Time delay compensation

of

distributed multiple microphones

in recording

-an experimental evaluation

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To my mother for her continuous support of my work, I dedicate this thesis.

LIST OF DIAGRAMS

ABSTRACT

In the search for improved reproduction of a classical music performance, the technique of time delay compensation is shown to be a useful tool for balance in recording. This paper investigates the importance and validity of small time adjustments in recordings to compensate for variation in distance between spot microphones and a main stereo pair. Conventional recording techniques, psychoacoustic considerations and technological aspects of the use of delays are researched in order to determine their validity in timbral improvement.

Multiple microphone set-ups are used to record both large orchestral works and smaller-scale classical compositions where auxiliary microphones may be needed to ensure an optimum balance in the final mix. Small time delays are derived from calculations involving the distances between microphones, the speed of sound, and humidity and temperature readings from the hall. Proper synchronization of these delays is desirable to preserve phase coherence and combat comb-filter effects. Precise delay units are used to compile musical excerpts for listening tests.

The results reveal any change in sound quality and provide a basis for investigating both the positive and negative effects through objective study of the value of time delay compensation in the live recording reproduction of classical music performances.

RESUME

Dans le cadre de recherches sur la façon d'améliorer la reproduction de la musique classique, la technique de l'égalisation de temporisation revêt une grande utilité pour l'équilibre de l'enregistrement. Dans cet article, l'auteur étudie l'importance et la valeur des petits ajustements temporels visant à compenser l'écart de distance entre les microphones isolés et la principale paire stéréophonique L'auteur analyse les techniques d'enregistrement classiques, les paramètres psycho-acoustiques et les aspects technologiques de l'emploi des temporisations pour déterminer leur valeur dans l'amélioration du timbre.

La mise en place de microphones multiples sert à enregistrer les oeuvres pour grand orchestre et les compositions classiques pour formation plus restreinte où il se peut qu'on ait besoin de microphones auxiliaires pour assurer un meilleur équilibre dans le mixage final. Les petits retards temporels sont tirés des calculs sur la distance entre les microphones, la vitesse du son et les relevés d'humidité et de température dans la salle La bonne synchronisation de ces retards est souhaitable pour préserver la cohérence de phase et lutter contre les effets des filtres en peigne. Les dispositifs de temporisation précis servent à compiler des extraits musicaux pour les essais d'écoute.

Les résultats révèlent le moindre changement dans la qualité du son et servent de fondement à l'étude des effets positifs et négatifs par l'étude objective de la valeur de l'égalisation de temporisation pour l'enregistrement en direct de concerts de musique classique.

Chapter 1

OVERVIEW OF CLASSICAL MUSIC RECORDING PRACTICE: THE NEED FOR BLENDING MAIN MICROPHONES WITH SPOT MICROPHONES

1.1 Classical Music Recording Considerations

The recording of a classical music performance is generally most successful when listeners are least aware of any signal processing by the recording technicians. There are arguments as to whether the reproduced sound should be as close as possible a replica of the performance, or whether the recording engineer should improve on the performance where feasible via signal processing.

Although there is no specific recording technique for classical music, the microphone technique will have a great effect on the reproduced sound. The choice of a particular technique will depend on various circumstances. Three basic philosophies developed regarding microphone placement with the introduction of stereophonic recording: 1) coincident stereo, 2) spaced omni, and 3) multiple microphones. Each technique has its own uses in the recording studio, some with advantages over others in certain situations, such as a live event when the room is not familiar. There are three basic categories into which these philosophies fall: single-point, main microphone pickup (this includes coincident, near coincident and spaced microphone techniques); single point with auxiliary microphones, and multi-microphone technique. In the first case, one stereo microphone is used to record the sound as accurately as possible. This is considered to be the minimalist approach. The second case uses one main stereo microphone as well with additional spot microphones to support weak instruments or groups. This requires mixing, as does the third case where each instrumental group is recorded with its own microphone.

Case two may offer an advantage over case one and three, especially where the hall is not familiar, by using spot microphones for emphasis. Also, by adding an appropriate time delay to the spot microphones, which will be discussed in Chapter 2, the sense of depth can be improved. This investigation will be regarded as another microphone technique to be explored in the same manner as more conventional recording techniques.

A proper understanding of the effects of microphone placement is necessary when making a decision with respect to the size of the hall, the distance from the program source and the dynamic range of the program. A recording will only give the impression of being close to the performance if a large amount of high frequency information is present Moving away from the source, the reverberation increases, the high frequency content diminishes, and the dynamic range compresses It is difficult to eliminate the variables as to why one recording sounds better than another-even when using the same main pair in the same hall. This could be due to the microphone placement, instrument placement, the Fletcher-Munson effect, proximity effect, or another acoustical phenomena The engineer must quantify these results as much as is possible but remember that the ear is the ultimate judge of the recorded quality.

Studies by W. Woszczyk [1] and B. Bartlett [2] provide a clear methodology on acoustical and timbral analysis of both multi-microphone and single microphone techniques on single instruments. The sound of a musical instrument or of a complete orchestra is formed by the direct sound waves and the reflections from the ceiling and walls of the concert hall The acoustical quality depends on the ratio of the direct to reverberant sound as does the perception of depth. Studies of the vibrational characteristics of the orchestra can determine the microphone choice and placement in the same manner as the study of vibrational characteristics of each instrument. Both close and distant microphone placements can be

[1] Wieslaw R. Woszczyk, "Improved Instrument Timbre Through Microphone Placement," Recording Engineer/Producer,Vol.10, (October 1979), 78-95

[2] Bruce A. Bartlett, "Tonal Effects Of Close Microphone Placement," J.Audio Eng.Soc., Vol.29, No.10, (October 1981), 726-738.

used to discover where the minimum number of microphones should be located in order to reproduce a reasonable balance of the total spectrum.

Again, by using spot microphones in combination with the main pair on an orchestra, small time delays can be implemented by calculating the distances between microphone capsules with reference to the speed of sound and humidity and temperature readings from the hall

When we listen to music. we hear a combination of the sound produced by the instruments and the resonance of the room or hall. Denon's anechoic recording samples[3] are invaluable in demonstrating the effect of different microphone techniques by removing this hall resonance They also demonstrate the effect of adding resonances from different halls to the same piece of music recorded in an anechoic chamber. The results can be misleading since the initial reflections are not present (even with the addition of the room resonance) which is not the case in normal recording situations

Rooms usually do the opposite of what recording engineers would like. They decrease the low frequency separation and will sometimes rather increase the high frequency separation The opposite is desirable especially with coincident recordings which may lack spaciousness Listening to the orchestra or ensemble as they rehearse in the room and comparing this sound with the initial microphone setup can be extremely beneficial. In a dead room, one main microphone pair cannot achieve a blended quality as it can in a live hall. Normally a close multiple miking technique would destroy depth but in a dead room the reflections are not present to provide a sense of depth. In this case it may be necessary to use spot miking techniques to get away from the negative effects of the room and improve overall balance.

Our hearing mechanism is responsible for localizing and judging acoustical sound quality. In order to simulate spatial depth, our ears need the appropriate order of direct

[3] Denon, Anechoic Orchestral Music Recording, (Japan:Nippon Columbia Co. Ltd., 1988).

sound, first reflections and reverb. In a good stereo recording, these early reflections must be captured with their correct placement to simulate natural hearing. They support the original sound and help us to localize the sound source with respect to distance

In order to evaluate the performance of the main microphone pair, we must consider the music, the instruments, the room and the purpose of the recording Experimental research in choosing a main microphone was conducted by Martin Wohr[4] in Germany. After conducting listening tests as to the best choice for a main microphone it was concluded that there is no clearly favorable main microphone technique. Although there is no "best" solution different factors can contribute to making a wise choice

In choosing a main microphone for an overall pickup, there are a number of possible approaches. These can be divided into three main groups: coincident, near-coincident and spaced-pair technique. They include X/Y, A/B, M/S, ORTF, Blumlein and the Dummy Head method. Depending on the technique, localization can be achieved through intensity differences, time differences or a combination of the two. Angling cardioid microphones with the diaphragms together in a coincident arrangement (X/Y, M/S,Blumlein) produces intensity differences. Spacing cardioid or omni-directional microphones (A/B,dummy head) produces time differences between channels Angling and spacing cardioid microphones (ORTF) produces both intensity and time differences between the channels. We use a variety of physical cues to determine the location of a sound source Beside time and intensity differences at the two ears, changes in the spectral composition of sounds due to head diffraction, pinna effects or sound source movements can also influence the perceived direction of the source. The more of these cues available, the more sure and accurate the sound source location will be.

[4] Martin Wöhr, "Untersuchungen zur Wahl des Hauptmikrofonverfahrens", (Munchen: Bildungswerk des Verbands Deutscher Tonmeister, Bericht 14, Tonmeistertagung, 1986).

Since classical music encompasses a large dynamic range, the microphones should have very low noise and distortion. Due to the size of an orchestra, in order for the tonal qualities of all instruments to be reproduced equally well, the microphone should ideally have a wide, flat response at all angles of incidence and the polar pattern should ideally be the same at all frequencies.

The final consideration in any miking situation is the sound. The engineer must decide whether or not the sound adequately represents the original sound source through localization, depth and presence, as well as clarity and balance of the individual components. Although there is no specific method for miking classical music, it would appear that spot microphones could be advantageous especially with the option of time delay compensation.

1.2 Spot Microphone Considerations

In recording an orchestral work in an unfamiliar room or hall, spot microphones may be used for improving balance, proximity, reverberation or depth (with time delay compensation) to embellish the sound from a main microphone pair. They also make up for the lack of visual cues of the listener at a live performance. In this manner we are taking advantage of both close and distant miking techniques.

The spot microphone can serve to improve clarity, definition and the high frequency transient information when combined at an appropriate level with the low frequency and reverberant energy captured by the main microphone pair. For optimal spot microphone placement the timbre and localization of the instrument should be listened to through the main pair before deciding on panning and the placement. It is important to solo each spot microphone, to determine whether it compliments the sound of the instrument through the main pick-up, and to listen to the combination of microphones for dramatic changes in sound

When considering the complete orchestra, a pair of woodwind spot microphones are placed up over and high above the woodwinds and a bit in front to add more front edge or presence to the woodwind and brass instruments, not to change the overall balance. The microphone acceptance area can be matched to the width of the group. A spot microphone for a soloist in front of the orchestra can be placed about one meter from the soloist and and mixed in at a low level-just enough to add definition. In spot miking the percussion, the microphone(s) can be placed about one meter above the timpani or overall percussive sections in an orchestra but much closer for a drumset such as in Listening Test recording B. If only one spot microphone is used for emphasis over the timpani in the orchestra an omni-directional pickup may be used (Listening Test A: recording no.1). The percussion section is also separated somewhat from the rest of the orchestra, therefore off-axis coloration should not be a problem.

Because of radiation patterns, the close miking perspective works better on piano, percussion and bass than on strings, brass and woodwinds. The effect of single close microphone placement on the timbre of an instrument would suggest that the instrument should be miked only as close as necessary. The closer the microphone is placed. the more selective the output will be. Most musical instruments are meant to sound best at a distance since an instrument radiates different tone qualities in different directions and different spectra are produced from different parts of the instrument. A flat-response microphone does not necessarily provide the most natural reproduced sound.

1.3 Negative Effects of Spot Miking

Before deciding whether or not to use spot microphones, the engineer must be aware of the negative effects that can result from this process. In: "Hauptmikrofon und Stützmikrofone--neue Gesichtspunkte für ein bewährtes Aufnahmeverfahren,"[5] Günther

[5] Günther Theile, "Hauptmikrofon und Stützmikrofone--neue Gesichtspunkte für ein bewährtes Aufnahmeverfahren," Bericht 13, (Tonmeistertagung: 1984), 170.

Theile states that the non-delayed spot microphone has a negative effect on the simulation of spaciousness. For simulation of spatial depth the ear needs the appropriate values which are given by the temporal order of direct sound, first reflections and reverberation. If the spot microphone is not delayed, the listener may perceive the direct sound as coming from this microphone with the main microphone simulating reflections and reverb, thus distorting the temporal order of the impulse. This is a result of the direct sound from the main pair being late due to distance. What should be perceived as first reflections from the spot microphone are arriving too soon.

There are some cases where these "negative" effects may prove beneficial as a result of any number of circumstances. For example, clarity can be improved due to the spot microphone signal arriving ahead of the main microphone signal. However, this can detract from the spaciousness of the sound.

1.3.1 Acoustic Phase Cancellation

Phase cancellation is not only produced by dry or direct sounds but also by reflected sound which room surfaces can reflect back into the microphone. If two waveforms are combined and their relative phase altered, a new waveform will result from both constructive and destructive interference and can cause distortion. Proper use of Burroughs' 3-to-1 rule will minimize acoustic phase cancellation.[6] This states simply that if instrument #1 is located one foot from its microphone, then instrument #2's microphone must be placed at least three feet away from microphone #1. This rule can be followed as a general guideline however, it does not take into consideration that different instruments will be playing at different intensities. Thus the ear is the best judge in deciding whether or not the spot microphone is placed correctly or is beneficial to the recording.

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[6] Lou Burroughs, Microphones: Design and Application, (Plainview, New York: Sagamore Publishing Company,Inc., 1974), 115.

1.3.2 Comb-Filter Effects

Time differences between direct and reflected sound (usually floor reflections between 1 and 15 ms) result in comb filter distortion through dips and peaks of response that can color the sound. This results from superimposing the direct sound on its reflections, causing cancellations and augmentations within the response range. The resulting frequency response corresponds to a comb-filter curve.

When recording using multiple microphones, tonal quality can also be deteriorated due to comb-filter effects. The result is a form of amplitude distortion, due to the sound of an instrument being picked up by many microphones at different time intervals resulting in acoustic phase cancellation and distorting the frequency response.

Although there is agreement as to the existence of comb filters, there may be disagreement as to their relative subjective importance in certain situations.

1.3.3 Microphone Off-Axis Coloration

When using multi-microphone techniques, the engineer must be aware that each microphone will receive off-axis sound leakage from instruments it was not intended to pick up. Although the microphone may have an acceptable frequency response on axis, it can have a colored or uneven response to sounds reaching it from other directions. It is important to keep in mind that the published frequency response of a microphone most often refers to sounds arriving directly on axis. Coloration can occur not only when several instruments are recorded with multiple microphones, but also when many instruments are recorded with one microphone.

The polar response plot of a microphone shows the sensitivity of the microphone at one frequency band, plotted for every angle at which sound arrives at the microphone. An omni-directional microphone may be omni-directional at low frequencies but becomes a weak super cardioid at high frequencies. This is due to a narrowing of the polar pattern at high frequencies because of diffraction, which results in a dulling of the sound off axis. Uni-directional microphones tend to have a flatter random-incidence response than omni-directionals because off-axis sensitivity is diminished at high as well as low to mid frequencies. However, a cardioid pattern does not totally dampen the frequencies at 180°. When choosing a microphone, the random incidence response is a good indication of the tonal coloration of the microphone to the reverberation.

When the spot microphone is used in combination with the main pair its purpose is to reinforce the transient character or add presence to the direct signal, thus directional microphones work best in capturing this on-axis response.

Microphone interference is not as great a concern when using only a few spot microphones throughout the orchestra in addition to the main pair due to the distance between the microphones.

1.4 Focus and Objectives of this Work

In order to determine the value of small time delays in live classical music recordings, I have compiled Listening Tests from recordings of the McGill University Orchestra and smaller chamber ensembles recorded at the Banff Centre. This paper will concentrate on subjective impressions of the effects of time delays revealed through listening tests.

Having reviewed the considerations for classical music recording and spot miking, chapter 2 introduces the concept of mixing with delay compensation and discusses previous work in this area. Chapter 3 presents an overview of factors contributing to a better understanding of time delay compensation.

Listening tests, described in chapter 4, indicate the validity of the time adjustments. Comments from the listeners help determine the effect the delays have on the recording (eg. if frequency response problems are improved by the time adjustment), and whether subjective preferences are separate from this. Participants in the Listening Tests include experienced musicians, composers, producers and recording engineers.

Chapter 5 gives an analysis of the data from the Listening Tests and the problems encountered. The results and comments from the participants determine any change in sound quality and provide a basis for concluding the validity of the time delays; whether or not there is an optimum time delay (should the delay arrive in step with the main pickup), an apparent increase in depth, and whether these delays are audible and helpful in both large orchestral works and smaller-scale ensembles.

A final discussion of the results, in Chapter 6, summarizes the value of time delay compensation and how this technique should be viewed as a recording tool. Potential problems as a result of this technique are discussed and suggestions are made for areas that should be explored in future work.

Chapter 2

MIXING WITH DELAY COMPENSATION

2.1 The Concept of Delay Compensation

A challenging yet relatively unexplored area of recording research deals with the use of compensating time delays (5-30 ms) in multiple microphone recording in order to synchronize the direct sound from the close microphone with the direct sound of that instrument arriving at the main microphone pickup. This technique is claimed to enhance the perception of depth which contributes to the impression of a realistic sound stage.

Major recording companies show great differences in their approach to recording a classical orchestra. Some believe in a minimalist approach while others prefer to use multiple microphones and multitrack techniques when recording classical music for extra control of balance and definition. As stated earlier, this can result in comb-filter effects. The amount of comb-filtering will depend on the levels of the summed signals and the width of the common frequency band. Despite this effect, an optimal balance may require the use of multiple microphones-especially when recording in difficult acoustic environments.

This study was inspired by an excellent article: "Digital Time-Coherent Recording Technique," by Takeaki Anazawa and Yukio Takahashi,[7] in which the authors determine the detectable and permissible limits of the comb-filter effect on various instrumental sounds in multiple microphone recordings through delay compensation of the direct sounds (see section 2.2)

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[7] Takeaki Anazawa and Yukio Takahashi, "Digital Time-Coherent Recording Technique." Audio Engineering Society Preprint, 2493 (H-2), (October 1987), 1-8.

Changes in recording techniques are a direct result of digital recording practices which capture the subtleties and wide dynamic range of orchestral performances. Although the clarity of digital recording is forcing a trend back to simpler techniques, the development of large-scale integrated circuit technology has made it possible to delay signals very precisely with no effect on frequency response or deterioration of sound quality, thus promoting the use of spot microphones in addition to the main stereo pair.

2.2 Previous Work in Time Delay Compensated Mixing

In his article: "A Different Way to Record Classical Music," [8] Jurg Jecklin first proposed that recording techniques for natural music are improved by time-delaying the spot microphone signals.

In Japan, the Denon recording company designed a time-aligned digital console to correct the anticipatory effects of spot microphones. In the process, Takeaki Anazawa and Yukio Takahashi concluded that sound quality deterioration was greatly reduced as a result of digital time-coherent recording and that it was important to secure a delay compensation precision of 1 ms. It was also concluded that deterioration to the sound is audible even when the level of the non-delayed spot microphones is 10 dB below the level of the main pair.[9]

Very little study has followed regarding the necessity and accuracy of time-coherency in improving the tonal quality of recordings. The effect of crosstalk between microphones and the effect of delay compensation on reinforcing the hall tone has yet to be determined. Long

[8] Jürg Jecklin, "A Different Way to Record Classical Music," J. Audio Eng.Soc., Vol.29, No.5, (May, 1981), 329-332.

[9] Takeaki Anazawa and Yukio Takahashi, "Digital Time-Coherent Recording Technique," Audio Engineering Society Preprint, 2493 (H-2), (October 1987), 3.

term listening is a necessary factor in determining the improvement and effect this technique has on the overall balance.

As referred to earlier, Denon manufacturers have since produced a compact disc "Anechoic Orchestral Music Recording" showing the effects of time delay compensation in an anechoic chamber. They have demonstrated the effects of this technique using time-coherent recording with various microphone setups. The anechoic recordings help to demonstrate that even without the room reflections, a close perspective may require spot microphones to improve upon the pickup from the main pair. However, the addition of spot microphones with room resonance can be misleading since support microphones bring out spectral components not present at the main microphone due to the lack of reflections.

Theile[10] suggests that when using spot microphones not only must time delay compensation be used, but an increment must be added to the calculated delay in order that the direct sound from the spot microphone arrives with the first reflections at the main pair.

2.3 Delayed Spot Microphones

By using carefully measured delays, it is possible to synchronize the arrival time of the sound from a spot microphone and the main pair. If sounds do not arrive at both microphones simultaneously, there will be some acoustic phase cancellation which can result in frequency cancellation thus reducing the volume and changing the color of the instrument. It is important to listen to the combination of the microphones for drastic changes in the sound, especially when using spot microphones in closer proximity on a smaller ensemble. The different path lengths from source to microphones result in different phase relationships between the microphone outputs for the same sound. Also, the supported instruments can be reproduced with considerable precedence and sound too much up front, thus disturbing the spatial quality of the recording. In order not to affect the

10 Günther Theile, "Hauptmikrofon und Stützmikrofone--neue Gesichtspunkte für ein bewährtes Aufnahmeverfahren," Bericht 13, (Tonmeistertagung: 1984), 170.

spatial sensation and depth perspective of the instruments when using spot microphones, the signals from these microphones must be delayed by the travel time of the signal to the main microphone.

Using carefully measured delays with the spot microphones maintains the recording properties of a good stereo main pair with respect to the arrival of direct sound, first reflections and reverberation. It also allows the engineer to increase the level of the spot signals without bringing the supported instrument too much up front. Because our ears need the appropriate order of direct sound, first reflections and reverb.to simulate spatial depth, these early reflections must be captured with their correct placement It would therefore seem that time delay compensation.would be the best solution when using spot microphones.

It is possible to measure the delay so that the effect between the direct sound and first reflections remain the same. Since early reflections carry most of the distance information, and the first sound gives the location, it would seem in theory that the delayed spot signal should arrive in time with the first reflections. Using Theile's suggestion of adding an increment to the delayed calculation would accomplish this. This could also relate to the earlier mentioned theory by Olive and Toole that a strong reflection can have a positive effect in renewing the precedence effect.

In an earlier article: "The Subjective Effects Of First Reflections In Concert Halls-The Need For Lateral Reflections",[11] Barron suggests that spatial impression is primarily a low frequency phenomenon which depends again mostly on lateral sound energy below 400 Hz arriving between 10 and 100 ms after the direct sound. This suggests that time delay is important since it is the reflected sound which should have a high level of lateral velocity. Barron's work also revealed that spaciousness depends on the level of low frequencies.

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[11] Barron, M. "The Subjective Effects Of First Reflections In Concert Halls-The Need For Lateral Reflections." J.Sound Vib. (1971), 475-494.

Theile also points out that it may have a positive effect to delay left and right spot signals differently in order to simulate sideways reflections which do not change localization but do increase volume. This supports the earlier work by Barron regarding the importance of low frequency lateral reflections for simulation of spaciousness. Delaying spot signals would therefore seem beneficial by increasing the chance for lateral reflections at low frequencies.

Chapter 3

OVERVIEW OF FACTORS CONTRIBUTING TO A GREATER UNDERSTANDING OF AUDIBLE EFFECTS OF DELAY COMPENSATION

3.1 Psychoacoustic Considerations

It is important to examine available psychoacoustic evidence to support the use of compensating time delays. In order to do so, it is necessary to first look at the way our hearing mechanism works.

Psychoacoustics indicate that the real study of sound takes place between our two ears Although the ear is a highly developed organ, it is useful only when coupled with the powers of the brain. The arrival time of a sound is used as a location cue by the ear-brain mechanism ("precedence effect") This being the case, delaying the sound from the spot microphone so that it coincides with the arrival time at the main pickup would seem necessary in maintaining a correct time relationship

Although natural hearing is more satisfactory when listening in a hall at a distance from the source, microphones cannot focus in on sounds as our ears can We may be left with a nice sound but with no presence or sense of definition from a particular instrument. The lack of visual cues, which also have a large influence on auditory localization, must be compensated for in miking a sound source. Using small time delays on the spot microphones to compensate for the distance from the main pair can help the listener compensate for the lack of visual cues in creating an illusion of depth.

Stereo hearing must be considered as being different from natural hearing since each ear in stereo hearing hears two copies of the sound rather than one copy of the direct sound from the source. Level and time differences at the listener's ears are not the same as those from the loudspeakers, thus various stereo techniques are used to ensure that the loudspeaker signals produce cues that are compatible with natural hearing.

In reviewing the process for natural hearing, it is obvious there are intensity and time differences between the two ears due to amplitude differences because of localization of the source and the physical difference in arrival time due to the spacing of our ears. There are also differences in the spectrum of the sound entering the two ear canals partly due to head diffraction.[12] Delays as small as a few hundred microseconds (because of the physical bumps, ridges and cavities of the pinna) are used as location cues by the ear-brain mechanism This being the case, the use of time delay compensation should be easily perceived by our ear-brain mechanism.

The Fletcher-Munson curves or Equal Loudness contours are a series of graphs of sensitivity of the ear versus frequency at different loudness levels. These are helpful in our understanding of complex tones, showing that the rate of increase in loudness with an increase in intensity is greater for low frequencies and to some extent for very high frequencies, than for middle frequencies. This information is also useful in our understanding of varying intensities in comb-filtering effects where multiple microphones may combine well only at certain frequencies. Here again small time delays may help in controlling the overall level of an instrument or section in combination with a main pair.

3.2 The Role of First Reflections

The delayed signal from a spot microphone acts as a type of reflection in that it arrives after the direct sound from the main pair. First reflections play an important role in recording. Together with the reverberation they provide a significant sound clue to the

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[12] Sams, Reference Data For Engineers: Radio, Electronics, Computers and Communications, seventh edition, (Indianapolis, Indiana: Howard W. Sams, 1985), 34.

room's size, proportion and wall structure.The impression of the room size increases with an increase in the delay time of the reflections.

The first experiment to use simulated reflections in order to better understand the role of first reflections was conducted in 1950 by Helmut Haas, who investigated the effect of single short-delay reflections with speech. This has since become better known as the Haas effect which states that reflections occurring within approximately 40 milliseconds of the direct sound (exact time dependent on frequency content and envelope of the sound) become fused with the direct sound. Haas noted changes in loudness, sound quality and body of the sound due to these reflections. He also discovered that fusion will occur even if the closely timed echo comes from a different direction than the original source and the location of the total sound is determined by the location of the first sound. The second stimulus would have to be made sufficiently louder (15dB) above the first sound before overriding the precedence factor.[13]

While direct sound determines localization, early reflections carry most of the distance information. The later diffused reflections are the reverberation, they give information about the size of the room and some information about the distance of the source but not its direction. Reverberation time is not the only determinant of acoustical quality. The early reflection sequence has an important influence on subjectively audible quality differences in different rooms or halls.

When listening to a live concert, the ear receives a half dozen or more reflected repetitions of the original sound. This early sound field also presents front, back, left, right, top and bottom views of the sound. These views provide the listener with the many varieties of tone color produced from the spectrum of the instrument. Our auditory system is capable of organizing this information into a single percept. It is important that the

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[13] Brian C.J. Moore, An Introduction to the Psychology of Hearing, Second Edition, (London: Academic Press Inc., 1982), 163.

recording engineer be able to recognize the subtle qualities of an instrument (due to its precise construction) by these first reflections in the hall, and be able to evaluate the total blend of these qualities with those of other instruments. This can only be done through very critical listening.

Reflected sounds rarely have the same spectrum as direct sound since the high frequency content is normally reduced. Experiments by Sean Olive and Floyd Toole[14] again indicate that lateral reflections generate much more of a sensation of spaciousness whereas vertical reflections are more apparent as having an effect on timbre. The focus of this article however is on the effect of reflected sounds in stereophonic reproduction in typical rooms. Results show that delayed sounds arriving from the same direction as the direct sound are often less audible than when arriving from other directions and a strong reflection following a reflection-free interval can have a positive effect in renewing the precedence effect. Once again this supports having the delayed spot signal arrive in time with the first reflections.

3.3 The Perception of Depth in Recording

Before trying to simulate spatial depth through small time delays it is important to understand how depth is perceived with our hearing mechanism.

Our binaural hearing capability is largely responsible for the perception of depth. Therefore in order to simulate spatial depth, the stereo signal must contain elements similar to the ear signals for normal listening. Again, the order of direct sound, first reflections and reverb at the main pair must be similar as for natural listening.

It is important to consider that the acoustics of a room have an influence on the depth perspective. With a fixed microphone distance, the more reverberant the hall, the farther back the rear of the orchestra will seem. By moving the main microphone further back in a

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[14] Sean E. Olive and Floyd E. Toole, "The Detection of Reflections in Typical Rooms," Audio Engineering Society Preprint, 2719 (F-1), (November, 1988), 1-15.

hall, the ratio of the front to back distance will decrease and the front of the orchestra will become softer in relation to the level at the back. Varying the height allows us to change the front-to-back perspective. When using cardioid microphones bass instruments always seem to be closer on the recording than when listening in the hall due to the distance-depending component. Many factors are involved when judging a microphones sensitivity to the ratio of direct to reverberant sound energy in a room. This will depend on both the instrument and microphones directional characteristics as well as the the position of the microphone to the instrument and the directional characteristics of the room.

Air absorption can contribute to the sense of distance in an acoustic environment. Also,as we move away from a sound source the high frequency content is reduced. These high frequencies can thus be manipulated in a recording to change the apparent front-to-back depth.

In his article: "Spaciousness and Localization in Listening Rooms and Their Effects on the Recording Technique",[15] Griesinger says that by electronically increasing separation at low frequencies, coincident microphone techniques can produce superior recordings. He proposes a technique of spatial equalization which works by applying a bass boost to the L-R signal and/or a bass cut to the L+R signal. In this manner the low frequency spatial impression of the hall is emphasized without affecting imaging at higher frequencies. This can be related to Barron's earlier discovery of the effect of lateral reflections at low frequencies in relation to spaciousness. The importance of the ability to control the directivity of low frequency information is apparent. A microphone technique proposed by Wieslaw Woszczyk[16] applies the technique of second order gradient unidirectionality to

[15] David Griesinger, "Spaciousness and Localization in Listening Rooms and Their Effects on the Recording Technique," J.Audio Eng.Soc., Vol.34, (April 1986), 255-268.

[16] Wieslaw Woszczyk, "A Microphone Technique Applying The Principle Of Second-Order Gradient Unidirectionality", Audio Engineering Society Preprint (Los Angeles, May 1981).

extend the high directivity of the pickup to low frequencies since low frequency energy is not easily dissipated in a room. This is achieved by creating four pressure points to be measured by the microphone configuration. Whereas pressure microphones sample the acoustic field at one point only, first-order gradient microphones will measure the pressure at two points while second-order gradient microphones will measure the output at four points. What Woszczyk has proposed is an arrangement of two cardioid microphones wherein a second-order system will be effective at low frequencies while a transition to a first-order system will occur with increasing frequency. Further research into this technique could reveal new possibilities with respect to improvement in spaciousness by the advancement of directional microphone pickup at low frequencies, thus developing yet another tool in sound recording.

3.4 Musical-Instrument Radiation Patterns

By studying the vibrational characteristics of the orchestra, in the same manner as studying the vibrational characteristics of each instrument, microphone choice and placement can be better determined. This applies to both the main microphone and spot microphones placed within the orchestra. Although time delay compensation may help when combining the sound from the spot microphone with that of the main pair, it depends largely on the placement of the spot microphone. Capturing a certain frequency component may make up for a lack of presence of that component at the main pair.

The timbre of a musical instrument is made up of the various spectra radiated from different parts of the instrument. It is essential to know the radiation patterns of the various instruments in order to better understand microphone placement, especially where spot microphones are involved.

Briefly, string instruments have very complex directivity patterns. Each string group has a nondirectional characteristic for the low frequencies with the exception of the double bass. At higher string frequencies, there are regions of preferred radiation that are different for every partial and can alter dramatically with frequency. Due to the complexity of the radiation of string instruments, the frequency response at close range can result in comb filtering. Where spot miking is necessary, placement is crucial and careful listening is required when combining the close microphone with the main pair.

Woodwind instruments have definite multi directional patterns since the sound is diffracted in all directions through the open tone holes. In general, the components that lie below a certain cutoff frequency of each instrument will radiate in an omni-directional pattern. For components near this cutoff, the radiation will be disc shaped while components above this will radiate in progressively smaller angles from the bell in the direction of the hall. Here microphone placement is more important than type since the timbre changes more depending on direction. Placing a pair of spot microphones over the woodwind section in an orchestral recording can add definition to this section without getting too close to any one instrument. Adding delay compensation can also increase the depth perspective.

Brass instruments are much more omni-directional in terms of sound radiation since the directional characteristic is largely symmetrical about the bell. Radiation becomes narrower with increasing pitch which results in a duller sound the further off axis the microphone is placed. The characteristics of the brass instruments help them to overcome distance in being placed at the back of the orchestra. They frequently sound closer than other instruments located at the same distance from the microphone. Although spot microphones may not be necessary on the brass section in an orchestral recording, using delay compensation on the woodwind spot microphones also helps in achieving a greater sense of depth for the brass instruments.

The recognition of musical instruments depends on transients and the structure of the total sound envelope. In recording we must balance the sharp transient detail with the rich harmonic information in the reverberant environment. Here the timbre of the instrument is the result of the interaction between the direct and reverberant sound fields. Microphone sensitivity to this interaction is a result of the directivity of the sound from the instrument, the position of the instrument to the microphone and distance from the microphone, the polar pattern of the microphone and the room characteristics.

The direct sound is important for clarity and definition especially for short tones yet the total spectra of an instrument is measured best in a reverberant room where multiple reflections combine to produce a full ensemble sound. Since musical instruments produce low frequencies mostly omni-directionally, very little of this energy arrives at the microphone as direct sound. The percentage of low frequency information is much greater in the reverberant field and adds warmth and fulness to the sound.

In looking at overall orchestral radiation, we must take into consideration that the arrangement of the orchestra was chosen for obvious reasons due to balance and the spectral radiation of sound from instruments. As we move higher up in a hall, we perceive more high frequencies since the high frequencies of many instruments, especially strings, radiate upward rather than forward. Most orchestral instruments have a preferred axis of sound radiation that is taken into account in various orchestral seating arrangements.

Chapter 4

RESEARCH PERFORMED: EXPERIMENTAL METHODS AND IMPLEMENTATION

4.1 Procedure

In order to investigate the subjective impressions of time delay compensation, I have created a set of listening tests to better determine their validity as a recording tool.

The purpose of the listening tests is to demonstrate whether small compensating time delays can be detected in the context of both large and small classical music ensembles through changes of balance, depth, timbre, level or other unknown factors. Comments should determine which factors are subjectively audible. They should also reveal whether the delays do affect depth and whether optimum delays exist in varying situations.

The subjective judgement of acoustical quality can be very elusive due to the number of variables which can affect the recording. Employing a set of listening tests for a group of trained musicians, composers and sound engineers can help in eliminating a number of prior assumptions. My initial experience in working with delays soon indicated that the most suitable subjects were those who were both used to listening to music and had some experience with audiological tests. A group of fourteen individuals were tested, seven of whom are professionally trained recording engineers. The rest of the group was comprised of composers, musicians and producers, only one of whom is not a professional. This person was added for subjective testing purposes. Although the number of subjects used was small, the variation in response between the listeners was smaller than expected. Listening tests were given individually over the same monitors in the same room. The test required concentrated listening. Instructions were given on the tape; each excerpt was indexed for convenience and to allow extra time for comments where necessary.

In addition to using one main pickup for a live to two-track stereo recording, a backup multitrack tape was made using various main stereo pairs and individual spot microphones for accenting individual instruments or sections. After each recording temperature and humidity readings were taken in the hall as well as precise measurements, described in the following section, for calculating time delays.

4.2 Methodology

In order to measure the results as objectively as possible, I chose to divide my test into four separate sections. Listening Test A deals only with the orchestral recordings from McGill. In most of the 10 sections making up this test, the subjects were asked to compare various excerpts to one reference (the main microphone pair). In this manner I was able to add varying amounts of time delay to the spot microphones (which are always compared to the main pair) or add no delay to test for depth perception, optimum delay time and the audibility of comb-filter effects or other relevant factors. Spot microphones are used on only the necessary instrumental groups and not more than four spot microphones are used at a time. This was found to be best in controlling the overall balance.

Listening Tests B and C are both recordings of smaller groups in the Project Studio at the Banff Centre. In these tests, the emphasis is on longer listening excerpts and on differentiating between main stereo pairs and close miking possibilities in this room. In both cases one main pair is chosen as the best pickup. To this I added a spot microphone to each instrument or group of instruments in order to reinforce the balance of any or all instruments and to avoid the negative effects of the room. The spot microphones in this case are all added to the main stereo pair first without delay compensation then with delay compensation. The excerpts are longer so as to demonstrate the long term effect on balance and not initial comparisons as may be the case in Listening Test A. These tests also demonstrate the results of using delay compensation on both a good (Test C) and bad (Test B) recording from the same hall.

Listening Test D is another extreme in showing the difference in using very short A/B excerpt comparisons. This test is used to show whether a difference can be perceived in very short time intervals.

Instrumentation: Delay units used in experiments:

McGill Recordings:

Eventide Ultra Harmonizer H3000 (Digital Delay program):
Left channel: 1.26 ms residual delay
Right channel: .2 ms residual delay

Roland E-660 Digital Parametric Equalizer(Digital Delay program):
Both channels: .2 ms residual delay.

Banff Recordings:

Lexicon 480L Digital Effects System (Twin Delay Program):
Both channels: .2 ms residual delay

A pulse generator and oscilloscope were used in order to check for residual delays within the units used. The .2 ms residual delay is minimal for both the Lexicon 480L and the Roland Digital units. A 1 ms delay was subtracted for the left channel calculated delays for the Eventide in order to compensate for the residual delay.

All units were tested for Frequency response, THD (Total Harmonic Distortion) and IMD (Intermodulation Distortion). Graphic displays and results are located in Appendix D.

Specifications given for each unit are as follows:

Lexicon 480L: Frequency Response: 20 Hz-20 kHz, +0.5 dB,-1 dB.
THD&Noise, <0.015% @ 1kHz limit level (+18 dBm unity gain)
IMD, <0.05% SMPTE IM @limit level

Eventide H3000: Frequency Response: 5 Hz to 20 kHz +/-1 dB, +/-.5 dB typical.

Distortion: .01% (.007% typical) @ 1 kHz, 1 dB below clipping in "pitch change" mode, 0 shift, levels all at 0 dB.

Roland Digital Parametric Equalizer E-660: Frequency Response: 20 Hz-20 kHz, +0/-3 dB.

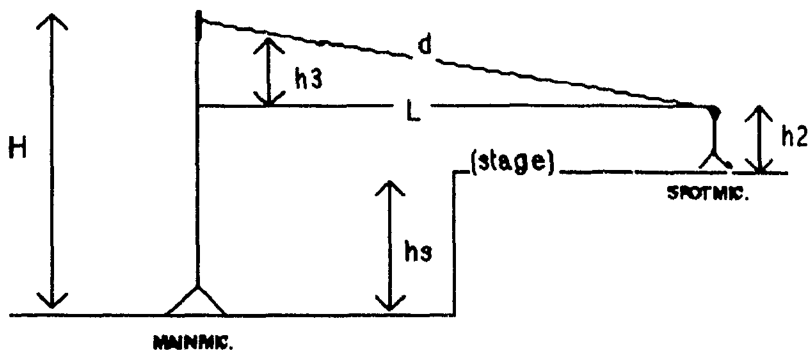
THD <0.015% (1 kHz at rated input)

All units show good specifications in all areas for the purpose of these tests-(see Appendix D)

The amount of delay is varied in both the positive and negative directions to investigate whether or not there appears to be an optimum time delay and to further investigate microphone crosstalk and the effects of over compensation.

The listening level was kept constant between the non-delayed and delayed spot microphone. This was achieved by connecting the oscillator to "multitrack in", at the desired channel on the patch bay (set to "0"VU), and to the left or right channel of the delay unit. The return of the unit was then patched to the desired channel, using the unit's level setting to match the "0" VU setting. Since comb-filtering drops some of the energy of the sound, the level may vary between delayed and non-delayed signals. In a few cases (see listening test solutions), I varied the level of the delayed spot microphone for comparative purposes where I felt the level difference was too noticeable. Also, increasing the delay will reduce the apparent amplitude of the signal. The calculated level, which was initially determined subjectively for the non-delayed spot, is always used in an earlier excerpt before any change is made in level.

4.2.1 A Calculation Model



The formula used for calculating the time delay between the spot microphones and the main pair as follows:

H represents the height from the audience floor to the main microphone capsule.

$\mathbf{h_s}$ represents the height of the stage (106cm)

$\mathbf{h_1}$ represents the height from the audience floor to the spot microphone capsule.

$\mathbf{h_2}$ represents the height of the spot microphone capsule from the stage floor.

$\mathbf{h_3}$ is the difference in the height between the spot microphone capsule and the main microphone capsule.

Therefore,

$h_3=H-h_1$

$h_1=h_s+h_2$

$h_3=H-h_s-h_2$

$d=\sqrt{h_3^2 + L^2}$ (from $A^2= B^2+C^2$)

$d=\sqrt{(H-h_s-h_2)^2+L^2}$

Time= $d/v_{(wet)}$

If the temperature reading is 27°C and humidity is 40%,
from the CRC handbook of Chemistry and Physics we know that
the velocity of sound at 40% humidity (20°C- 2k frequency) is 343.95 m/sec.
and 343.56 m/sec. for 0% humidity (20°C@2k).

We also know the velocity of sound in dry air at 27°C is 347.44 m/sec.

$v_{(wet)}=v_{(dry)}$ x v 40% humidity(20°C)/v0% humidity(20°C)

$v_{(wet)}$=347.44 x 343.95/343.56=**347.84 m/sec**

If d=6 m, then T=6/347.84=**17.2 ms**

4.2.2 Temperature and Humidity Effects on the Speed of Sound

Tight control of temperature and relative humidity must accompany the use of very small time delay increments to improve room response. In implementing small time delays to compensate for differences in distances between microphones, temperature and humidity readings must be used in calculating the speed of sound, which is used in the overall calculation of the amount of delay.

The speed of sound is dependent on the temperature and humidity reading in the hall. Tiny changes in these conditions will affect the phase relationship of direct and reflected waves. Both temperature changes and moisture affect the density of air and therefore the speed of sound in air. Since moist air is less dense than dry air there is an increase in the speed of sound with an increase in humidity. Also, an increase in sound absorption due to the humidity reading will cause a decrease in reverberation time where surface absorption is low; this is not significant for frequencies below 2 kHz. The frequency response of a microphone would not be significantly altered by humidity changes; there would be a small increase in absorption only at very high frequencies.

An acoustical delay is a direct result of the finite speed of sound. For normal temperatures (20-22°C) sound travels 344 m/sec. This will vary with both temperature and humidity changes. For the purpose of this experiment the steady state reading at the end of the performance is used since it can slightly affect the calculation for the speed of sound The CRC Handbook of Chemistry and Physics[17] publishes results of varying temperature and humidity readings on the speed of sound.

4.3 Listening Test Recordings

1. McGill Symphony Orchestra: Pollack concert hall; January 25,1989
   Microphone placement for recording: (see Diagram no.1)

   Mozart: Oboe Concerto in C, K. 314
   Bruckner: Symphony NO. 7

In order to capture the best overall pickup, I set up four main microphone pairs during rehearsal. Since the performance took place on two consecutive nights I was able to improve on the microphone placement for the main pairs and spot microphones for the second performance (which was used for the listening tests). My goal was to capture the best overall sound with one main microphone pair and use spot microphones to give definition to the woodwind, string and percussion sections without destroying depth A hypercardioid is used on the bass section to give extra definition of the bow on the strings.

Precise measurements were taken between microphone stands and the height of each microphone was calculated from both the audience and stage floor each night. A temperature and humidity reading of the hall was taken during each night's performance.

For the initial mix, I decided between a stereo Blumlein (AKG C422), an AKG 414 Blumlein, and the closely spaced B&K 4003's as the main pair.

The C422 Blumlein configuration did not produce an even pickup between capsules, and so was not used. The 4003's provided a more spacious reproduction while the AKG 414 Blumlein setup provided a closer, more intimate sound reproduction. This worked

[17] R.C. Weast, Ed., "CRC Handbook of Chemistry and Physics", Boca Raton, Florida: CRC Press Inc., 1979.

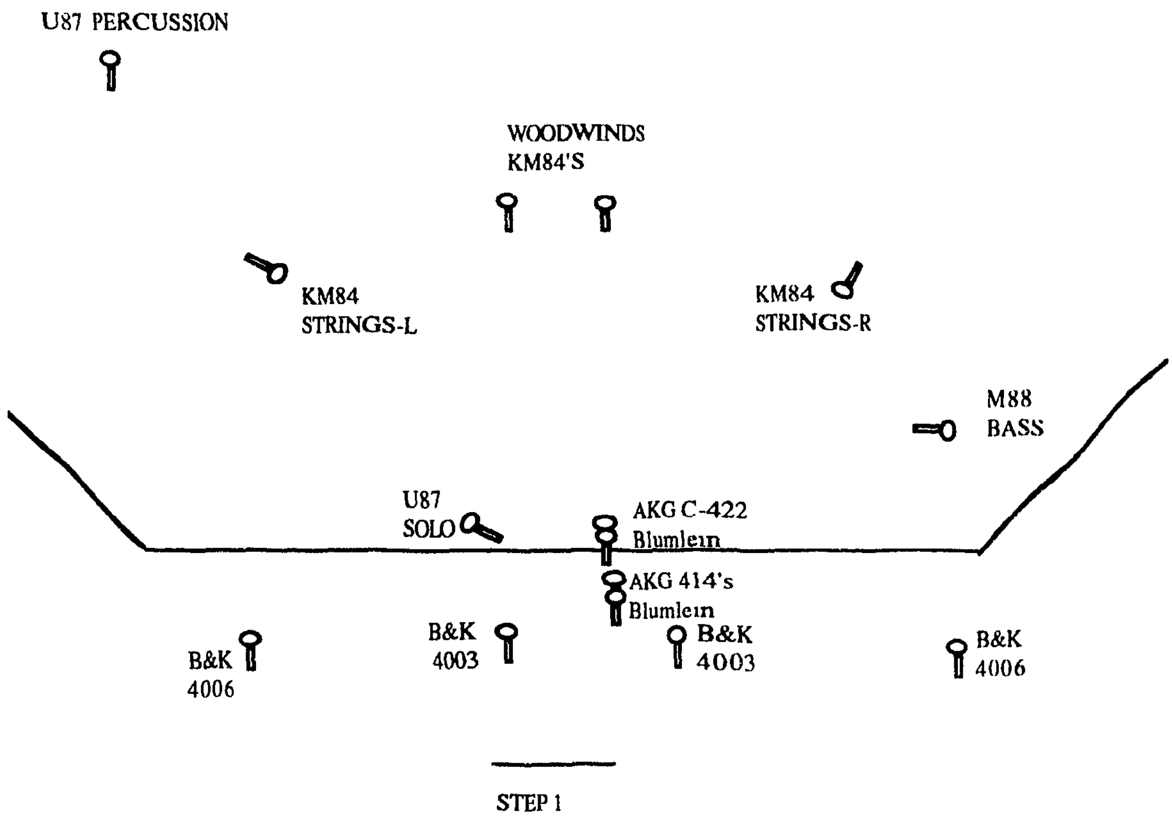
especially well on the Mozart piece in helping to distinguish quiet musical lines being played simultaneously by the orchestra. It was also easier to judge the relative distances of different sounds in the reproduced image with the Blumlein pair. Thus the addition of delayed spot microphones lends itself better to this technique rather than to the spaced omni pair where the time difference would have to be considered separately for each channel due to the greater spacing between capsules.

The addition of the B&K 4006's worked well in combination with the 4003's in creating a 4-omni wall technique, yet they caused phasing problems when combined with the precise imaging of the Blumlein pair.

Having chosen a main pair, I experimented with the addition of the delayed spot microphones. In all cases the faders were brought up discretely (just before they became audible as additional microphones). The woodwind microphones were placed 2 meters apart above the wind section and angled toward the left and right center of the section just before the brass. This was most helpful for the deep seating arrangement of the Bruckner. The addition of the solo microphone (Mozart), helped to stabilize the instrument and improve the attack without adding much more finger noise. The spot microphone on the first row of basses improved articulation of the sound of the bow on the strings and helped to capture the lower notes which the double bass is unable to radiate effectively. The string microphones were used in an attempt to give body to the string sound. Being placed behind the strings, the microphones gave a slightly richer sound than when placed in front, especially for the violins. A single percussion microphone was used above the timpani (Bruckner). The U87 was set to omni since it was the only microphone used and provided a better overall pickup.

Diagram No.1

1. McGill Symphony Orchestra: Pollack concert hall; January 25,1989



Main Microphone: AKG 414's (Blumlein configuration) H=5.56m

spot mics	h3	L	d	v (wet)m/s	T
1.percussion (U87) o	2.5m	9m	9.3m	347.84	26.9ms
2.strings-L (KM84)	2.1m	6.5m	6.83	347.84	19.6ms
3.woodwind-L (KM84)	2.2	6.1	6.48	347.84	18.6ms
4.woodwind-R (KM84)	2.1	6	6.36	347.84	18.3ms
5.strings-R (KM84)	2.15	5.3	5.7	347.84	16.4ms
6.Bass (M88)	3.8	7.5	8.4	347.84	24ms
7. solo (U87)∞	3.75	1.2	4.2	347.84	11ms

Note: The two sets of spaced B&K omni microphone pairs as well as the AKG C-422 were not used for the experiments.

2.McGill Symphony Orchestra: Pollack concert hall; April 1, 1989
Microphone placement for recording: (see Diagram no. 2)

Bruch: Violin Concerto
Mahler: Symphony No.1

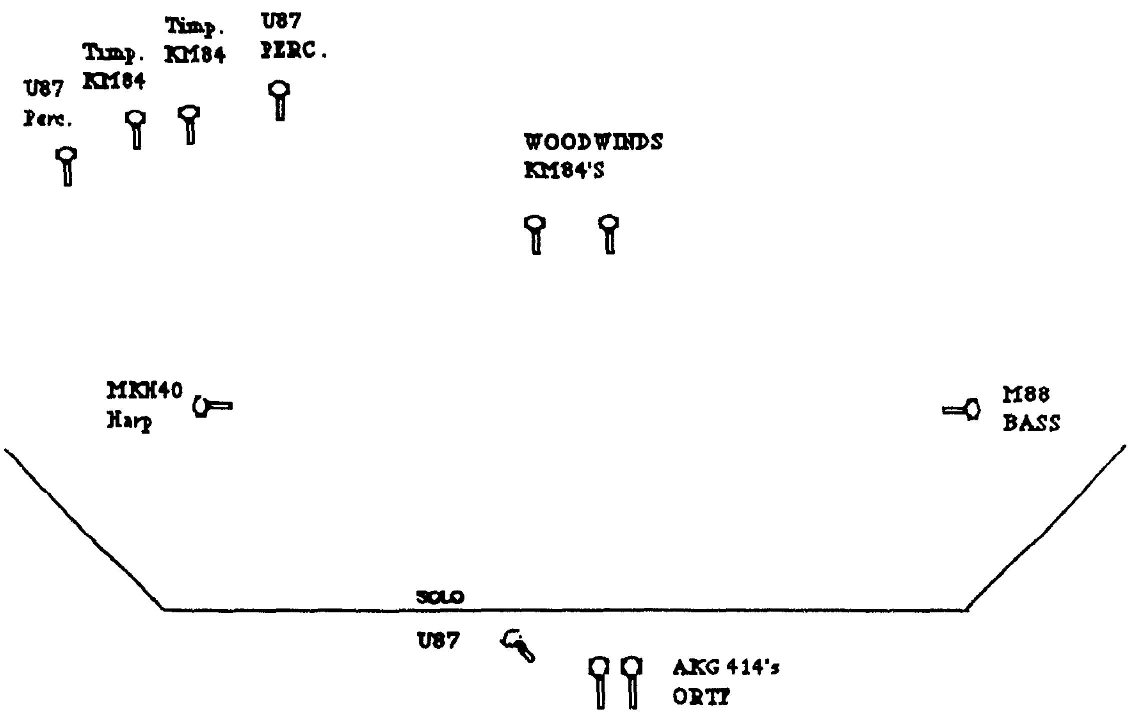
During the rehearsal, both a Blumlein pair (M130's) and an ORTF (AKG 414's) pair were placed on a large stand quite high over the orchestra and angled for the best overall pickup and to eliminate some of the noise from the audience picked up by the figure-eight's back lobes. An initial listening test between these two main pairs and a spaced pair of B&K 4006's resulted in my choice for using only the ORTF configuration for the concert.

The Blumlein pair, although capturing an "open" ambient hall sound, lacked breadth. Although they gave a good sense of depth, the total sound lacked power. The ORTF pair proved to be a better choice. There was less room felt, yet the solo violin was more integrated with the orchestra. The placement was generally off-axis to the brass (this suited the repertoire) and placed higher above the strings for a more even pickup. The overall string sound was much clearer and the pickup much more immediate on the percussion section. Since the percussion microphones are added only for attack (low end frequency rolled off at 100Hz), a high-pass filter was used on all four microphones. Again, I placed a spot microphone on the first stand of basses (mic. approximately 50 cm from instrument) for articulation, and a solo microphone to help localize and define the attack of the violin. A pair of microphones were placed at an angle over the woodwind section (1.75 meters apart). The spot microphones were mixed initially without delays and at a low level just to the point where they added clarity to the overall mix. This was a subjective choice on my part. The levels were not changed with the addition of the delay for listening test purposes-unless specified as such.

Precise measurements were taken between microphone stands as well as the height of each microphone from both the audience and stage floor each night. A temperature and humidity reading of the hall was taken during each night's performance.

Diagram No.2

McGill Symphony Orchestra: Pollack concert hall; April 1, 1989



Main microphone: AKG 414's (ORTF configuration) H=6.06m

spot mics	h3	L	d	v(wet)	T
1.harp (MKH40)	4.25	8.5	9.5	347.84	27ms
2.perc1 (U87)∞	3.5	10	10.6	347.84	30ms
3.timp1 (KM84)	3	9	9.5	347.84	27ms
4.timp2 (KM84)	3	9	9.5	347.84	27ms
5.perc2 (U87)∞	3	9	9.5	347.84	30ms
6.Woodwind-L(KM84)	1.9	6.5	6.7	347.84	19.5ms
7.Woodwind-R(KM84)	1.9	6.5	6.7	347.84	19.5ms
8.Bass (M88)	4.3	8.4	9.4	347.84	27ms
9.solo (U87)∞	3.1	2.7	4.1	347.84	12ms

3. The Banff Centre: Project Studio Recording; December 2, 1989
Microphone placement for recording: (see Diagram no. 3)

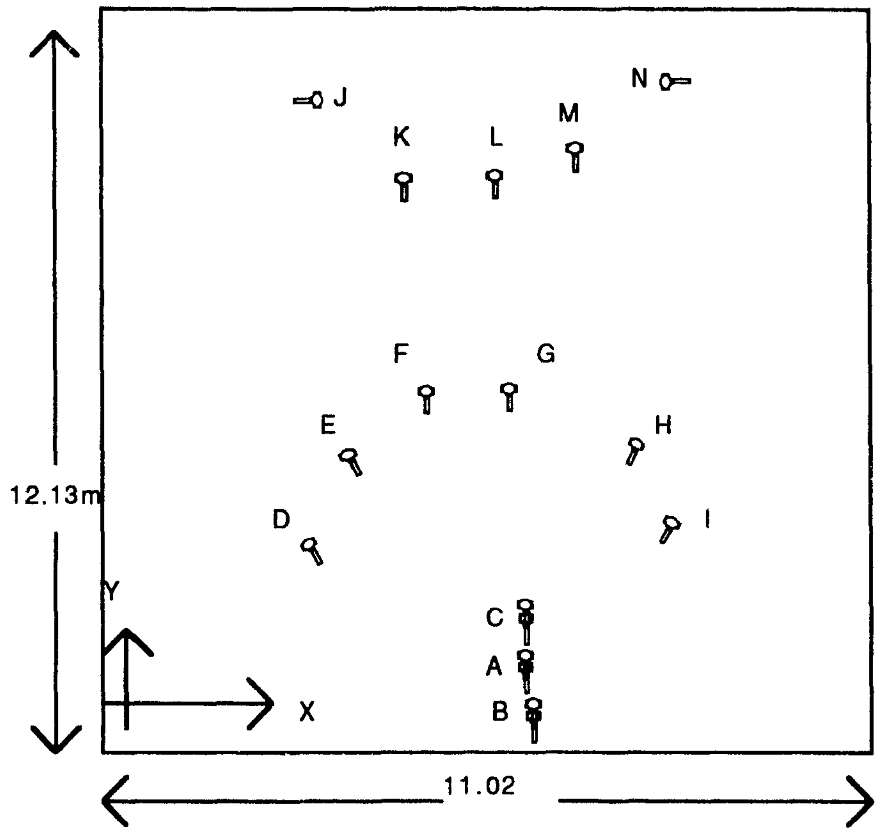
Stravinsky: L'histoire du soldat

For the recording of A Soldier's Tale, I set up three main microphone pairs in the Project Studio at the Banff Centre. As can be seen from Table 3, a spot microphone was used on each of the six instruments as well as five spot microphones to cover the percussion section placed at the back of the studio. The initial placement of the instruments was chosen for best compatibility with the main microphone.

After listening carefully to each microphone pair, I chose the Dummy Head configuration as the main pickup. It provided the best localization af the group yet there was a slight hole in the middle of the reproduced image. Using spot microphones in combination with the main pair eliminated this. Because I used spot microphones on all instruments in this case, the spot microphones were panned to match the position of the instruments at the main pair . I then did a mix using only the spot microphones, creating an even balance at a low level. The main pair was mixed in at a higher level with the spot microphones at a lower level so as to add clarity to the mix. While maintaining the spot levels, calculated delays were added to the spot microphones and recorded on empty tracks so as to create both a delayed and non-delayed multitrack mix.

Diagram No.3

The Banff Centre: Project Studio Recording; December 2, 1989
Stravinsky: Soldier's Tale



Three main pairs are represented by A,B, and C.

A=Sennheiser Dummy Head (using two B&K 007 microphones)
B=Blumlein (Beyer M130 microphones)
C=X/Y (AKG 422 Stereo microphone)

The Dummy Head was chosen as the main microphone in combination with the spot microphones.

Humidity: 35% Temperature:23°C

v(wet)=v(dry) x v 35% humidity;(20°C)/v 0% (20°C)
=345.12 x 343.89/343.56=**345m/sec**

Due to the method of measurements taken for recording sessions 3 and 4, L is calculated from $L^2=B^2+C^2$ (B=AX-X; C=AY-Y); h3=H-Z; $d=\sqrt{h_3^2+L^2}$

Microphone	X	Y	Z(height)	h3	L	d	T
A-Dummy Head (B&K4007's)	5.07(AX)	4.10(AY)	2.21(H)				
B-Blumlein (Beyer M130's)	5.07	3.88	2.61				
C- X/Y (AKG C-422)	5.07	4.50	2.26				
D-Vln (U87)	4.10	5.70	1.60	.61m	1.87m	1.97m	5.7ms
E-Cornet (TLM170)	4.11	6.44	1.33	.88	2.53	2.68	7.8ms
F-Clarinet (AKG460)	4.55	7.49	1.30	.91	3.43	3.55	10.3ms
G-Bassoon (B&K 4011)	5.30	7.96	0.77	1.44	3.47	3.76	10.9ms
H-Bass (BeyerM88)	6.33	6.00	0.40	1.81	2.66	3.2	9ms
I-Trombone (AKG414)	6.46	5.25	0.88	1.33	1.36	1.9	5.5ms
J-Snare (Senn.441)	4.48	9.67	0.60	1.61	5.6	5.8	16.8ms
K-Toms (AKG414)	5.13	9.44	0.46	1.75	5.34	5.6	16.2ms
L-Cymbal (KM140)	5.40	8.84	1.54	.67	4.75	4.8	13.9ms
M-Drum (M88)	6.35	9.59	1.35	.86	5.64	5.7	16.5ms
N-H. Hat (KM140)	5.97	10.18	1.50	.71	6.15	6.2	17.9ms

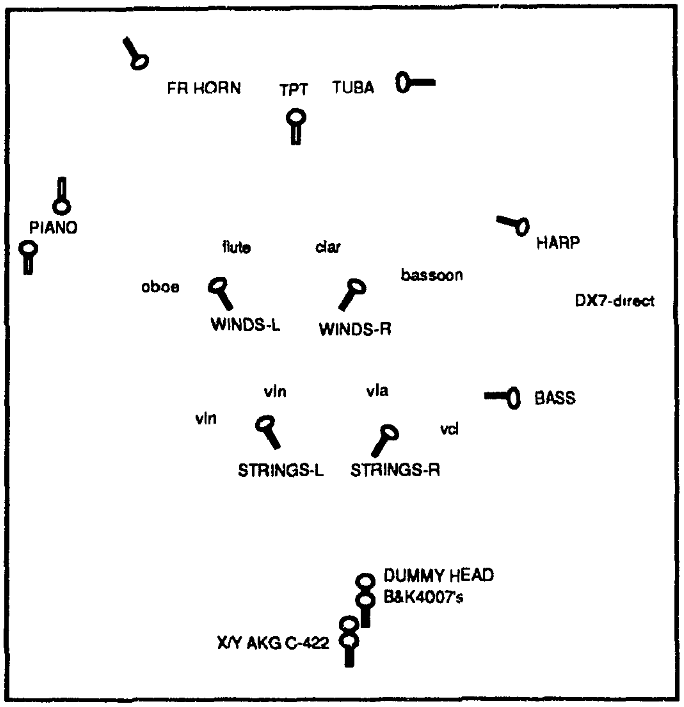
4. The Banff Centre: Project Studio Recording; January 29, 1990
Microphone placement for recording: (see Diagram no. 4)

Pecou, Thierry: Un temps jusqu'au bout de la fibre.

An X/Y AKG C-422 and a Dummy Head were set up for choice in determining a main pair. Spot microphones, as can be seen in Table 4, were used on all instruments or groups of instruments with exception of the DX-7 (I used a direct pickup rather than mike the amplifier). The four string and four wind players were arranged in such a manner as to allow for one pair of microphones to be used on each group. A pair of ribbon microphones were used to record the piano. These were placed under the left and right side of the sound board which provided a nice pickup and good isolation. In this case the AKG C-422 stereo microphone was chosen over the Dummy Head configuration as the best overall pickup. Having matched the position of the spot microphones with the main pair, I did a mix using only the spot microphones to obtain the best balance. I then brought up the main microphone pair and made a final subjective judgement as to the levels of each microphone for the best overall balance. A mix was done using the main microphone only, the main microphone with spot microphones and the main microphone with delayed spot microphones (keeping the same levels).

Diagram No.4

4. The Banff Centre: Project Studio Recording; January 29, 1990
Pecou, Thierry: Un temps jusqu'au bout de la fibre.



Humidity: 35% Temperature:23°C
v(wet)=v(dry) x v 35% humidity;(20°C)/v 0% (20°C)
=345.12 x 343.89/343.56=**345m/sec**

Note: distance measured directly from each microphone capsule to the main microphone capsule in this case.

The AKG Stereo microphone (X/Y) was chosen over the Dummy Head microphone as the main pair in combination with the spot microphones.

spot microphones	d	v(wet)	T
1. Strings-L(U87)∞	3.4m	345m/sec	9.9ms
2. Strings-R(U87)∞	3.5m	345m/sec	10.2ms
3. Bass(M88)	3.4m	345	9.75ms
4. Piano-L(M130)	5.8m	345	17ms
5. Piano-R(M130)	6.1	345	17.7ms
6. Woodwind-L(Sony C38)	5.02	345	14.5ms
7. Woodwind-R(Sony C38)	4.80	345	14ms
8. Harp(M88)	7.95	345	23ms
9. DX-7 (dir.)	7.95	345	23ms
10. Fr. Horn(B&K4011)	8m	345	23ms
11. Tpt.(AKG414)∞	6.60	345	19ms
12. Tuba(AKG414)∞	6.80	345	19.7ms

Chapter 5

LISTENING TEST RESULTS

Initially the Listening Tests were devised to determine whether 1) the use of compensating time delays does add depth, 2) the delay is perceivable by the participants and 3) whether there is an optimum time delay. From the results of the Listening Tests, it is obvious that the use of delay does add depth regardless of the subjective choice of the listener. That the delay is perceivable is suggested by the consistency of differences being noted. An optimum time delay seems preferable when delaying more than one section of the orchestra-in most cases with four spot microphones in combination with the main pair. Both the overall calculation for preferences and the individual comments of the listeners must be taken into consideration in the analysis of the test results. I will summarize the results in section 5.1 and explain these results in more detail through the interpretation of comments in section 5.2.

5.1 Analysis of Data

Listening Test C: shows the most positive result for time delay compensation as a useful recording tool. A good stereo recording should capture accurate imaging, good acoustics, tonal accuracy and depth. Listening Test C captures these elements with a stereo microphone pickup in "excerpt 2". Five excerpts are used to demonstrate two different main microphone techniques and to clarify the effects of delay compensation.

Listening tests reveal the following:

1) all participants chose main pair no. 2 to be the best main microphone pickup.

2) some participants chose main pair no. 2 with spot microphones as an audible improvement over an already good stereo pickup and all heard the effect of the spots despite preference (see Appendix C)

3) all participants noticed an improvement in depth in excerpt 4 and 90% (including all recording engineers) chose excerpt 4 with time delay compensation as the best choice.

This shows a positive result for time delay compensation with an already good main microphone pair. Both Listening Tests B and C were recorded in the same room and both use longer listening examples, which seem to work best in determining overall preferences with respect to balance.

By contrast, **Listening Test B** proved to be the least acceptable recording. The main pickup is not good; although comments reveal that the listeners hear an improvement in depth, individual preferences are very divided. Although comb-filter effects can deteriorate the sound, Listening Test B reveals that time delay compensation will not necessarily improve the recording and an increase in depth may actually be detrimental to the recording. No reverb was added to this recording in order to better determine the effect of delayed spot microphones.

The comments as to the preferred technique differed and listeners were divided on the benefit of the addition of delays to the spot microphones, even though the majority heard the addition of depth with the delays added (see Appendix C). Auxiliary microphones were needed to ensure an optimum balance in the final mix xcerpt no. 3 (spots added without time delay compensation) was preferable for balance with the proximity of the close microphones. This could be partially due to the circular placement of the musicians in front of the main pair. In this case the closer instruments can mask the more distant ones The further instruments sound too distant or dull in a not so reverberant space. Because of this seating arrangement, the extra presence of the instruments with the spot microphones added is more important than resulting comb-filtering effects.

In **Listening Test A**, multiple microphone setups were used to record large classical works. General conclusions from Listening Test A reveal that not only is time delay compensation detectable, it does increase depth. Changes in sound quality with reference to the main pair were highly detectable by all participants. A delay was almost always preferable in this set of tests and there was a majority preference, with few exceptions, for one excerpt.

A calculated delay is preferable whenever four microphones are used in combination with the main pair. In Test III: 100% preferred the calculated delay for the combination of both string and woodwind microphones with the main pair, and 80% preferred the calculated delay again with strings now 2 dB lower in level (excerpt 6). This seems to be due to an improved sense of balance throughout the orchestra (see Appendix C).

Delaying only one spot microphone or a microphone pair (woodwinds or strings) resulted in greater differences of opinion between listeners as to the preferred amount of delay. Using spot microphones on the string section did not necessarily improve the balance as the strings sound very strident in general. Another problem may arise with the "optimum" time delay being optimum only for the instrument(s) being supported whereas the delay may have a negative effect on surrounding instruments. Further discussion of this matter can be found in section 5.3.

Where level changes were made, they are always viewed as an improvement by the majority with respect to balance. This was mentioned earlier with reference to Test III. In Test V: again 100% preferred the calculated delay and 75% of the listeners opted for the calculated delay with an additional change in level for bass and percussion spot microphones (see solutions). In Test VIII,90% preferred the calculated delay, 80% of whom preferred the changes made in level (all sound engineers preferred this change in level). Test IX is one further example with 85% hearing the calculated delays as an improvement in the percussion section and 75% of this group preferring the adjustment in level.

Listening Test D demonstrates that even with very short A/B comparisons differences were heard by a large majority for all examples and delays are generally preferable to non-delayed spots. However, comments indicate that the excerpts are too short for the listeners to define the differences. An example of this can be seen when comparing Listening Test C, excerpt 4 to Listening Test D. Whereas 90% of all listeners chose excerpt 4 in C, only 35% chose the same excerpt as a preference for a much shorter comparison in Listening Test D.

5.2 Interpretation of Comments

In order to gain a better perspective on what the listeners really heard, a list of comments have been compiled in Appendix C. Both individual responses and overall judgements support the above analysis of the tests.

In interpreting the results of these comments, I will begin with **Listening Test A** (Tests I through X) TEST I supports the idea that the delay is perceived by the listeners. Various recording engineers reference to the horns s being "rounder", "louder" and "further back" with the time delay shows that the perceived image is best for this instrument. Other listeners are commenting on different sources which also support the perception of the delay and the increase in depth. This is obvious with such comments as "the brass section could be more prominent" or there being "more strings than before?...this is the best natural balance".

Number II demonstrates diversity in personal preferences. The use of time-delay compensation as a recording tool in this case is a matter of preference which would require longer term listening. Here, spot microphones and/or delay could be used to improve an already bad pickup for the strings. Although an improvement in balance is noted by several listeners in excerpt 6, a slightly larger majority prefer excerpt 4. The perception of the delay and an increase in depth are supported by the comments.

As stated earlier, an optimum (calculated) delay is preferable whenever four microphones are used in combination with the main pair; this is found in Tests III, IV,V, VIII and IX. With Test III, the 80% preference in excerpt 6 seems to be due to the 'balance' factor, with comments referring directly to a "better" or "nicer" balance. The comments listed for Tests V and VIII are similar and point toward an overall improvement in balance. The four microphones used in Test IX work very well for the percussion balance but not necessarily for the overall balance. Some comments indicated that the main pair alone was "best" yet others feel that excerpt 4 best supports the percussion instruments. One particular comment supports both: "In terms of volume and spatial balance, I definitely prefer the main pickup; in terms of percussion timbre and sound, excerpt no. 4 seems to best support the percussion instruments, but they are way in front of the rest of the orchestra and much louder".

Test IV also reveals a majority preference for the addition of delay to the four spot microphones, yet long term listening should be used to better determine overall balance as the comments are varied in terms of why excerpt 2 is preferable.

Tests VI and VII, dealing with 1 or 2 spot microphones, both reveal varied subjective preferences. There is no consensus through comments as to what determines a better balance, despite a majority preference for one excerpt. In Test VI, excerpt 4, comments indicate that the listeners hear the presence with the solo spot microphone yet differ in opinion as to its value. Comments in these two sections support the earlier analysis that delaying only one spot microphone or a microphone pair (woodwinds or strings) results in greater differences of opinion between listeners as to the amount of delay.

Where level changes were made, comments reveal an improvement by the majority with respect to balance. This is demonstrated in Tests III,V,VIII and IX. In Test III, where the string spots are lowered in level by 2 db, the comments refer to an improvement in the overall balance..."like no. 4 but better balance";"...nicer balance between sections". This same reference to balance is found with level changes made in Tests V,VIII and IX through such comments as: "...clearer, and well blended/balanced."; "good depth and solo blend"; "best blend"; "best balance".

Listening Test B, as stated earlier, reveals that time delay compensation will not necessarily improve the recording. In excerpt 4, comments support the addition of spot microphones (though by a small percentage) without the addition of delay..."percussion more defined, violin closer,woodwinds are too present..."; "better images of separate instruments..."; "cleaner in percussion, dry sound"; "...improved placement of violin".

Comments in favour of the addition of the delay to the spot microphones demonstrate that the delay is heard: "definition is there without loss of depth..."not much different from excerpt 4. I preferred 4 but can't pin-point the reason. It was more 'natural', but this sounds a little deeper".

Obviously both points of view are subjective. Long term listening is required in order to better determine the results. The value of time delay compensation as a recording tool is demonstrated by the ability of the listeners to discern an improvement in presence by the addition of spot microphones, an increase in depth and in some cases an overall improvement in balance.

As stated earlier, **Listening Test C** shows positive results for time delay compensation as a useful recording tool where the main pickup alone has captured an already good stereo recording.

Comments reveal that some participants chose main pair no. 2 with spot microphones as an audible improvement over what they considered an already good pickup:

"spot on bass? more realistic, good sound, much clarity-beautiful".

"cello clearer, strings clearer, french horn clearer, piano a bit weak, basses better-more defined"

Other comments reveal that the listeners hear the addition of non-delayed spot microphones and point out both positive and negative effects. This results in varying subjective opinions:

"flatter stage (less sense of depth and space). Piano and bass both more present and up-front. Not as spot-lighted as no. 2 in terms of precise instrument placement, but still easy to determine where each point source is".

"highlights certain strands of the melody but texture is less homogeneous than excerpt no. 2"

"string section is clearer now "

"narrow image-instruments on top of each other"

One listener comments on the effect of comb filtering on the clarinet: "my guess is no delay to spot mics; clarinet shows comb filtering-sounds dull on lower frequencies; some brass floats".

All participants noticed an improvement in depth in excerpt 4 and 90% (including all recording engineers) chose excerpt 4 with time delay compensation as the best choice: "Now we have depth!..."; "best, all timbres are clearer"; "preferable to excerpt 3", "better balanced imaging"; "wider imaging and spatially better"; ".. has the most detail"; "This sounds best, most natural. Good stereo spread, all instruments clear, present and defined, without over emphasis...".

Listening Test D...as stated earlier, reveals that differences were heard by a large majority with very short A/B comparisons The delays are generally preferable to non-delayed spots yet the excerpts are too short for any consistency in the listeners responses. An example of the variety of comments to support this statement is evident from Test 1: "Quite a subtle difference..."; "not much difference...no delay?"; "Can hear slight difference; perhaps lower instruments a bit more prominent in B"; "oboe louder in B"; "more orchestral sound in B".

In conclusion, although there are subjective differences of opinion through the various comments, the results show that the use of compensating time delays 1) adds depth, 2) is perceivable by the participants and 3) an optimum time delay seems preferable when 4 or more spot microphones are used in combination with the main pair. This seems especially true for orchestral recordings where careful choice of spot microphones can improve the balance without destroying depth. There is no consistency in the comments to support Theile's theory of adding an increment to the calculated delay for optimum results. This theory refers to over-compensating (larger) delays in order that the signal arrive at the main microphone in time with the first reflections (as opposed to strictly compensating (exact), or under-compensating (smaller) delays. Long term listening and level changes may also be necessary in making subjective decisions for improving balance.

Chapter 6

SUMMARY AND DISCUSSION

The results of the listening tests show time delay compensation to be a useful tool, with both positive and negative results. From a positive viewpoint, it can provide extra spaciousness, reverb and sustain of the sound by improving the sense of depth. These same positive attributes can become undesirable in certain situations, destroying perspective and manipulating depth by pushing the present elements further back. Although the delay does align the sound at the spot microphone with the arrival time of that sound at the main stereo pair, this may not be desirable or it may not be the most important factor to consider in a given situation. Improving balance, proximity or reverb may be more beneficial to the recording.

6.1 Time Delay Recording Technique: A Tool For Balancing

When combining the output of several microphones, the audibility of comb-filter effects is a major factor in the subjective assessment of a recording. Results from the listening tests confirm that the addition of spot microphones in reinforcing the direct sound can be important for clarity and definition, and there can be an audible improvement in the sound quality and depth perspective with delay compensation of the spot microphones. However, comb filter effects may not be the most important aspect as far as a specific recording is concerned. The spot microphones may be used to enhance proximity or improve the balance without the use of delay compensation.

Time delay recording is important as a recording tool because a noticeable difference can be achieved in the time relation regardless of how desirable the result may be. It is important for the recording engineer to be able to list both positive and negative effects in any given situation before making a decision as to whether or not to use this technique.

As a positive effect, time delay compensation can provide extra spaciousness, reverb, balance and sustain of the sound. However, depending on the recording, it can destroy perspective and manipulate depth by pushing the closest elements further away. In certain situations, extra reflections and reverb can destroy the clarity of the direct sound.

Long term listening is necessary for assessment of balance. Upon hearing more reverb or a brighter sound, our initial reaction may be that this is better without our taking the time to really concentrate on the overall or long term listening effect of the perceived balance. This seems to be the case in reviewing the results of Listening Test A, in comparing the different excerpts to the main pair. Due to the length of the excerpts listeners seemed to immediately respond to spaciousness, brightness and loudness as being immediately preferable. The excerpts may not be long enough to determine the effect of the brightness or other changes in sound quality on the overall balance.

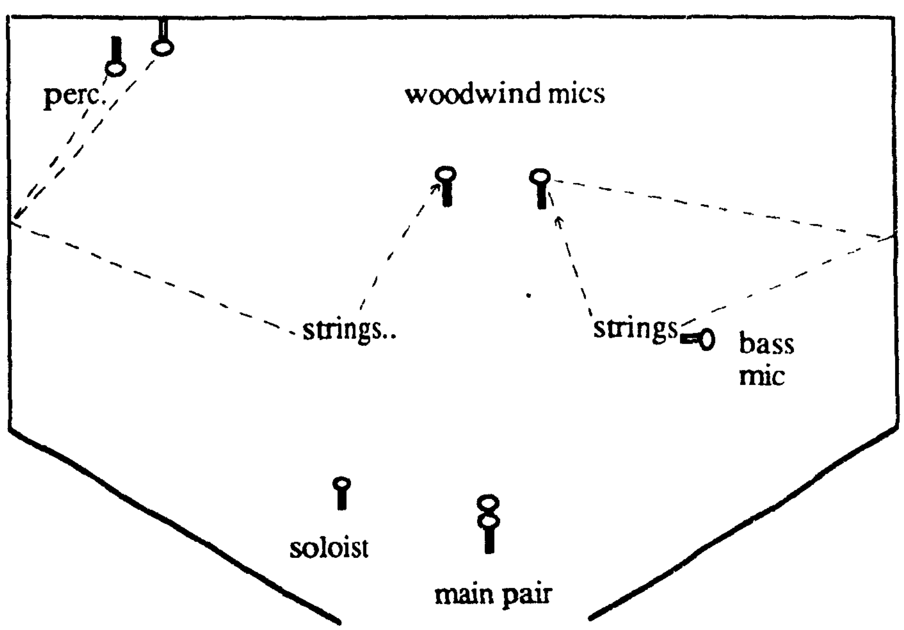
Although the use of a spot microphone may not effectively simulate early reflections, this may not be the intent. In some cases the presence or intimacy achieved through the use of multiple microphones will help in reinforcing the direct sound, especially when the room tone is not desirable. It can improve the dynamic range and the overall balance.

It should be the goal of the engineer to produce a captivating presentation of the music. Although there is no one "right way" to accomplish this, it is important that we learn and experiment with as many recording techniques as possible so that the relevant knowledge and recording tools available will help us, especially in problem situations.

6.2 Potential Problems

One very important discovery that seems to have been overlooked in previous research is the increased effect of reverberation or amount of reflected sound, with the use of time delay compensation. Although we have been dealing mostly with the direct sound, depending on the situation, time delays can have a great deal of effect on the recording due to their potential effect on reflected sound. Thus an optimum delay may only be good for the instrument concerned but damaging for surrounding groups.

It is important to make a decision as to what we are listening for. The Denon anechoic CD experiments show that the technique does work in terms of spaciousness yet may hurt the close perspective of the strings. The use of time delay compensation in an anechoic situation must be regarded as a different kind of tool that is used for a different purpose. We lose the perspective of the music without the interaction of the room It can be seen from the following diagram that although spot microphones are not used specifically on the strings, other spot microphones influence the sound of such a large section. The spot microphones in this case can be thought of as destructive "flashlights" of sound. There is no proof that the use of delays does not extend the reverb Through time delay compensation of the spot microphones, the strings are also delayed, not only by the woodwind microphones, but also by the percussion microphones. In some recordings, again depending on numerous factors, this effect may be more obvious than in other cases.



The quality of the recording seems to be a major factor in determining the validity of the time delays. The results from the Stravinsky recording indicate that the poor quality from the overall pickup was a prime factor in varying subjective impressions.

Mixing the direct sound from multiple microphones will not automatically achieve the proper balance of reverberant sound. This may not be as crucial in the case of main microphone-spot microphone techniques if the reverberant sound is balanced in the main pickup and spot microphones are used for improving clarity and definition of the direct sound.

Further research into the pickup of first reflections at the spot microphone must be investigated and could pose dangerous problems to the sound in certain situations; especially in the case of floor reflections which can be destructive to the timbre of the instrument depending on the room.

According to Anazawa, it is desirable to secure a delay compensation precision of 1 ms or better. In the case of aligning the direct sound from the spot microphone, comb filtering effects may be more detrimental with very short delays (1 ms) , especially with short sounds. Thus time delay compensation would have to be very precise. This can be difficult to ensure due to the many variables that can effect the final measurement. The direct sound may contribute to first reflections which may or may not be beneficial depending on the sound quality of the room.

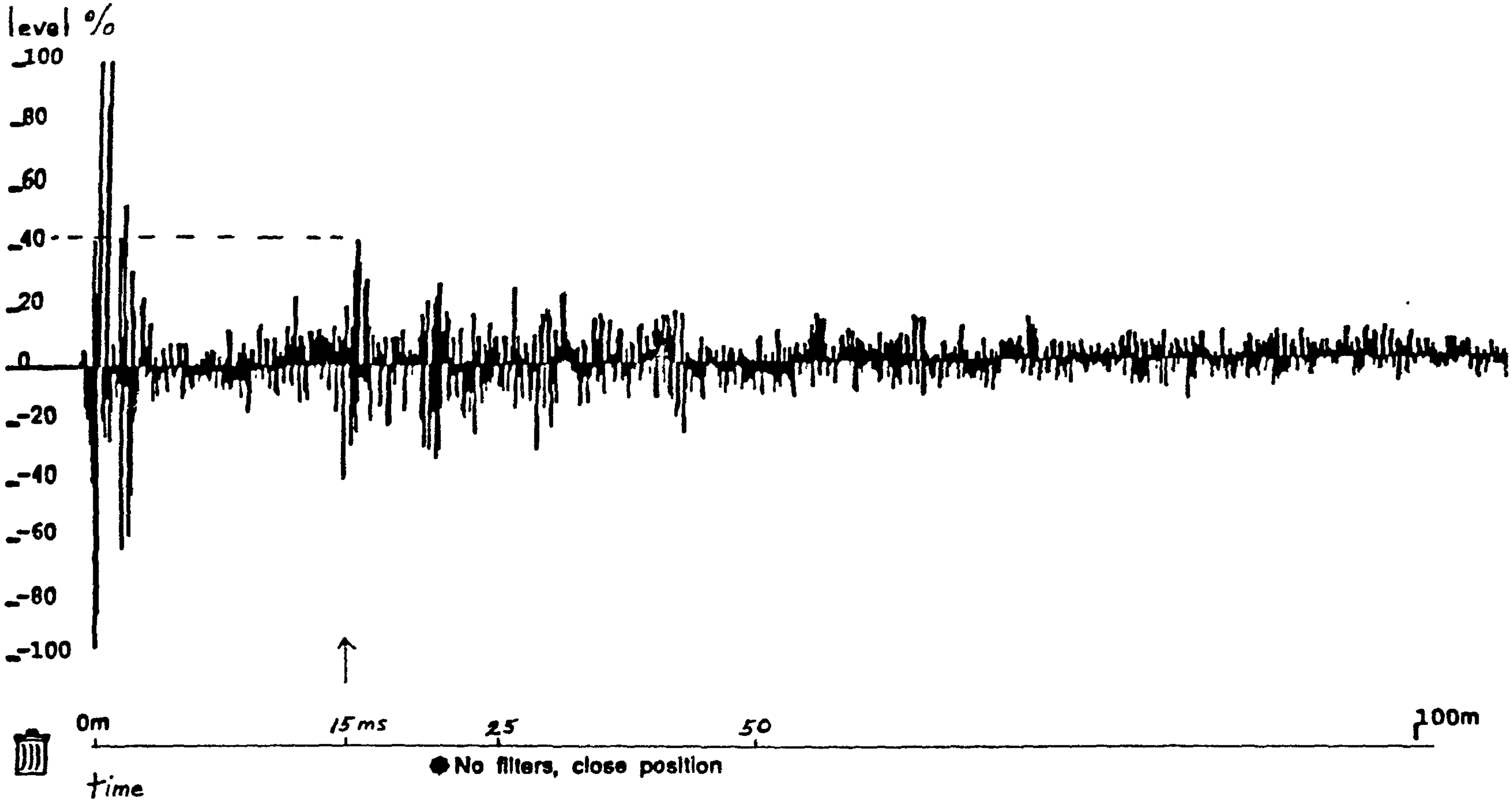
The potential problem in sending the signal through the A/D and D/A filters in digital delays is most important for fidelity of the direct sound. The accuracy of A/D conversion is the heart of the digital recording process; the quality of the delay unit is most important. There is the danger of overloading the delay unit depending on the dynamic range of the direct sound, if the device is not properly set. Although there should be no effect on the frequency response,the transients may cause overshoot and ringing that may occur when the sound passes through the converters.

6.3 Future Considerations

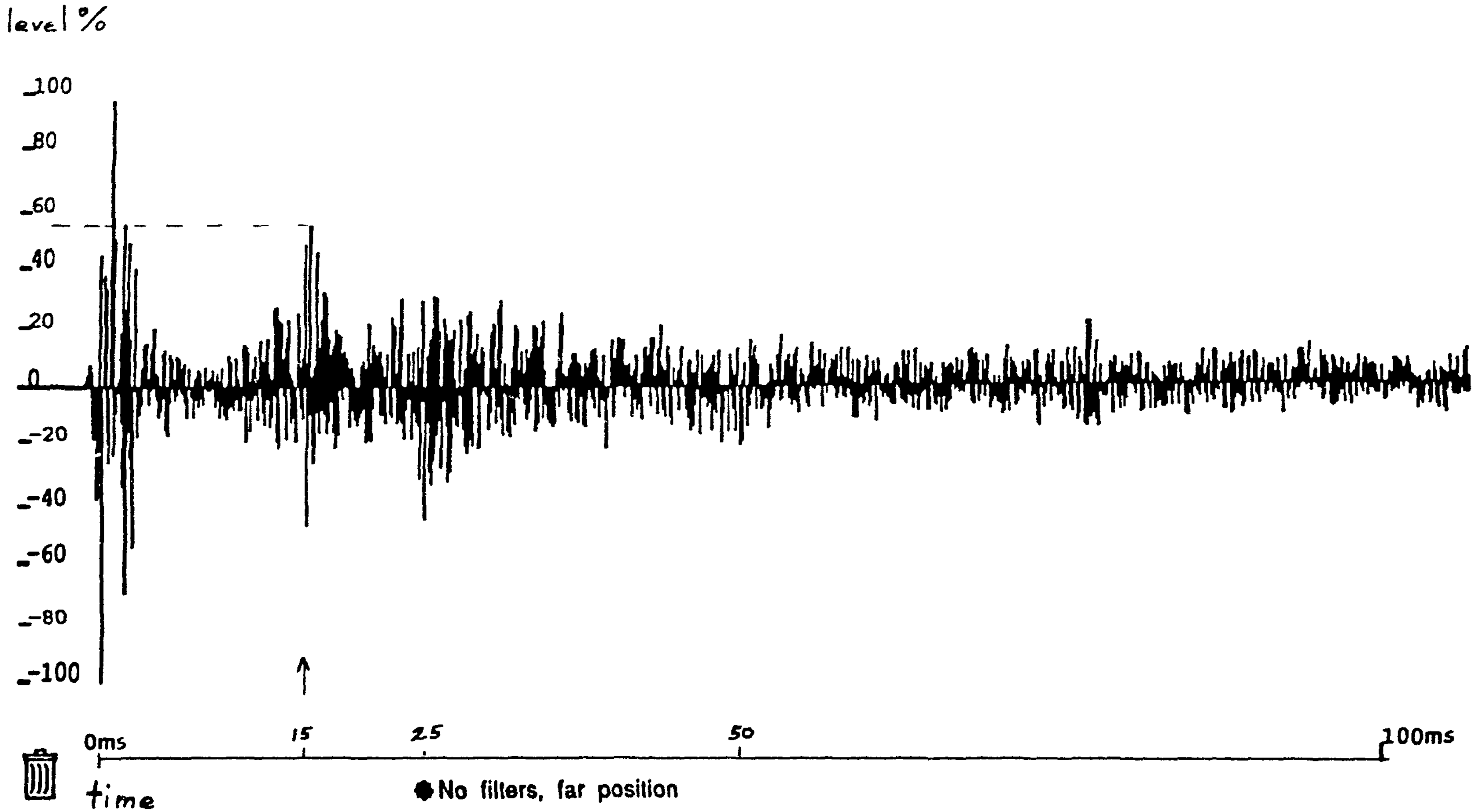
As stated earlier, the room plays an important role in carrying timbral information. Room response problems must therefore be investigated as a major factor in determining the validity of compensating time delays. Obviously precise calculations are only valid if there are no inherent delays of the sound because of room response. I have included two graphs that were created using Sound Designer II software[18]. They represent the response of the Project Studio to a one millisecond impulse of pink noise sent through two speakers placed back to back to capture the full sphere of sound radiation in the room. Two B&K microphones (4007) are used to capture the response of the room to the pink noise. This test was done following the Stravinsky: L'histoire du Soldat recording in the Project Studio at the Banff Centre. The speakers were placed in the approximate location of where the clarinetist sat. The far microphone position is in the location of the main stereo pair (3.5 meters) with the close position moving the second B&K microphone one-half meter closer

The results reveal an inherent delay already present in the room at 15 ms where the signal appears as high as 50% of the level of the impulse. This is especially noticeable in the far position.(graph #2)

[18] Sound Designer II software, Version 2.0 ©1990 by Digidesign Inc.



1. Project Studio room response-close mic. position



2. Project Studio room response-far mic. position

In the future this technique may become a more flexible tool in sound recording by being a built-in function on a recording console. In this manner spot microphones will provide not only the possibility of helping balance through changes in level and equalization, but also in time relationships-influencing both the direct sound and reverb-thus adding a new dimension in the balancing process.

Adjustments in perspective, imaging, loudness and balance are possible through easily accessible changes in the time relationship of the spot microphones to the main stereo pair This requires critical listening skills on the part of the sound engineer. Perhaps the possibility of the opposite relationship-in delaying the main pair to emphasize the importance of the spot microphones, would be beneficial in another situation. With so many factors contributing to the overall sound, the ear must be the ultimate judge.

Finally, these tests have convinced me of the advantages of using multiple microphones as a tool for improving balance. This technique is not meant to replace other microphone techniques but can be viewed as yet another method which can give very good results in some applications.

I have made reference to many recording techniques, new discoveries and suggestions for improvement through careful analysis throughout this paper. Without going into detail or having tried to prove any one technique as being superior to another, I hope I have demonstrated the way each has to be considered as a valid recording tool in the art of sound recording.

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APPENDIX A

LISTENING TEST A

I.Mozart Oboe Concerto: Woodwind spot microphones (**listen for horn entries**)
8 excerpts (all odd numbered excerpts are the same...comment on the differences you hear with the even numbered excerpts which include the **woodwind spot microphones** in combination with the main pair)
Please rate the even numbered excerpts to the main pair :
1=same or little difference
2=better
3=worse
4=best(if applicable)

DAT
Index #'s

1 1. main pair reference

2 2. comments:

3 3. main pair reference (**same as 1**)

4 4. comments:

5 5. main pair reference

6 6. comments:

7 7. main pair reference

8 8. comments:

II. Mozart Oboe Concerto: 7 excerpts
String Spot Microphones

9 1. main pair reference

10 2. comments:

11 3. main pair reference (**same as 1**)

12 4. comments:

13 5. main pair reference

14 6. comments:

15 7. main pair reference

III. Mozart Oboe Concerto: 6 excerpts

Woodwind and String Spot Microphones

16 1. main pair reference

17 2. comments:

18 3. main pair reference

19 4. comments:

20 5. main pair reference

21 6. comments:

IV. Mozart Oboe Concerto: 2 excerpts

2 woodwind, 1 bass, 1 solo (oboe) spot microphones (main pickup included in both)

22 1. comments:

23 2. comments:

V. Bruckner: Symphony No. 7(not 3 as stated on the tape): 6 excerpts

2 woodwind, 1 bass, 1 percussion spot microphones

24 1. main pair only

25 2. comments:

26 3. main pair only

27 4. comments:

28 5. main pair only

29 6. comments:

VI. Bruch: Violin Concerto: 5 excerpts

Solo Violin Spot Microphone

30 1. main pair only

31 2. solo mic added; comments:

32 3. solo added; comments:

33 4. solo added; comments:

34 5. solo added; comments:

VII. Bruch: Violin Concerto: 5 excerpts

Woodwind Spot Microphones: all include main pair

35 1. comments:

36 2. comments:

37 3. comments:

38 4. comments:

39 5. comments:

VIII. Bruch: Violin Concerto: 3 excerpts

2 woodwind, 1 bass, 1 percussion spot microphones...

..all include main pair

40 1. comments:

41 2. comments:

42 3. comments:

IX. Mahler: Symphony No. 1: 4 excerpts

4 percussion spot microphones

43 1. main pair only

44 2. spot mics added; comments:

45 3. spot mics added; comments:

46 4. spot mics added; comments:

X. Mahler: Symphony No. 1: 4 excerpts

2 woodwind, 1 bass, 1 timpani spot microphones

47 1. main pair only

48 2. spot mics added; comments:

49 3. main pair only

50 4. spot mics added; comments:

LISTENING TEST B

DIFFERENCES IN SOUND RECORDING METHOD

I. The following 5 excerpts are used to demonstrate the sound quality obtained by using various miking techniques and to clarify the effects of delay compensation. I have included five excerpts from Stravinsky's L'histoire du soldat, using 3 miking techniques plus auxiliary microphones and delay compensation effects.

51 1. **Stereo Microphone X/Y** recording technique (level difference); comments:

52 2. **Blumlein** recording technique (level difference); comments:

53 3. **Dummy Head** technique (time difference); comments:

54 4. **Dummy Head with spot microphones**; comments:

55 5. **Dummy Head with spot microphones**; comments:

LISTENING TEST C

56 1. main pair no.1; comments:

57 2. main pair no.2; comments:

58 3. main pair with spots; comments:

59 4. main pair with spots; comments:

LISTENING TEST D

This test is based on an A-B comparison test. The excerpts are short. Note any change you hear between the two.

Pause Dat Tape Between Excerpts for more time.....you may use the index numbers to repeat these short excerpts

60 1. comments:

61 2. comments:

62 3. comments:

63 4. comments:

64 5. comments:

65 6. comments:

66 7. comments:

NOTE: excerpt no.8 incorporates the main pair and spot microphones in both A & B!

67 8. comments:

APPENDIX B

CONDITIONS

LISTENING TEST A

I.Mozart Oboe Concerto: Woodwind spot microphones (listen for horn entries)
8 excerpts (all odd numbered excerpts are the same...comment on the differences you hear with the even numbered excerpts which include the woodwind spot microphones in combination with the main pair)

1. main pair reference-AKG 414's (Blumlein)

2. comments: -no delay

3. main pair reference (same as 1)

4. comments: -calculated delay (18 ms)

5. main pair reference

6. comments: -shortened delay by 2 ms (16 ms)

7. main pair reference

8. comments: -longer delay (21 ms)

II. Mozart Oboe Concerto: 7 excerpts
String Spot Microphones

1. main pair reference

2. comments: -no delay

3. main pair reference (same as 1)

4. comments: -calculated delay (19/16 ms)

5. main pair reference

6. comments: -shorter delay (16/13 ms)

7. main pair reference

III. Mozart Oboe Concerto: 6 excerpts

Woodwind and String Spot Microphones

1. main pair reference

2. comments: -no delay added to 4 spot microphones

3. main pair reference

4. comments: -calculated delay for spots

5. main pair reference

6. comments: -calculated delay with strings 2 db lower level

IV. Mozart Oboe Concerto: 2 excerpts
2 woodwind, 1 bass, 1 solo (oboe) spot microphones (main pickup included in both)

1. comments: -no delay to all 4 spot microphones

2. comments: -calculated delay for all 4 spot microphones (see table 1)

V. Bruckner: Symphony No. 7(not 3 as stated on the tape): 6 excerpts
2 woodwind, 1 bass, 1 percussion spot microphones

1. main pair only -AKG 414's (Blumlein)

2. comments: -no delay on 4 spot microphones

3. main pair only

4. comments: -calculated delay on all spots (see table 1)

5. main pair only

6. comments: -delayed with change in level for bass and percussion spot (bass up 1 db: per: down 2 db)

VI. Bruch: Violin Concerto: 5 excerpts
Solo Violin Spot Microphone

1. main pair only -AKG 414's (ORTF)

2. solo mic added; comments: -no delay

3. solo added; comments: -calculated 12 ms delay

4. solo added; comments: -10 ms delay

5. solo added; comments: -15 ms delay

VII. Bruch: Violin Concerto: 5 excerpts
Woodwind Spot Microphones: all include main pickup

1. comments: -no delay to spot microphones

2. comments: -19 ms delay (19.5ms actual calculated delay)

3. comments: -20 ms delay

4. comments: -23 ms delay

5. comments: -15 ms delay

VIII. Bruch: Violin Concerto: 3 excerpts
2 woodwind, 1 bass, 1 percussion spot microphones...all include main pair

1. comments: -no delay to spots

2. comments: -spots added with calculated delays (see table 2)

3. comments: -same as #2; yet woodwind spots down 2 db in level/ bass up 1 db

IX .ıler: Symphony No. 1: 4 excerpts
4 percussion spot microphones

1. main pair only -AKG 414's (ORTF)

2. spot mics added; comments: -no delay

3. spot mics added; comments: -calculated delays (see table 2)

4. spot mics added; comments: -calculated delays but lower level by 2 db.

X. Mahler: Symphony No. 1: 4 excerpts
2 woodwind, 1 bass, 1 timpani spot microphones

1. main pair only

2. spot mics added; comments: -no delay

3. main pair only

4. spot mics added; comments: -calculated delays (table 2)

LISTENING TEST B

DIFFERENCES IN SOUND RECORDING METHOD

I. The following 5 excerpts are used to demonstrate the sound quality obtained by using various miking techniques and to clarify the effects of delay compensation. I have included five excerpts from Stravinsky's L'histoire du soldat, using 3 miking techniques plus auxiliary microphones and delay compensation effects.

1. Stereo Microphone X/Y recording technique (level difference); comments:

2. Blumlein recording technique (level difference); comments:

3. Dummy Head technique (time difference); comments:

4. Dummy Head with spot microphones; comments:-no delay

5. Dummy Head with spot microphones; comments: -calculated delays (see table3)

LISTENING TEST C

1. main pair no.1 -Dummy Head (using B&K 4007 microphones at ears)

2. main pair no.2 -AKG C-422 Stereo microphone

3. main pair with spots -AKG Stereo microphone with spot mics. (no delay)

4. main pair with spots -AKG Stereo microphone with spot mics. (calculated delays-see table 4)

LISTENING TEST D

This test is based on an A-B comparison test.

1. comments: A-main pair; B-with woodwind spots (no delay)

2. comments: A-main pair; B-with delayed woodwind mics.

3. comments: A-main pair; B-string spot microphones (no delay)

4. comments: A-main pair; B-woodwind, bass and solo spot microphones (calculated delays)

5. comments: A-main pair; B-woodwind, bass & percussion spot mics. added (calculated delays)

6. comments: A-main pair; B-solo mic added (no delay)

7. comments: A-main pair only
   B-main pair with spot microphones (calculated delays)

NOTE: excerpt no.8 incorporates the main pair and spot microphones in both A & B!

8. comments: A-main pair with spot microphones (no delays)
   B-main pair with spot microphones (calculated delays)

APPENDIX C

RESULTS-PREFERENCES AND COMMENTS
LISTENING TEST A

I.Mozart Oboe Concerto: Woodwind spot microphones (listen for horn entries)
8 excerpts (all odd numbered excerpts are the same...comment on the differences you hear wi. the even numbered excerpts which include the woodwind spot microphones in combination with the main pair)

1. main pair reference-AKG 414's (Blumlein)

2. comments: -no delay

3. main pair reference (same as 1)

4. -calculated delay (18 ms) ...13%
   comments:
5. main pair reference

6. -shortened delay by 2 ms (16 ms)...20%
   comments:
7. main pair reference

8. -longer delay (21 ms)...67%
   comments: "French horns rounder, oboe fine here"....recording engineer
   "Horns sound "best" here, but may be too prominent (as in loud), but finest sound." .....recording engineer
   "less present....horns further back....most delay?".....sound engineer
   "more ambiance with horns, low strings good"....musician
   "brass section should be more prominent".......composer
   "more strings than before?...this is the best natural balance"....composer

II. Mozart Oboe Concerto: 7 excerpts
String Spot Microphones
..........too strident....5%
   "strings sound too strident throughout all examples for my taste"....composer
1. main pair reference

2. comments: -no delay....
   "strings advanced"...composer
   "enhances the viola line but not advantageous to others"...composer
   "horns louder because of spot mics"...recording engineer

3. main pair reference (same as 1)

4. -calculated delay (19/16 ms)......55%
   comments: "louder string tutti's, almost distorted; more depth with first cello"....musician
   "flatter dimension"...recording engineer
   "delayed string spots seem to make oboe recede, adds depth"..recording eng.

5. main pair reference

6. -shorter delay (16/13 ms)....40%
comments: "most definition...after hearing the delay in no. 4 this would seem the flattest sound".....recording engineer
"good balance"....composer
"better balance"....sound engineer
"clearer in lower strings (viola, cello)"...composer
"very slight difference again, but strings sound "muddled"...individual instruments less distinct. Not as clean or "good" a sound as no. 4.

7. main pair reference

III. Mozart Oboe Concerto: 6 excerpts

Woodwind and String Spot Microphones

1. main pair reference

2. -no delay added to 4 spot microphones
comments: "french horns present, strings closer, not much depth"..recording engineer
"brighter, much more definition of french horn, clarity of oboe"...composer

3. main pair reference

4. -calculated delay for spots.....20%
comments: "more depth, definition of strings very good"...recording engineer
"like no. 2 but strings seem better placed, integrated into overall pickup"..recording eng.
"noticeable delay in string mics."...recording engineer

5. main pair reference

6. -calculated delay with strings 2 db lower level.........80%
comments: "like no.4 but better balance"....recording engineer
"less woodwind/horns, oboe sounds better this time; nicer balance between sections"...composer
"better definition but flatter depth of field"...recording engineer

IV. Mozart Oboe Concerto: 2 excerpts
2 woodwind, 1 bass, 1 solo (oboe) spot microphones (main pickup included in both)

1. -no delay to all 4 spot microphones...23%
comments:
"winds (and brass) are better integrated with strings in excerpt no.1
-in terms of space and timbre as well; strings sound lighter"...recording eng.

2. -calculated delay for all 4 spot microphones (see table 1).....77%
comments: "has better depth of field-but some funny ambience slapping

(delays?)"..recording eng.
"best, hear more of the high strings"...composer
"we hear the bass better"...composer
"oboe reclines-distracting"...recording engineer
"no. 2 has more clarity and spaciousness and sounds better"...recording eng.
"oboe floats, horns are louder, strings are more defined but not better"...recording eng.

V. Bruckner: Symphony No. 7(not 3 as stated on the tape): 6 excerpts
2 woodwind, 1 bass, 1 percussion spot microphones

1. main pair only -AKG 414's (Blumlein)

2. comments: -no delay on 4 spot microphones

3. main pair only

4. -calculated delay on all spots (see table 1)....25%
comments:"Woodwinds much better. Nicely imaged and balanced. Bottom too heavy but not so muddy"...recording engineer
"Woodwinds sound "behind" the rest of the orchestra in terms of space, but timbre is nice"...recording engineer

5. main pair only

6. -delayed with change in level for bass and percussion spot (bass up 1 db: perc. down 2 db) 75%
"more of no. 4 (clarinet and flute more defined, trumpet richer, strings round)ith percussion focused now"...recording engineer
"heard the horn this time"...musician
"cleaner, clearer"...composer
"bottom end and timps best in this version. Clearer, and well blended/balanced.Woodwinds now sound too dry and forward"...rec.eng.
"tighter low end"...recording engineer

VI. Bruch: Violin Concerto: 5 excerpts
Solo Violin Spot Microphone

1. main pair only -AKG 414's (ORTF).........07% doubling unpleasant yet #5 more like chorus

2. solo mic added; comments: -no delay......07%
"solo mic causes vln. to move to centre; space is different for the solo compared to the orchestra"...recording engineer
"rosin noise distracting from soloist"...recording engineer
"this seems to have the liveliest sound"...musician

3. solo added; comments: -calculated 12 ms delay...07%
"violin too present"...same comment by two recording engineers
"this sounds a little unnatural- hard to say why-confused violin positioning?"...recording engineer

4. solo added; comments: -10 ms delay.....64%
"sound of the violin is somewhat between Ex. 2 and 3 (both in terms of timbre and spaciousness; orchestra gain, depth and richness is more integrated with the soloist"...recording engineer
"this sounds very similar to no. 2 (violin richer, bigger, and even more forward: but I like this better)-coherent image"...recording engineer
"image stable for violin, now pizz. is really audible in orchestra, violin rounder"...recording engineer
"more resonance, more 'airy'"...musician
"more clarity for woodwinds" ..recording engineer

5. solo added; comments: -15 ms delay ... 5%
"vln tone rounder-and clearer"...composer
"vln tone seems more rounded"...composer
"doubling less "unpleasant" here compared to excerpts 3 and 4; almost like chorusing...recording engineer
"lower strings get lost; orchestra is not as defined" ..recording engineer

VII. Bruch: Violin Concerto: 5 excerpts
Woodwind Spot Microphones: all include main pickup

1. comments: -no delay to spot microphones..... 15%

2. comments: -19 ms delay (19.5ms actual calculated delay)...20%
"woodwinds clearer than no. 1"...composer
"more of brass and timpani"...composer
"woodwinds sound 'supported' by spot microphones"...recording engineer

3. comments: -20 ms delay

4. comments: -23 ms delay.....45%
"airy violin, orchestra too boomy"...musician
"best woodwind balance"...musician
"best balance here; percussion good"...recording engineer
"I like this balance between clarinet / vln. soloist"...recording engineer
"'slap' of delay not so noticeable here"...recording engineer

5. comments: -15 ms delay....20%
   "woodwinds increasingly more present (no. 3-5)"...composer
   "a bit muddy"...composer
   "like no. 4 (winds well integrated and positioned, still very focused and clear)-but a little better; I like the sound and balance of this one the best"...recording engineer
   "Balance of all groups best here"...recording engineer
   "flute too prominent, clarinets lost, brass loud"...recording engineer

VIII. Bruch: Violin Concerto: 3 excerpts
2 woodwind, 1 bass, 1 percussion spot microphones...all include main pair

1. comments: -no delay to spots-10%
   "most appealing of three excerpts"...composer
   "best"...composer

2. comments: -spots added with calculated delays (see table 2)...10%

3. comments: -same as #2; yet woodwind spots down 2 db in level/ bass up 1 db...80% all engineers included here
   "better than no 2; especially percussion"...recording engineer
   "good depth and solo blend"...recording engineer
   "I like this best; winds natural and not too forward-well blended. Bass is weak though and timps a little muddy".. recording engineer
   "still not a lot of bass definition; clarinet and timpani better here"...recording eng.
   "best blend"...recording engineer
   "depth of field seems best here but left heavy ambiance"...recording eng.

IX. Mahler: Symphony No. 1: 4 excerpts
4 percussion spot microphones

1. main pair only -AKG 414's (ORTF).....15%
   "lower timpani very close but I liked this excerpt best overall"...recording engineer
   "In terms of volume and spatial balance. I definitely prefer the main pickup; in terms of percussion timbre and sound, excerpt no. 4 seems to best support the percussion instruments, but they are way in front of the rest of the orchestra and much louder"... ..recording engineer
   "best-rest have unnatural close percussion sound which takes away from the rest of the orchestra"...composer

2. spot mics added; comments. -no delay...m

3. spot mics added; comments: -calculated delays (see table 2)...10%

4. spot mics added; comments: -calculated delays but lower level by 2 db. ..75%
   "better percussion depth of field-this actually sounds better than the main pair" ...recording engineer
   "best balance" ..recording engineer
   "this sounds best: percussion clean and defined, better placed-not so forward"...recording engineer
   "this is preferable, sound is not choked" ..recording engineer
   "percussion seems most integrated"...composer

X. Mahler: Symphony No. 1· 4 excerpts
2 woodwind, 1 bass, 1 timpani spot microphones

1. main pair only

2. spot mics added; comments: -no delay..40%
   "bass and timpani clearer-best"...composer

3. main pair only

4. spot mics added; comments: -calculated delays (table 2)...60%
   "stronger horn section, stronger crescendo in timpani"...musician
   "less presence than excerpt 2"...musician
   "good balance, more definition"...recording engineer
   "definitely the best sound. Timpani improved, bass still weak". .recording engineer
   "bass sounds up front, I like that french horns sound chorus-y from the delay".. recording engineer
   "more woodwinds and timpani" .composer
   "woodwinds are less present than in excerpt 2 but still more present than in excerpt 1. Bass section sounds more substantial than in both excerpts 1 and 2 , timpani similar to 2 but clearer. I would choose no. 3"...recording engineer
   "best-timpani better in both excerpts 2 and 4-bass too close?"...recording engineer

LISTENING TEST B

DIFFERENCES IN SOUND RECORDING METHOD

I. The following 5 excerpts are used to demonstrate the sound quality obtained by using various miking techniques and to clarify the effects of delay compensation. I have included five excerpts from Stravinsky's L'histoire du soldat, using 3 miking techniques plus auxiliary microphones and delay compensation effects.

1 Stereo Microphone X/Y recording technique (level difference); comments:
"small sound stage, not much depth, dry"...recording engineer

2. Blumlein recording technique (level difference); comments: 15%
"good stereo image, violin a bit distant"...composer
"horns more present, more depth, small sound stage"...rec. eng.
"I like this better than excerpt 1-less artificial sounding stereo". .rec. eng.

3. Dummy Head technique (time difference), comments. 5%...but notices depth in 5
"better than excerpt 1 but not so nice as 2"...composer

4. Dummy Head with spot microphones, comments:-no delay.....50%
"percussion more defined, violin closer, woodwinds are too present, stereo picture slightly distorted, my guess is spot miking with no delay"..recording engineer
"better than excerpt no. 3" ..composer
"better images of separate instruments"...musician
"very wide yet percussion better"...recording engineer
"cleaner in percussion, dry sound"...composer
"similar to excerpt no. 3 (flat image), but improved placement of violin. Also seems like nore depth on percussion and brass. Quite likeable" .recording engineer
"excerpt no. 4 is best followed by no 2; woodwinds too distant in all selections (especially bassoon), but clarinet and bass are better in 4". recording engineer

5. Dummy Head with spot microphones; comments: -calculated delays (see table 3)........40%
"definition is there without loss of depth, timbre of instruments is not distorted by close mics with delay-level is crucial"...recording engineer
"less effective than excerpt no. 4"...musician

"not much different from excerpt 4. I preferred 4 but can't pinpoint the reason. It was more 'natural', but this sounds a little deeper"...recording engineer
"best- appropriate for this music"...composer/musician

LISTENING TEST C

1. main pair no.1 -Dummy Head (using B&K 4007 microphones at ears)
   "piano, bass and violin sound 'out of place', too far back. In general, the stereo image is indistinct, hard to tell where each instrument is" .recording engineer

2. main pair no.2 -AKG C-422 Stereo microphone
   "coincident? No separation, but good frequency response". .recording eng
   "muc better sense of space and depth than no 1 Strings better defined, piano more present. Stereo image more coherent, very precise placement of sound sources, also good balance between instruments. Sound clear and natural" recording engineer

3. main pair with spots AKG Stereo microphone with spot mics (no delay) 10%
   "spot on bass? more realistic, good sound, much clarity-beautiful" composer
   "flatter stage (less sense of depth and space). Piano and bass both more present and up-front. Not as 'spot lighted' as no. 2 in terms of precise instrument placement, but still easy to determine where each point source is" recording engineer
   "my guess is no delay to spot mics, clarinet shows comb filtering-sounds dull on lower frequencies, some brass floats" rec. eng
   "cello clearer, strings clearer, french horn clearer, piano a bit weak, basses better-more defined". composer
   "string section is clearer now" . composer
   "highlights certain strands of the melody but texture is less homogenous than excerpt 2"...musician
   "narrow image-all instruments on top of each other". rec. eng

4. main pair with spots -AKG Stereo microphone with spot mics (calculated delays-see ta! e 4)......90%
   "now we have depth! Clarinet is distant but defined, frequency response seems accurate. Wish main pair had been omnis-coincident doesn't showcase this demonstration as well" .recording engineer
   "best, all timbres are clearer". composer
   "preferable to excerpt 3". .composer/musician
   "better balanced imaging"...musician
   "wider imaging and spatially better"...recording engineer
   "low oboe or bassoon is louder, which is nicer-(unpleasant room sound on oboe later)-has the most detail"...recording engineer
   "This sounds best, most natural. Good stereo spread, all instruments clear, present and defined, without over emphasis. A bit of "build up" in the middle?". rec. eng.

LISTENING TEST D

1. comments: A-main pair; B-with woodwind spots (no delay)...........100% hear differer.t
"No depth in A"...composer
"Quite a s/-btle difference. Winds a little more defined in the B excerpt, but sound too dry for the strings (i.e. the winds seem forward in comparison with the strings)"...recording eng.
"Liked the bit more french horn in D"...rec. eng.
"not much difference...no delay?".. recording engineer
"Woodwinds better in B".. composer
"Can hear slight difference; perhaps lower instruments a bit more prominent in B".. composer
"oboe louder in B"...recording engineer
"more orchestra sound in B"...composer

2. comments. A-main pair; B-with delayed woodwind mics
a.........20% prefer main pair
"string section too present in B"...composer
"prefer A"...composer
B..........30% prefer delayed woodwinds
"B has delay; more depth"...Andre
"woodwind delay? French horn sounds nice, deep"...recording engineer
"improved definition and tone of solo oboe in B. Strings sound muddled compared to main pair reference. Hor 's and winds may be a bit too loud?"..rec. eng.
"better depth in B"..composer
50% unsure
"sound is the same"...rec. eng.
"hear no change"...composer

3. comments: A-main pair, B-string spot microphones (no delay)..........80%
"B-better base response"..musician
"B-woodwinds closer"...rec. eng.
"B-has close mics-no delay"...rec. eng.
"B-audible chorusing from string spots-delay?"...rec. eng.
"B-can't really hear a difference Except the violins (hard left!) may be clearer and more defined in B" ..rec eng.

?..20%
"A and B same"...composer
"no difference". .composer
"first a bit bright? Left speaker emphasized"...composer

4 comments: A-main pair; B-woodwind, bass and solo spot microphones (calculated delay;
B..70%
A 20%
?(unsure)10%

"solo oboe more present"...composer
"not much difference. Strings a bit richer, smoother, better blended in B"..rec. eng.
"horns spot microphones-no delay? Didn't mind the up frontness"..rec. eng.
"B has delay"...recording engineer
"too much delay"...composer
"oboe and strings stronger in B"...rec. eng.

"too much delay"...musician
"second may be slightly preferable to first"...composer

5. comments: A-main pair; B-woodwind, bass & percussion spot mics. added (calculated delays) ....90% (10%-?)
"b is better"...musician
"b-woodwinds and horns stronger" rec eng.
"b has no delay". rec. eng
"woodwind lines clearer in b"...rec. eng
"unsure"...rec. eng
"perc. louder in b". .composer
"b-brass, perc. and bass delayed"...composer
"slightly more detail in b" ..composer

6. comments: A-main pair, B-solo mic added (no delay)...........60%
hear difference...30%; no difference.. 10%
"can't hear difference although in B vln may be highlighted somewhat" . composer
"in A french horn (brass) better"...composer
"a-very distant; b-closer" composer
"better solo violin sound in B, but sounds too present and up front, also too dry. Overal better sound, though"...rec. eng
"violin spot not blended" .. rec. eng
"B-no delay; timbre is affected" ..rec. eng.
"violin closer".. composer
"B-solo stronger (different space); 2nd violins stronger; violin solo-middle"..rec. eng
"more string sound"...musician

7. comments: A-main pair only. 10%
B-main pair with spot microphones (calculated delays) .30%
no dif..60%
"seems to have less presence". musician
"no difference"...rec eng
"little difference"...composer
"no difference" .rec. eng.
"?"..rec. eng
"no difference". .composer
"B more balanced"..composer

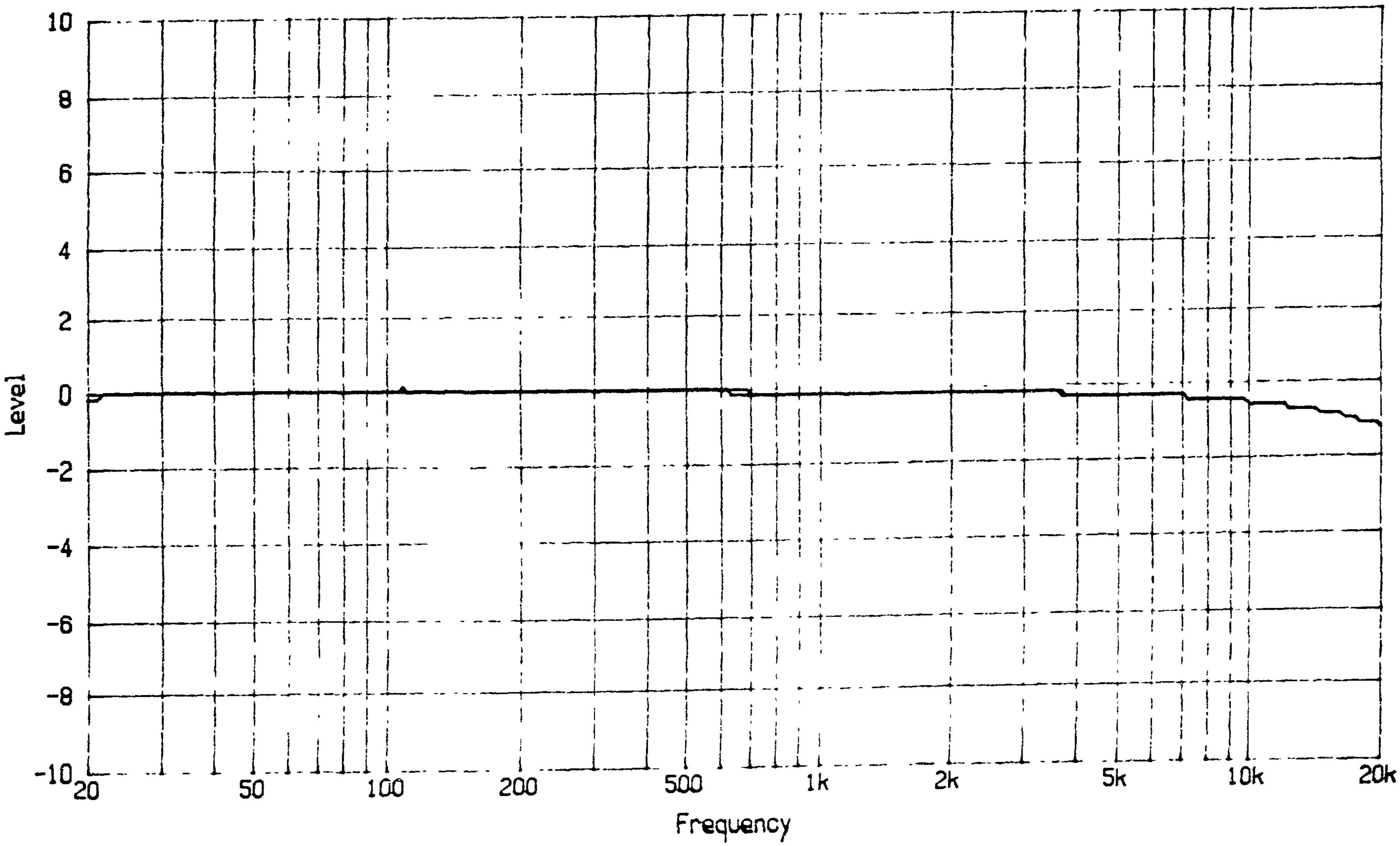
NOTE: excerpt no.8 incorporates the main pair and spot microphones in both A & B'

8. comments. A-main pair with spot microphones (no delays)..35%
B-main pair with spot microphones (calculated delays)...35%
hear difference...30%
"A has more depth than B".. composer
"B-instruments are more separated"...composer
"solo inst seem not as intrusive in B" ..composer
"A-could have used more piano; B-liked the harp spot"...rec. eng
"B-close mics-no delay"...rec. eng
"B-sounds more distant but also more dramatic"...composer
"violin, harp, piano-stronger, closer"...rec. eng.
"better strings in A"...musician

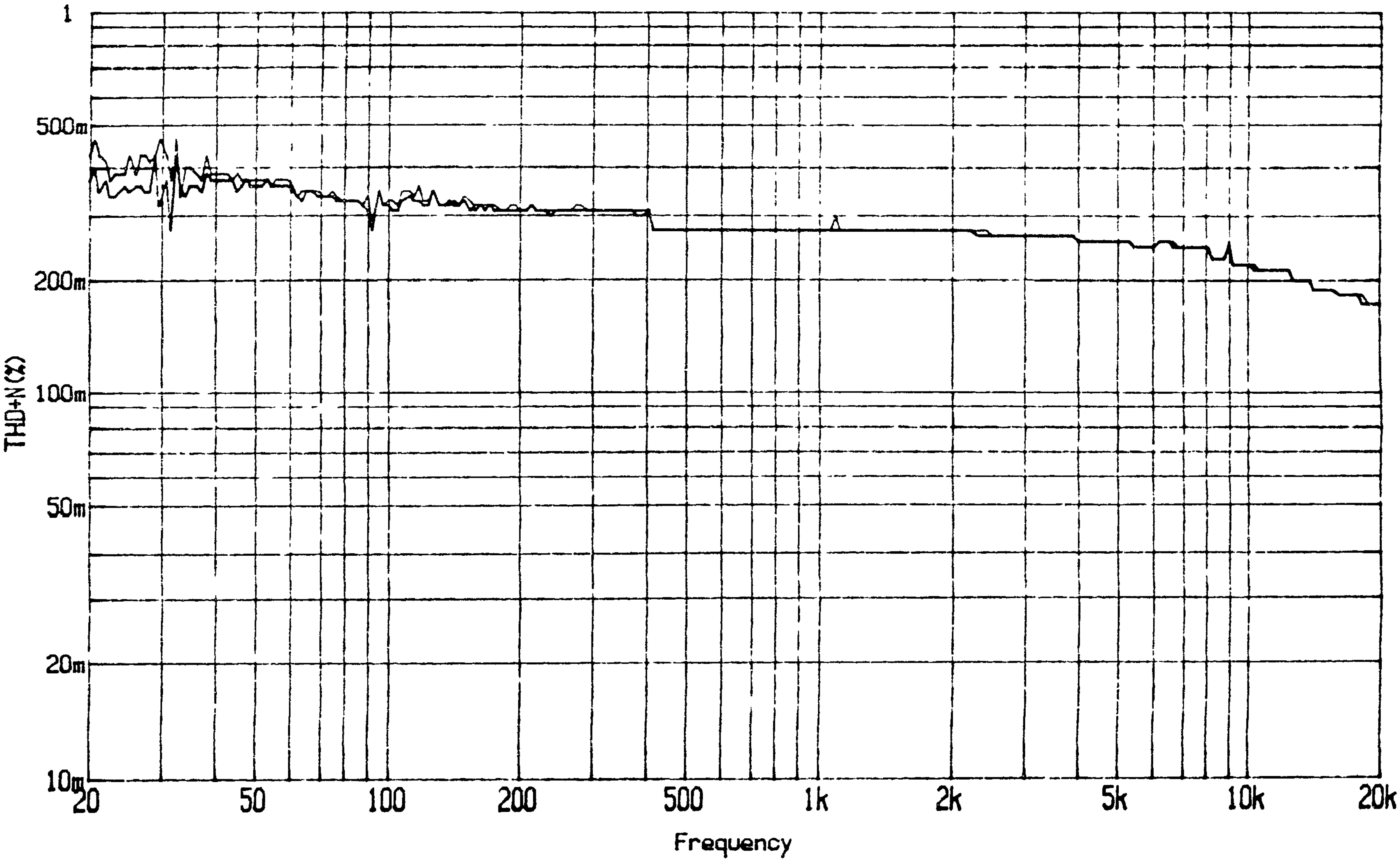
APPENDIX D

LIST OF GRAPHS

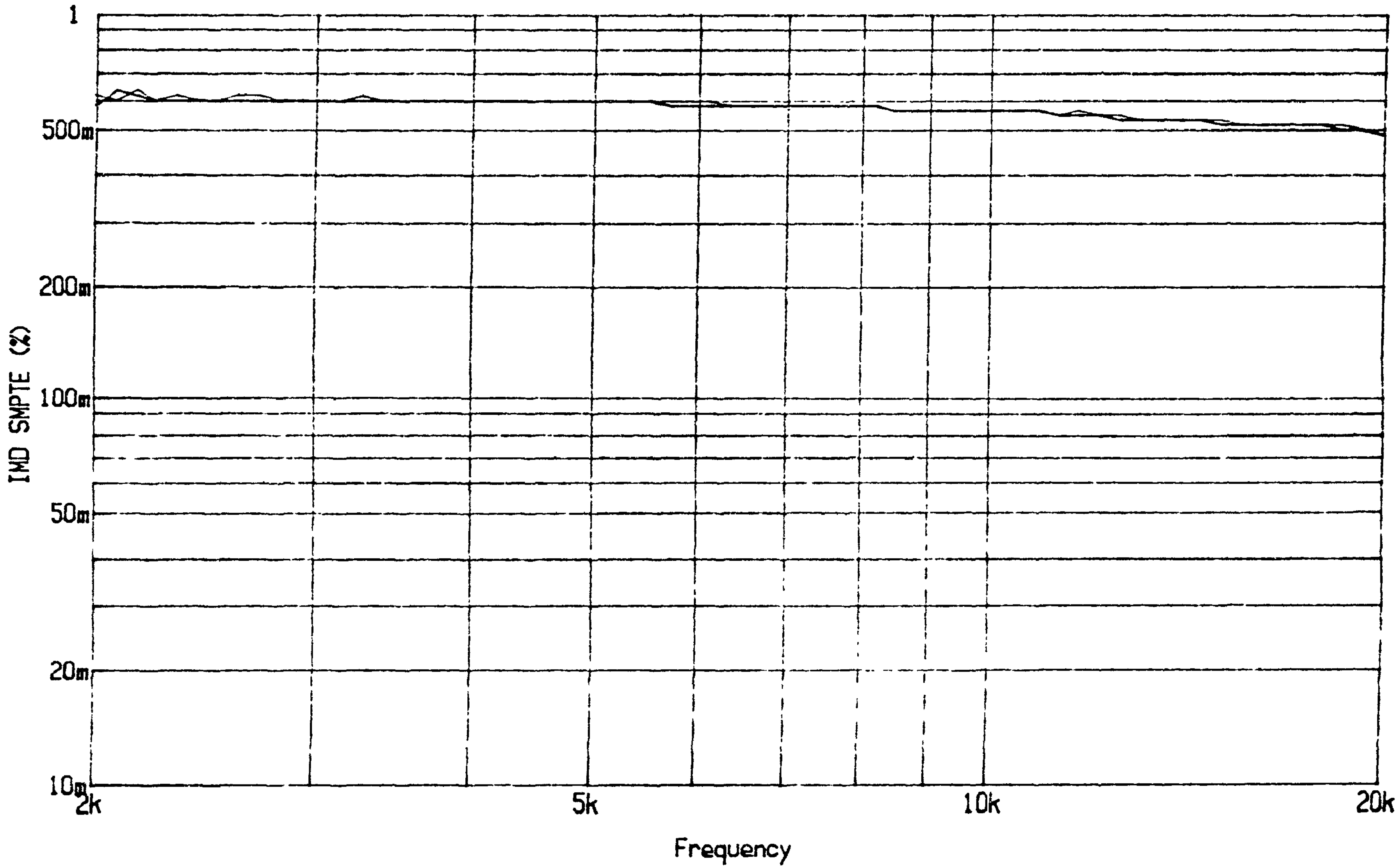
EVENTIDE ULTRA-HARMONIZER H3000
FREQ. RESPONSE



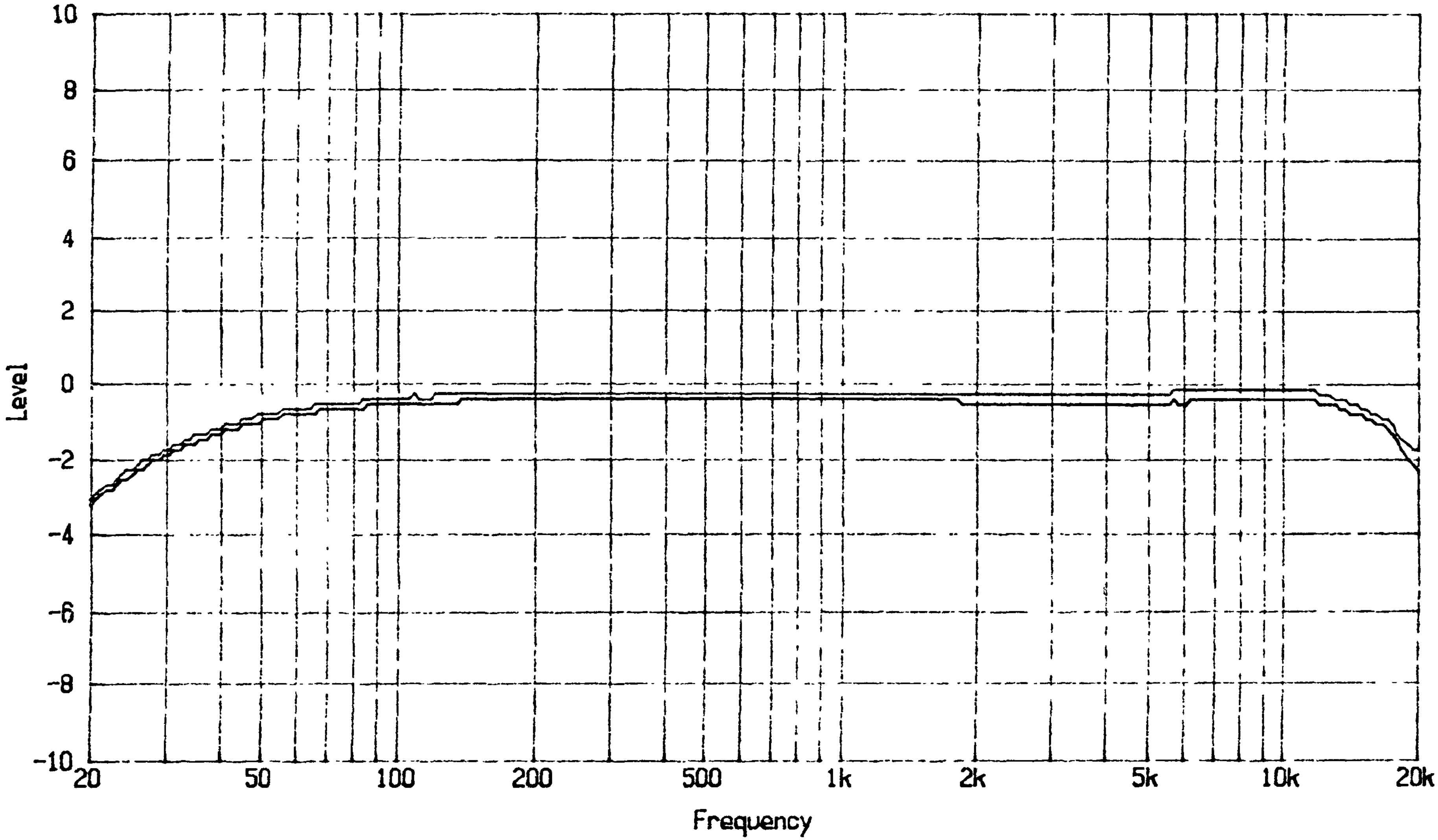
EVENTIDE ULTRA-HARMONIZER H3000
THD+N vs FREQ.



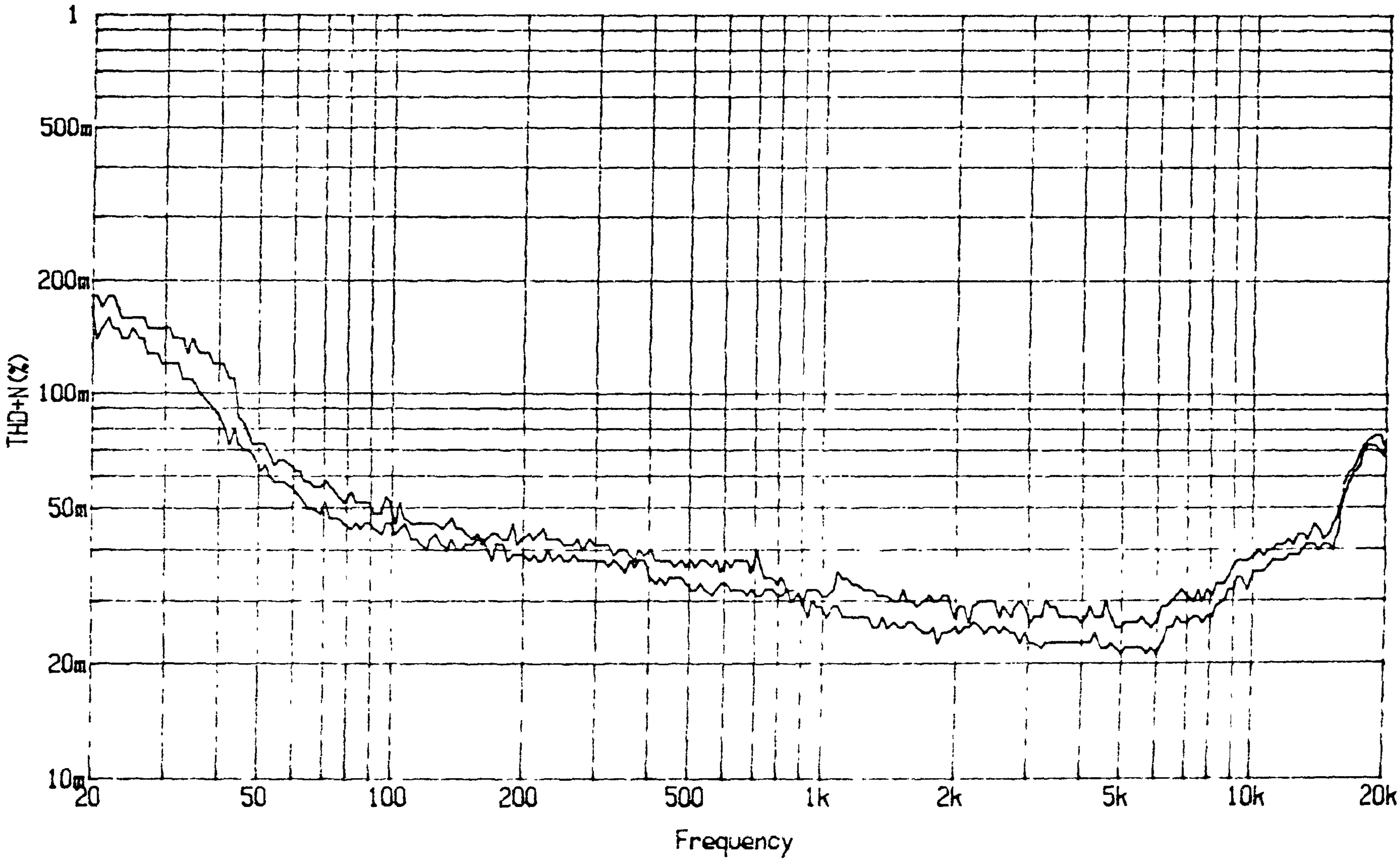
EVENTIDE ULTRA-HARMONIZER H3000
IMD vs FREQ.



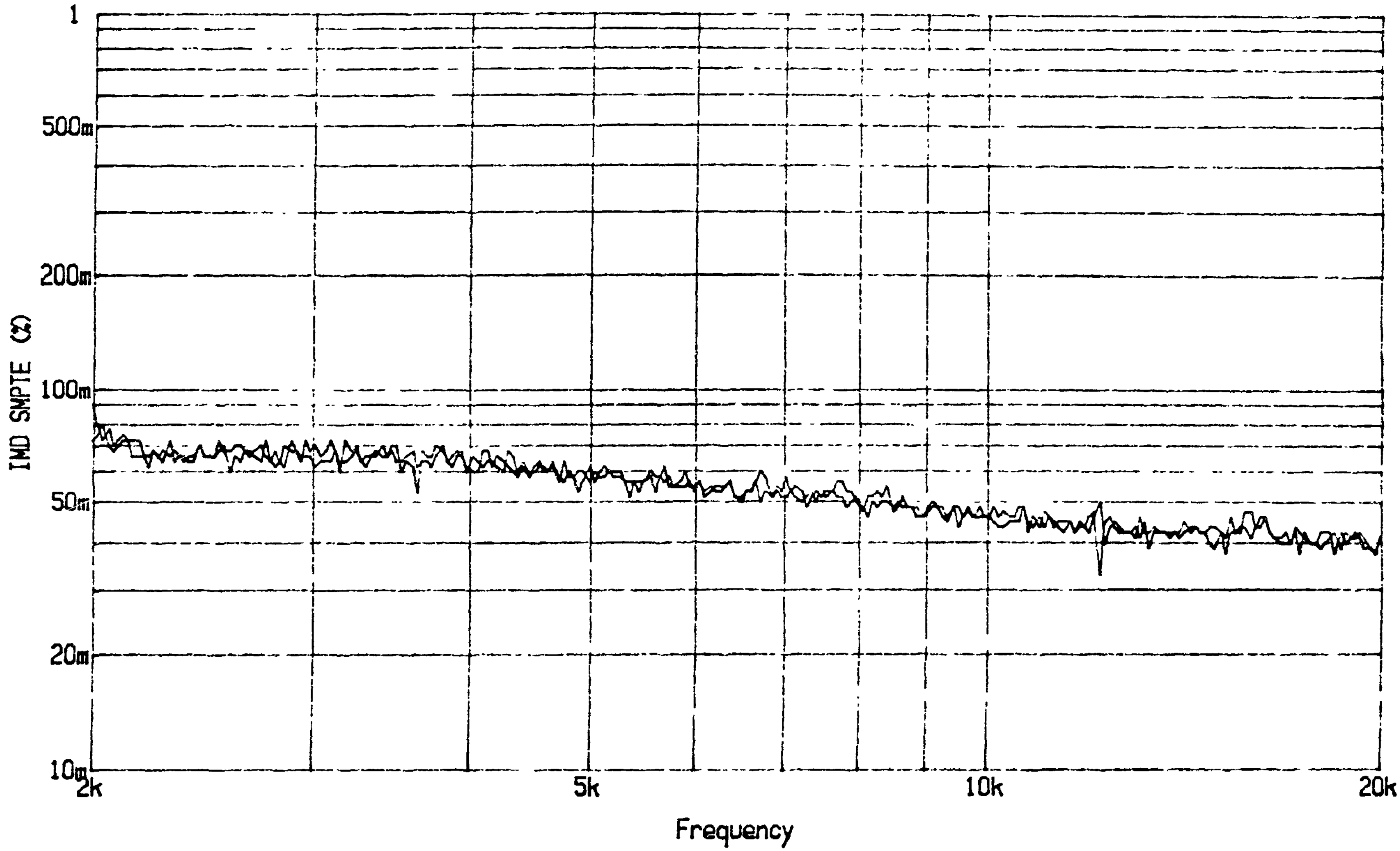
ROLAND E-660 DIGITAL PARAMETRIC EQUALIZER
FREQ. RESPONSE



ROLAND E-660 DIGITAL PARAMETRIC EQUALIZER
THD+N vs FREQ.



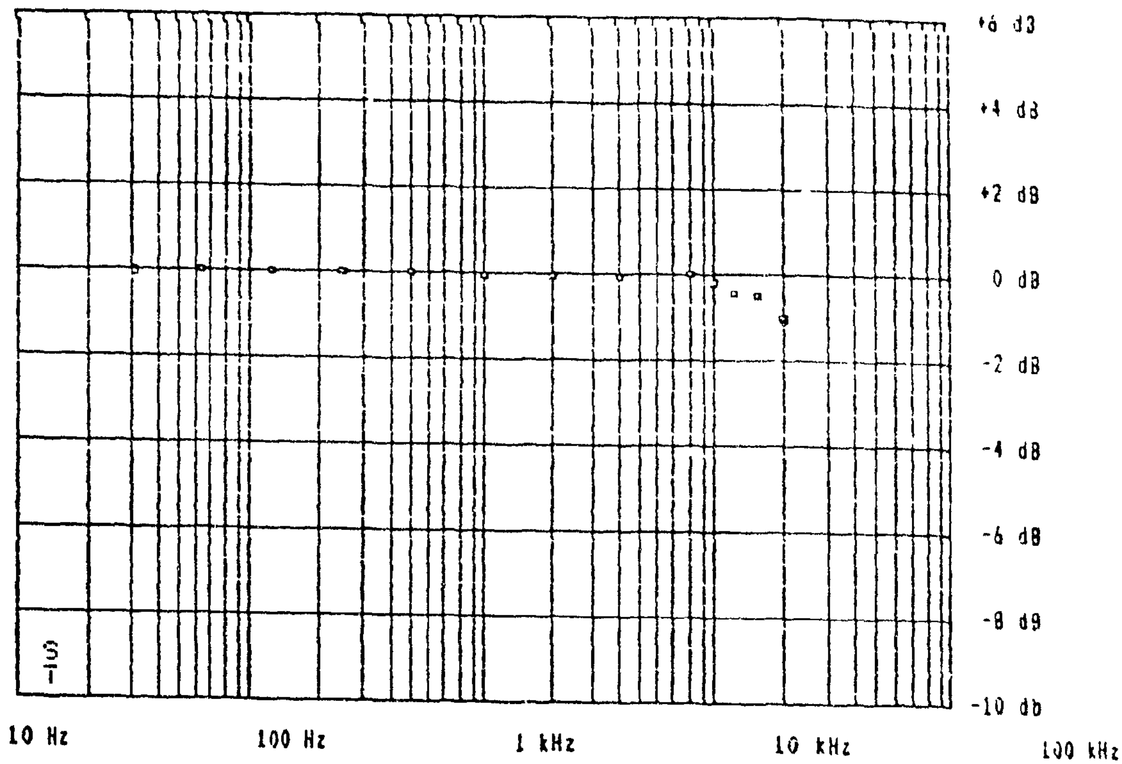
ROLAND E-660 DIGITAL PARAMETRIC EQUALIZER
IMD vs FREQ.



*** THE BANFF CENTRE *** ST SN# 103-AM2901 ***

TEST No. 1 *** RATIO *** LEXICON 480L DIGITAL EFFECTS PROCESSOR
MACHINE A OUT-LEFT

Time : 10:55 26 Nov 90
Input A RMS Detector
A Ref 10.8 dBU 1.001 kHz
High Pass OFF Low Pass OFF Not Weighted

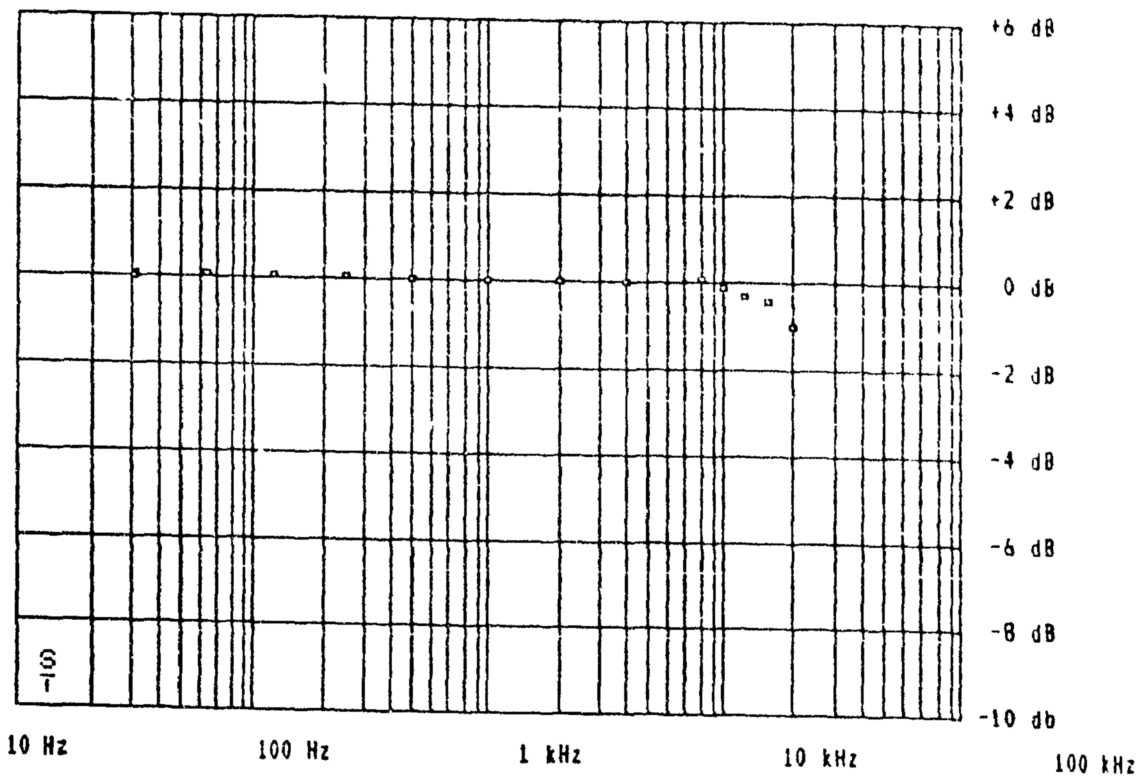


RATIO	FREQUENCY
- 1.09 dB	20.004 kHz
- 1.01 dB	20.004 kHz
- 0.51 dB	16.004 kHz
- 0.41 dB	12.504 kHz
- 0.21 dB	10.002 kHz
- 0.02 dB	8.002 kHz
- 0.09 dB	4.001 kHz
- 0.09 dB	2.001 kHz
- 0.09 dB	1.000 kHz
- 0.03 dB	500 Hz
- 0.01 dB	250 Hz
- 0.00 dB	125 Hz
- 0.01 dB	63.0 Hz
- 0.01 dB	31.5 Hz
- 0.07 dB	31.5 Hz

*** THE BANFF CENTRE *** ST SN# 103-AM2901 ***

TEST No. 2 *** RATIO *** LEXICON 480L DIGITAL EFFECTS PROCESSOR
MACHINE A OUT-RIGHT

Time : 10:56 26 Nov 90
Input A
A Ref 10.8 dBU
RMS Detector
1.001 kHz
High Pass OFF Low Pass OFF Not Weighted

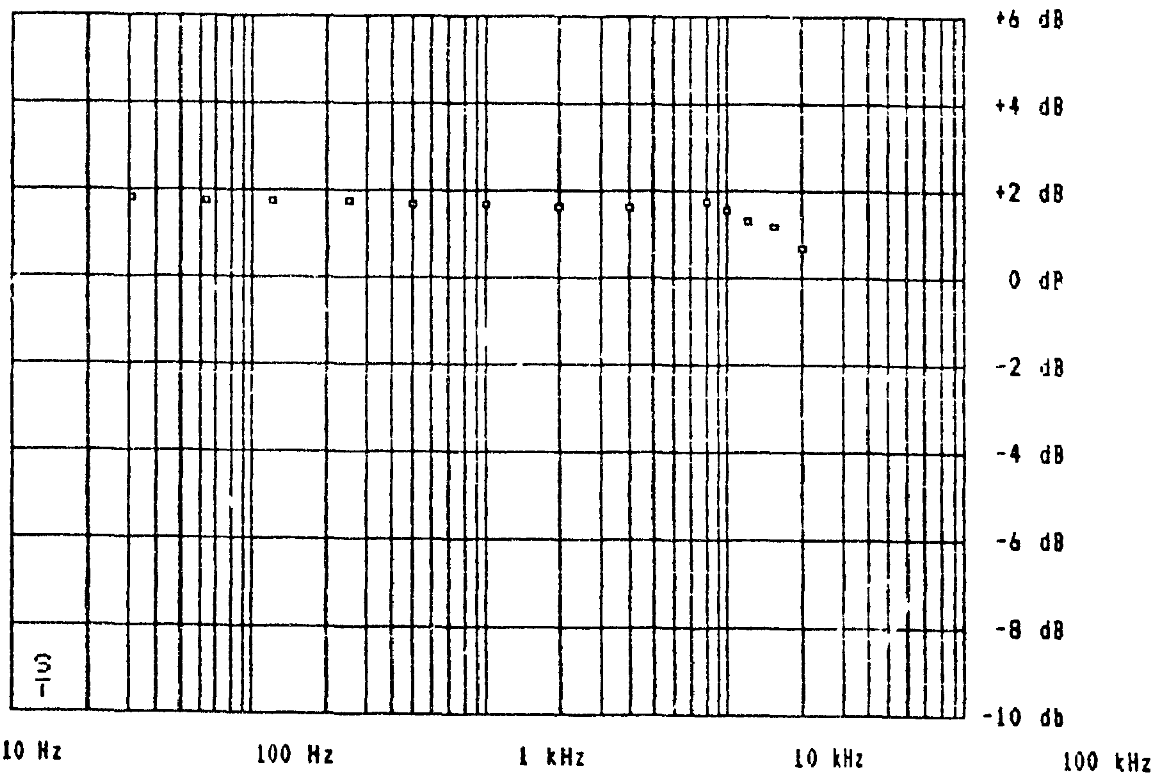


RATIO	FREQUENCY
- 0.99 dB	20.004 kHz
- 0.46 dB	16.004 kHz
- 0.34 dB	12.504 kHz
- 0.12 dB	10.002 kHz
0.06 dB	8.002 kHz
- 0.00 dB	4.001 kHz
- 0.01 dB	2.001 kHz
- 0.01 dB	1.000 kHz
0.03 dB	500 Hz
0.05 dB	250 Hz
0.05 dB	125 Hz
0.05 dB	63.1 Hz
0.04 dB	31.5 Hz
- 0.02 dB	31.5 Hz

*** THE BANFF CENTRE *** ST SN# 103-AM2901 ***

TEST No. 1 *** RATIO *** LEXICON 480L DIGITAL EFFECTS PROCESSOR
MACHINE B OUT-LEFT

Time : 11:25 26 Nov 90
Input A RMS Detector
A Ref 10.8 dBU 1.001 kHz
High Pass OFF Low Pass OFF Not Weighted

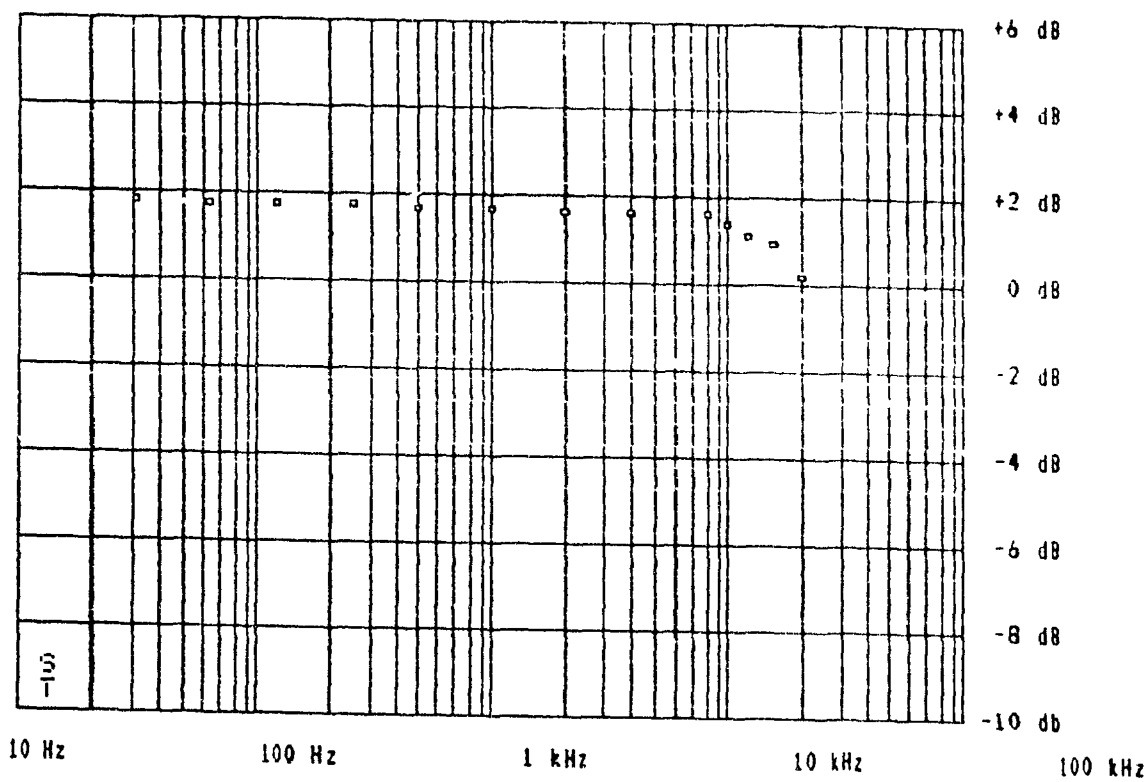


RATIO	FREQUENCY
0.68 dB	20.011 kHz
1.21 dB	16.008 kHz
1.34 dB	12.507 kHz
1.54 dB	10.006 kHz
1.72 dB	8.005 kHz
1.65 dB	4.003 kHz
1.65 dB	2.001 kHz
1.66 dB	1.001 kHz
1.71 dB	500 Hz
1.74 dB	250 Hz
1.75 dB	125 Hz
1.77 dB	63.0 Hz
1.79 dB	31.5 Hz

*** THE BANFF CENTRE *** ST SN# 103-AM2901 ***

TEST No. 2 *** RATIO *** LEXICON 480L DIGITAL EFFECTS PROCESSOR
MACHINE B OUT-RIGHT

Time : 11:26 26 Nov 90
Input A
RMS Detector
A Ref 10.8 dBU
1.001 kHz
High Pass OFF Low Pass OFF Not Weighted



RATIO	FREQUENCY
0.19 dB	20.009 kHz
0.91 dB	16.008 kHz
1.12 dB	12.507 kHz
1.39 dB	10.006 kHz
1.62 dB	8.005 kHz
1.62 dB	4.003 kHz
1.65 dB	2.001 kHz
1.66 dB	1.001 kHz
1.71 dB	500 Hz
1.73 dB	250 Hz
1.75 dB	125 Hz
1.75 dB	63.1 Hz
1.79 dB	31.5 Hz

Lexicon 480L Digital Effects System (Twin Delay Program):
THD measurement (&) versus Frequency

FREQ.	Mach.A-Left	Mach.A-Right	Mach.B-Left	Mach.B-Right
30Hz	.021%	.013	.016	.008
60Hz	.021	.013	.016	.008
120Hz	.021	.013	.016	.008
250Hz	.021	.013	.016	.008
500Hz	.021	.013	.016	.008
1K	.021	.013	.016	.008
2K	.020	.013	.015	.008
4K	.020	.013	.015	.008
8K	.018	.012	.014	.008
16K	.020	.013	.015	.009
20K	.70	.61	.61	.58

*** THE BANFF CENTRE *** ST SN# 103-AM2901 ***

TEST No. 1 *** IMD *** MACHINE A LEFT

Time : 11:29 26 Nov 90
Input A RMS Detector
IMD LEVEL
0.0597 % 10.3 dBU

TEST No. 2 *** IMD *** MACHINE A RIGHT

Time : 11:29 26 Nov 90
Input A RMS Detector
IMD LEVEL
0.0217 % 10.8 dBU

TEST No. 3 *** IMD *** MACHINE B LEFT

Time : 11:29 26 Nov 90
Input A RMS Detector
IMD LEVEL
0.0480 % 12.7 dBU

TEST No. 4 *** IMD *** MACHINE B RIGHT

Time : 11:30 26 Nov 90
Input A RMS Detector
IMD LEVEL
0.0152 % 12.6 dBU