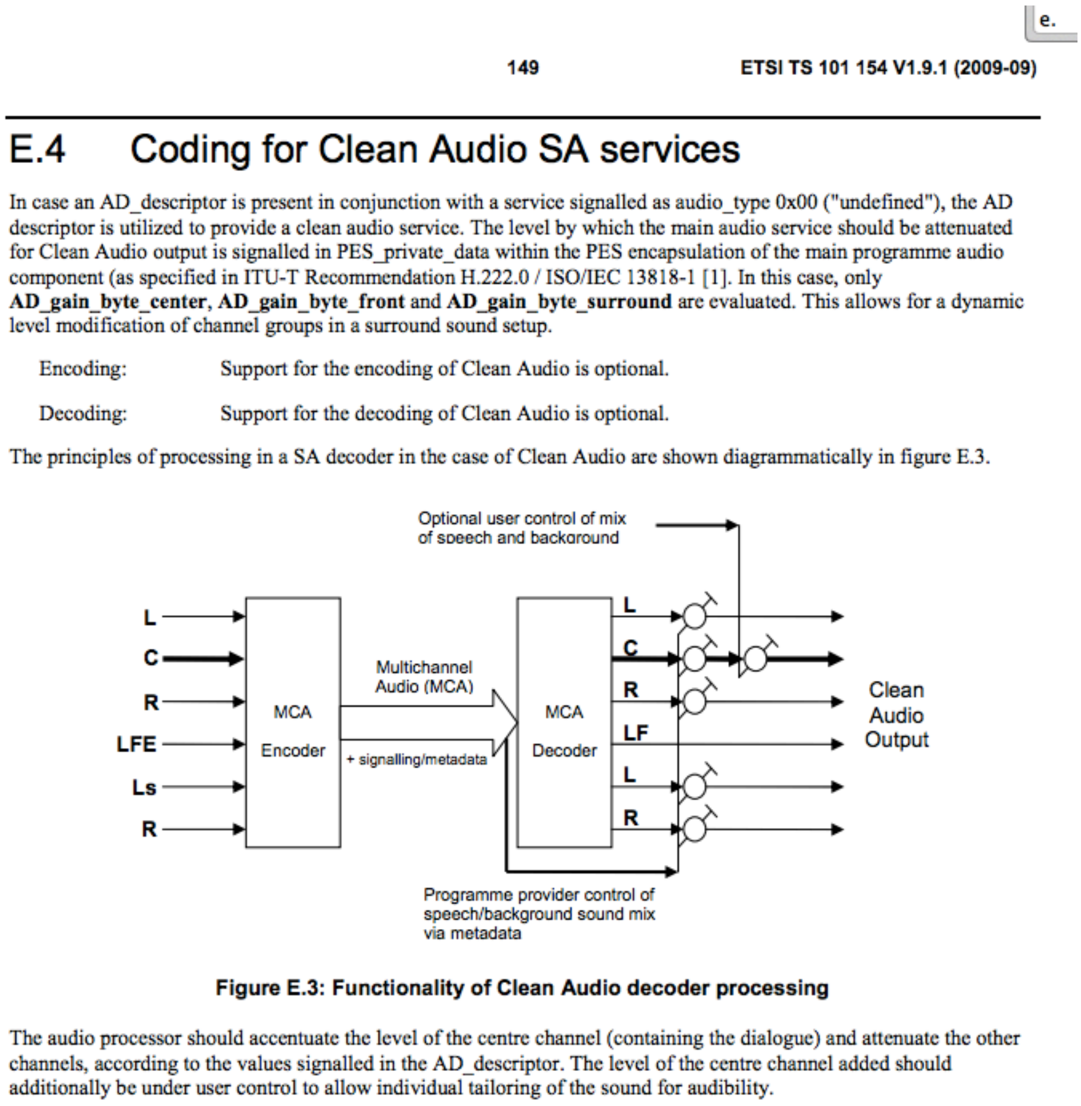
- The timing of the Clean Audio with respect to vision as delivered by the receiver should not differ from that of the multi-channel audio.

4 Example System Reference Model based on speech in centre channel

Note that other styles and methods of speech delivery may be possible.


Optional user control of mix of speech and background sound
L
C (speech)
R
LFE
Ls
Rs
MCA Encoder
Multichannel Audio (MCA) bitstream
(+ signalling/metadata)
MCA Decoder
L
C
R
LCLEAN
Clean Audio to stereo speakers / headphones
RCLEAN
Multichannel Clean Audio Outputs
Reduced level of L, R, and zero level of Ls, Rs, LFE,

APPENDIX L Annex E.4 from ETSI TS 101 154 V1.9.1 (2009-09) Digital Video Broadcasting (DVB); Specification for the use of Video and Audio Coding in Broadcasting Applications based on the MPEG-2 Transport Stream (Technical Specification)



E.4 Coding for Clean Audio SA services

In case an AD_descriptor is present in conjunction with a service signalled as audio_type 0x00 ("undefined"), the AD descriptor is utilized to provide a clean audio service. The level by which the main audio service should be attenuated for Clean Audio output is signalled in PES_private_data within the PES encapsulation of the main programme audio component (as specified in ITU-T Recommendation H.222.0 / ISO/IEC 13818-1 [1]. In this case, only **AD_gain_byte_center**, **AD_gain_byte_front** and **AD_gain_byte_surround** are evaluated. This allows for a dynamic level modification of channel groups in a surround sound setup.

Encoding: Support for the encoding of Clean Audio is optional.

Decoding: Support for the decoding of Clean Audio is optional.

The principles of processing in a SA decoder in the case of Clean Audio are shown diagrammatically in figure E.3.

Figure E.3: Functionality of Clean Audio decoder processing

The audio processor should accentuate the level of the centre channel (containing the dialogue) and attenuate the other channels, according to the values signalled in the AD_descriptor. The level of the centre channel added should additionally be under user control to allow individual tailoring of the sound for audibility.

APPENDIX M CM-AVC0084 DVB CM-AVC Commercial Requirements: Delivering “Clean Audio” for the Hard of Hearing

DVB CM-AVC Commercial Requirements:

Delivering “Clean Audio” for the Hard of Hearing

Author: Tim Sheppard with most input from the UK Clean Audio Forum
Date: 18th September 2006
Version: 1
Status: Draft

Introduction

An estimated 50 million people in Europe find speech (dialogue and narrative) in television programmes difficult to follow. Reducing the background sound (music, effects and ambient sounds) can make the speech clearer thus providing 'clean audio'. Ofcom sponsored research at the University of Salford in the UK has found that 'clean audio' improves the clarity, quality and enjoyment for hard-of-hearing people and their families.
People with good hearing are able to more easily discern the speech from background, but this ability declines naturally with age. For people with even minor hearing loss or reduced cognitive ability, a high level of background audio information often leads to reduction in audibility or intelligibility of the speech; this often manifests itself before hearing loss would be clinically identified by traditional audiometric measures. This can sometimes lead to the viewer turning up the volume, but doing so will not necessarily improve intelligibility and may even exacerbate the problem.

With the increasing popularity of multi-channel audio (also known as surround sound) it is now possible to separate the speech from background sound in the TV transmission. This document proposes that a method of signaling such an audio stream is introduced into the relevant DVB standards.

Background

For the purposes of this document, Clean Audio is defined as:

> "The provision of television sound in a manner that achieves clarity of speech (programme narrative and dialogue) for the maximum number of viewers, especially those with hearing loss."

Various measures might be taken to achieve this.

- *taking particular care in production and/or*
- *providing specific mechanisms for delivery which allow the user to select and/or control a mix of speech and background sound (music, effects and ambient sounds) to suit his or her taste/capabilities, for example*
- *by transmitting metadata to generate a separate alternative mix in the receiver or*
- *by providing a separate alternative mix*

The additional costs of producing and transmitting a dedicated sound channel to offer an alternative mix solely for Clean Audio preclude the latter approach.
However, the introduction of multi-channel audio opens up a new opportunity to offer choice to viewers. In particular, this will be helped over coming years by the widespread

introduction of high-definition television programme material in the home from a variety of sources, leading to multi-channel audio becoming more generally available in consumer equipment.

The commonly employed approach to surround sound production for film where the front centre channel is typically used to carry speech has already resulted in a large and so far untapped archive of programme material suitable for delivering Clean Audio.

University research has found that a good combination of enjoyment and clarity of speech is obtained when the speech can be delivered in the front centre channel and the level of left and right channels is reduced by a modest amount.

To maximise the potential market, it is foreseen that a common approach should be adopted for the mechanisms used to provide assistive Clean Audio, whether for packaged media, or for broadcast or IP delivery.

Commercial Requirements

Technical specifications for Clean Audio are required that meet the following commercial requirements covering the interests of end users, service providers and network operators.

General

1. The specifications shall define optional (rather than mandatory) mechanisms.
2. The mechanisms defined in the specifications shall not prejudice the enjoyment of television sound for those who wish to listen to television sound produced and delivered conventionally.
3. The mechanisms in the specifications shall not prejudice the delivery of other assistive services such as Audio Description.
4. The specifications for delivering Clean Audio and associated data shall be simple and coding-technology neutral, i.e. shall be capable of being employed in conjunction with any multi-channel audio coding technology that is supported by DVB.
5. The specifications shall be suitable for use on any platform that is employed for delivering multi-channel audio, including packaged media, broadcast and IP delivery.
6. End users should be able easily to select Clean Audio using a simple interface and (optionally) to control the mix of speech and background sound.

Signalling and metadata

1. The specifications shall enable the receiver to identify the incoming multi-channel audio as offering a Clean Audio service.

2. The specifications shall enable electronic programme guides to optionally include an indication of which programmes are available with Clean Audio.

Types of Clean Audio

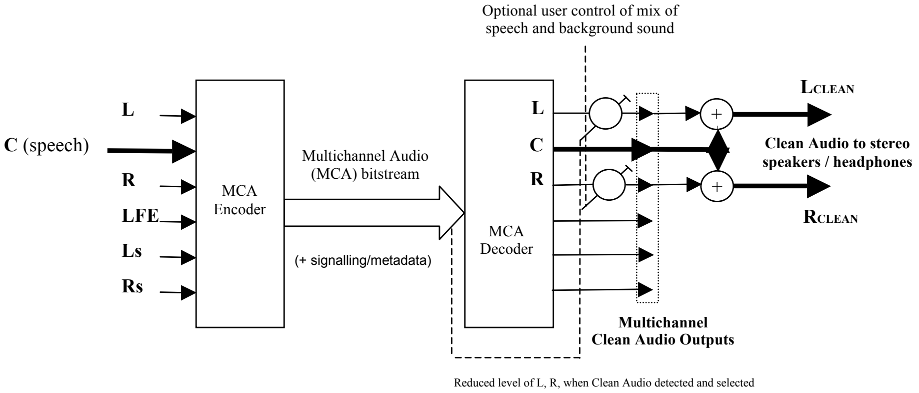
1. The following is a non-exhaustive list of possible types of Clean Audio, the signalling specifications should support at least these and allow room for later additions:
    a. Clean Audio as a separate audio stream
    b. Conventional television soundtrack authored as Clean Audio
    c. Clean Audio derived dynamically using transmitted metadata
    d. Clean Audio derived automatically without use of metadata
    e. Programme material not suitable for Clean Audio

Quality of Service

1. The timing of the Clean Audio with respect to vision as delivered by the receiver should not differ from that of the multi-channel audio.

System Reference Model

This model shows the case when the audio is output through stereo speakers (or headphones).


Optional user control of mix of speech and background sound
L
C (speech)
R
LFE
Ls
Rs
MCA Encoder
Multichannel Audio (MCA) bitstream
(+ signalling/metadata)
MCA Decoder
L
C
R
+
+
LCLEAN
Clean Audio to stereo speakers / headphones
RCLEAN
Multichannel Clean Audio Outputs
Reduced level of L, R, when Clean Audio detected and selected