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THE PERCEPTION OF SMALL CHANGES IN REVERBERATION TIME WITHIN RECORDING STUDIO CONTROL ROOMS

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ABSTRACT

The perceptibility of small changes in Reverberation Time (RT), when music is reproduced within recording studio control rooms, is judged by a number of subjects. The aim of this study is to determine the difference limen for Reverberation Times shorter than 0.6 sec which are usually encountered in control rooms. The experiment took place in a real control room and consists of two parts:

The first was the measurement of difference limen for Reverberation Time in full scale by changing the amount of absorption in the room. At this part of the experiment the measurement stimulus (music) was reproduced to the subjects by one loudspeaker located in the center in front of the mixing console.

The second part of the experiment was the measurement of difference limen for Reverberation Time with measurement stimuli pre-recorded in the control room using a dummy head and replayed to the subjects by headphones.

Differences between the two measurements were compared and estimated as well as the values of the difference limen itself.

1. INTRODUCTION

Although Reverberation Time is not the only important acoustical parameter which characterizes the soundfield in a critical listening space as a control room, it is the most important for the late portion of the sound decay in the room. As a result of this the assignment of just perceptible differences in RT is of crucial importance in designing the acoustics of a control room.

Seraphim [1], was the first who conducted detailed investigations in order to determine the difference limen for RT. The results which are based on an octave band of noise from 800 to 1600 Hz and a level difference of 30dB between the beginning and the end of the reverberant decay, revealed that:

- For the region of T between 0.6 and 4.0 sec the value of the relative difference limen ($\delta T/T$) is ($\delta T/T$) =4%. Changes in frequency and reduction of the level difference to 20 dB, had a minor effect in the results.
- For RT<0.6sec the difference limen increases up to a value of 12% for T=0.2sec. As stated by Seraphim, below T=0.6sec, the absolute rather than the relative change in T determines the difference limen. Its value is approximately T=0.024sec. A possible explanation, as it has been given by himself, is that in this case it is the "duration" of RT that is judged.

Plenge [2], investigated the "absolute perceptibility" of "humps" and "valleys" in the curve of reverberation vs frequency. The results from this study, in which only the statistical portions of the decay were compared, revealed that:

- In general, the "humps" are more easily recognised than the "valleys".
- The limen decreases for frequencies below 1000Hz. At lower frequencies the results approximately coincide with the values found by Seraphim for octave band noise between 800-1600Hz

The rest of this paper deals with two subjective experiments, which have been carried out in a real control room located in the Department of Acoustics and Audio Engineering in the University of Salford. For that purpose, a number of reverberant conditions were created in the control room by changing the amount of absorption in it. Each one of them constituted a different soundfield. The soundfields created, covered a range from the most dry to the most reverberant condition that could be achieved in the room under consideration. Both of these experiments are explained separately and the results are analysed.

2. EXPERIMENT 1: Full scale experiment

2.1 Method

During the preparation and the conduction of the experiment, all our attention was oriented, at first, to identify and at second, to control the possible sources of bias related with the physical environment, the listeners and the experimental procedure Toole [3].

2.1.1 Psychometric Method of Minimal Changes. The psychometric method that was used for getting the DL, was the method of Minimal Changes. [4]. The concept of this method is based in the perception of a Just Not Noticeable Difference (Jnnd) and a Just Noticeable Difference (jnd) by the human observer. The reproduction of the same music sample under the influence of the- each time different- absorption treatment of the room, constituted the various soundfields. The experimental process began by setting the two first soundfields-as far as RT is concerned- noticeably different. The soundfields were being presented in pairs to the subjects. Each pair, each time, consisted of the previous and the present soundfield. For this purpose it had been requested from the subjects to keep the impression of each presented soundfield as a reference for the comparison with the next one. After the end of two successive presentations the subject should have reported if he perceived a difference or not. The differences in RT between the successive presentations was following a decreasing order until the subject reports a *just not noticeable difference(jnnd)*. This was the point of subjective equality. Then the experimenter was continuing the procedure by introducing increasing differences until the subject reports a *just noticeable difference(jnd)*. This was the end of the series. Jnnd and Jnd give two different measurements for the DL. The true limen lies between them and it can be reached by averaging the Jnnd and Jnd.

2.1.2 Room, Equipment, and Music. One of the most important variables associated with the physical environment, is the room itself. Our control room was not treated with any special acoustical treatment that usually one encounters in modern control rooms (LEDE, RFZ etc.), since that was out of the project's objectives. Furthermore, some of these design philosophies are associated with targeted reverberation times in a restricted range. Our control room was a control room under construction with all the necessary audio equipment in it. It was a rectangular room with dimensions: L=4m, W=3.105m, H=3.192m.

Loudspeaker and listening positions were "fixed" during the experiment. The selection of the listening position was a compromise between the monitor builder's recommendations and the least detrimental position in the room, from the point of view of modal effects [5, pp. 1145]. One monitor was used. It was a mid-field loudspeaker from Alesis (model: Monitor two), and it was placed in the centre in front of the mixing desk at the height of a seated listener's ears, with its back 20cm from the front wall, as the constructor suggests. The mixing desk was located in the centre of the room, while the listening position was located in the 2/3 of the distance from the front wall of the room with coordinates: L=2.60m W=1.53m H=1.12m.

The music was reproduced to the subjects by means of the equipment already existing in the control room The experiment was conducted in monophonic reproduction avoiding the complications of the stereo presentations. Our intention was to allow the acoustical coupling of the loudspeaker to the listener through the control room but the decision was made to avoid the acoustical interference occurring at the listener's ears due to the excessive number of signal components and positional restrictions that are involved in stereo listening. [6].The program material for the test was selected for minimal interference from stereo-to-mono conversion. During the presentations, the absolute and subsequently the relative loudness of compared sounds was kept almost constant at 74 dBA. (There were slight fluctuations of no more than 2 dBA).

The test material that had been selected was the first 11 bars from a special anechoically recorded version of Handel/Harty: No. 6 Water music suite [DENON PG 6006], and it was the same for all the presentations. Its duration was quite long (21sec), in order to help the listeners to improve their confidence in judgement. It was a mid tempo sample and the most of the frequencies of the audible spectrum were included in its frequency spectrum

The subjective test was conducted for each one of the subjects separately. The average duration for each one of the experiments was 1h and 15min (minimum 1:10min and maximum 1:20min). A total of 14 experiments were conducted on five consecutive days.

2.1.3 Subjects. The subjects who participated in the listening test were lecturers, postgraduate students or staff members in the Department of Acoustics and Audio engineering at the University of Salford. The majority of the listeners were between the ages of 22 and 30 years old and all of them had previous listening experience. (All of them had participated in other psychoacoustic experiments in the laboratory). All the subjects had normal hearing at the standard audiometric frequencies.

Detailed written (and oral) instructions, about the experimental procedure and the objectives of the project, had been distributed to all the subjects in order to get them used to the experiment. The experiment was conducted for each one of the subjects separately.

2.2 Results

The initial number of listeners who participated in the experiment was 14 but due to the nature of the experiment some of the results were rejected. More details about the latter are in the discussion following the analysis. The statistical analysis was based on the results from six subjects and it was performed using two different methods: A t-test, and a Two-Way ANOVA. Both of them revealed the same results. The estimated value for the Difference Limen is, **Difference Limen=** $0.026 \pm 0.022\,\text{sec}$

3. EXPERIMENT 2: Binaural experiment

The first experiment was repeated using a dummy head in place of the subject. These binaural recordings were later replayed to the subject using the same psychometric method. Other features of the Method remained unchanged.

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3.1 Binaural recording method.

3.1.1 Binaural recording apparatus and set up. For the recording of the soundfields an artificial head from KEMAR (Knowles Electronics Manikin for Acoustic research) was used [7]. The artificial head was supplied with a pair of DB-100 Zwislocki occluded-ear simulators and DB-050 ear canal extensions, which simulate the ear canal and the acoustic impedance of the ear drum and the middle ear. Two external ears (DB-065/DB-066), typical of American and European male pinna sizes were used. A pair of ½ inch microphones Bruel and Kjaer type 4134 were placed in the ear drum positions in order to convert the sound to an electrical signal. Then each one of the microphones was connected to its associated preamplifier by means of a 90-degree adaptor (B&K UA0122). This adaptor was used due to space limitations inside the head. The microphones that were used are random incidence pressure mics with a frequency response between 3.9 and 20kHz and both of them were calibrated over the frequency range from 63-8kHz using a multi-frequency calibrator. Their deviation from the flatness was not more than 1 dB.

For the recording of the soundfields the dummy head was placed in the same listening position which was used for the full scale experiment, with its ears located at the height of a siting listener's ears. The output signals from the left and right channels of the KEMAR were digitally recorded, through two measuring preamplifiers B&K 2609, on to a Tascam DA-P1 DAT at a sampling rate of 44.1 kHz. The program had been already digitally transferred with the desired order in to another DAT (Tascam DA-30 MkII) and it was reproduced in to the control room by means of the existing equipment fig. 1.

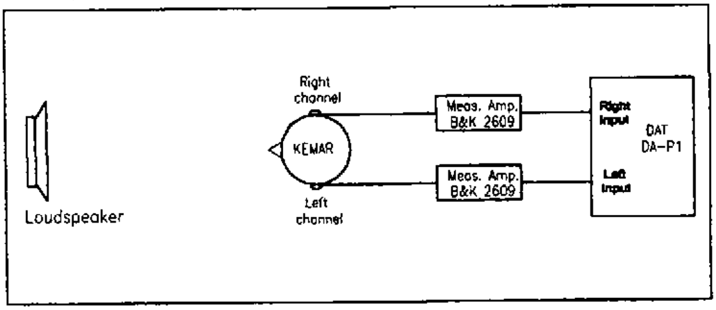

Figure 1: Set up for the binaural recordings of the soundfields in the control room

During the recording of the soundfields, great attention was given for the Sound Pressure Level to match those used for the live experiment (74 dBA). That was done for each one of the created soundfields, using digitally recorded white noise which was including with the program in the same DAT tape. The recording levels then were adjusted just below clipping.

One of the programs (program 1), was recorded for a second time with all the soundfields equalized in the listening position, in order for the experimenter to investigate the equalization effect in the perception of the reverberation. For that purpose a YAMAHA GQ 1031BII graphic equalizer was connected between the mixer and the power amplifier in the responsible for the reproduction of the program in to the room, chain.

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When the recording of the soundfields was completed, they were transferred in a computer for some final adjustments to be made. Cool Edit Pro (vers. 1), was used to edit digitally the recorded soundfields. Special care was given to ensure that the final files contained no audible artifacts or clues (like clicks, door slams, coughs) that might otherwise affect the listener's judgement. In addition, very careful level matching of all the recordings was considered essential since the louder recording will always sound more reverberant. Then, the soundfields were arranged in pairs and then they were transferred to a master DAT tape. That master tape then was transferred in a CD using an HHb CDR-800 cd recorder to form the final result.

3.2 Binaural reproduction

3.2.1 Optimization of the dummy head recordings for headphone reproduction. Our target was the dummy head's ears to become an extension of the listener's ears. Therefore, special efforts were made for the dummy head recordings to sound as natural as possible through the particular set of headphones that was used. The reason for that was due to experimenter's intention to reduce the chance of perceptual errors and fatigue.

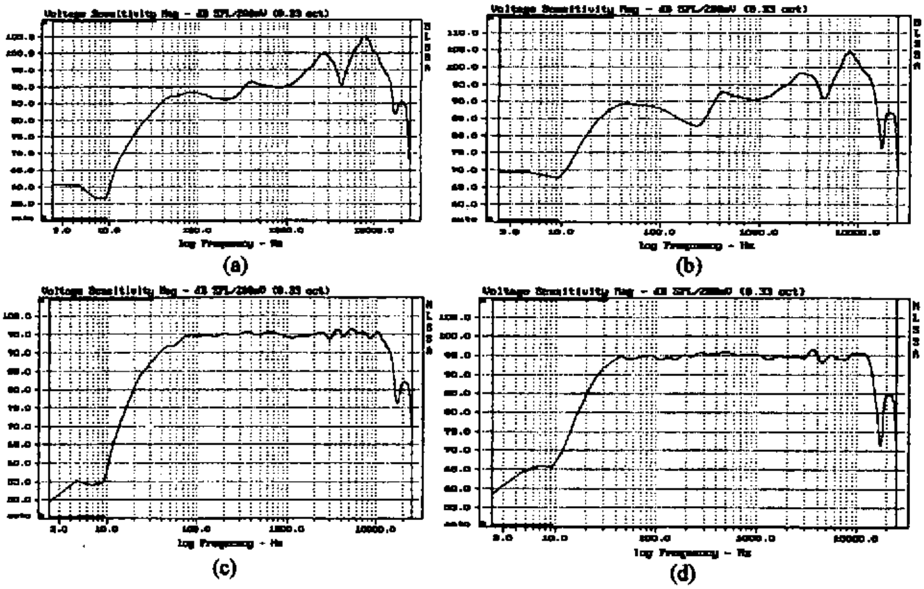

Figure 2:(a) and (b) are the frequency responses of the loop which includes headphone, pinna, coupler and the located in the ear drum position mic, for left and right channels respectively. (c) and (d) are their corresponding equalized frequency responses

Preliminary trials were made in order to test the quality of the recording when it is reproduced via headphones. In these trials only one music motif was used and the subject was the experimenter himself. The results were quite disappointing. There was a lack of bass and mid frequencies and it was sounded too distant from the source and not so well- balanced between the left and right ears.

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That was by no means an accurate representation of the live event. The frequency response of the closed-loop which includes the headphone, pinna, coupler, measuring amplifier and the located in the ear drum position microphone for each one of the left and right channels of the KEMAR, confirms that remark. [Fig.2., (a) and (b)]. Our attention was oriented to the equalization of this closed loop in order to achieve a flat frequency response. The considerable left-right asymmetry in the response of the dummy head was compensated by radiating MLS noise through the headphones and afterwards by adjusting the measuring amplifiers. The equalization process was made separately for the left and right channels closed loops of the KEMAR. The resulted frequency responses are shown in [Fig. 2., (c) and (d)].

After the above equalization the sound image was much improved. Although still slightly more reverberant than the live event, the, reproduced music sounded almost natural and all the frequencies were present.

Bearing in mind that the recording sounded more reverberant than the live event, we made the assumption that maybe the dummy head picks up more reverberant energy than the microphone which was used for the objective measurements in the dummy head's position. In order to investigate that, apart from the objective measurements, in which a B&K 4165 was used located in the dummy head's position, RT measurements were conducted in the room for each one of the soundfields using as a microphone the one located in the left ear of the dummy head. The results did not confirm that assumption. The RT measured, were slightly lower than their corresponding values measured with the other microphone. This is not surprising since the presence of the pinna restricts the exposure of the dummy head's microphone in to the soundfield. Placing the dummy head closer to the sound source was a remedy for that which was rejected [8]. Keeping the listening position the same for both the binaural and the full scale experiment was considered important for the comparison of the results from both the experiments. On the other hand this would be true for all the recorded sound fields and since the experimental procedure involves comparisons between soundfields, it was considered that it would not affect the results.

3.2.2 Reproduction of the binaurally recorded soundfields over headphones. A Micromega stage 1 CD player was used for the reproduction of the binaurally recorded soundfields.

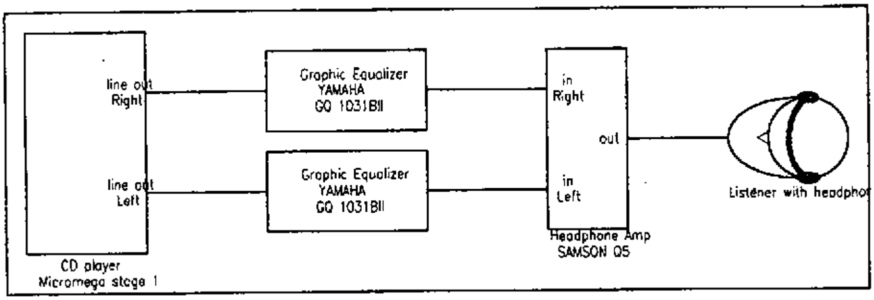

Figure 3: Set up for the binaural reproduction of the binaurally recorded soundfields

The output signals from the left and right channels of the CD were led, via two YAMAHA GQ 1031BII graphic equalizers, in to the left and right inputs of a SAMSON Q5 headphone amplifier, in which a pair of Beyerdynamic DT-770 headphones was connected. [fig. 3]. The closed-loop (Headphone - pinna - coupler) frequency response correction, for each one of the KEMAR'S channels, had been restored to the settings of its corresponding equalizer.

Attention was paid in matching the average playback levels of the headphones, measured at the eardrum, to those used for the live experiment (74dBA). This was done by putting the headphones on to the KEMAR system and adjusting the playback level of binaurally recorded white noise until it matches the levels measured live in the room using the same apparatus.

3.2.3 Testing procedure -Test material. The test material consisted of five different programs. All of them were passages from the following works with an average duration of eight sec:

Program 1: Handel/Harty: Water music suite	Mid tempo classic music
Program 2:Glinka: Overture - "Ruslan and Lyudmila"	Fast tempo classic music
Program 3: Debussy: Prelude a l' apres midi d' un Faune	Flute solo classic music
Program 4: Red Hot Chili Peppers, "Funky Monks"	Mid tempo slightly reverberant Pop music

Program 5, was the program 1, recorded with the listening position equalized.
The programs differ in style, tempo and frequency spectrum. The reason why five different music motifs were selected as test material, was in order for the experimenter to investigate the effect of music style in the perception of reverberation.

The psychometric method and the testing procedure were kept the same as those in the live experiment. The soundfields were presented in pairs to the listener for A-B comparison. The selection of the first presented soundfield of each pair between the two soundfields-members of the pair, was decided to be random. The time interval between them was not more than one sec.

Eleven listeners participated in the subjective test which took place in a listening room of the Dept. of Acoustics and Audio Engineering, in Salford University Detailed written and oral instructions were given to the listeners explaining the testing procedure, drawing their attention to do their judgements only with regard to the perception of reverberation. Preference evaluation of the presented sound fields for each one of the programs was also required from the listeners.

The listening test was conducted separately for each one of the subjects, and it was presupposing the presentation of the whole set of programs to each one of them. The average test duration was 35 min for the whole set of five programs.

3.3 Results

The collected results from the subjective test were statistically analysed using the Analysis Of Variance (ANOVA). Three factors were chosen to be tested: Program, Direction and Subject, of which the factor Subject was considered random, since it represents a sample from the whole population of listeners. For this purpose a Three-Way Analysis Of Variance, was employed to test the significance of the interactions between these factors, as well as the significance of each one of the factors itself. Unfortunately in the matrix which contained the whole set of data from the 11 subjects, there were missing cells since some of the results were rejected. These results correspond to subjects who reported that they perceived a difference at the point of objective equality, for a particular music program. Only a set of five subjects gave valid results for all the music programs. The easy solution was to use only the set of those five subjects for the statistical analysis and the rest to be rejected. We decided to use the whole set of data doing the following assumption: For a particular program (pop music), all the results from all the subjects were valid. It was decided a 3-way ANOVA to be performed for the complete set of those 5 subjects and a 2-way ANOVA for that particular program using all the subjects. The factors for the 2-way ANOVA were Subject and Direction, and the objective was to test the significance of these factors. If results from this analysis were not different, from the aforementioned 3-way ANOVA,(as long as the significance of these factors is concerned), then we could assume that this five-set of data is representative of the whole set. The results from the 2-way ANOVA, revealed no difference from those in the 3-way ANOVA and consequently the above table it is considered representative of the whole set of data. The estimated values for DL for each type of program are the following:

Type of Program	Program 1	Program 2	Program 3	Program 4	Program 5
DL (sec)	0.072±0.036	0.043±0.017	0.06±0.028	0.051±0.024	0.056±0.03

Table 1: Averaged values for jnnd, jnd and DL, from the whole set of data (11 listeners)

4. DISCUSSION

4.1 Full scale experiment

A number of factors related to the experimental procedure should have been predicted and controlled, since they might be possible sources of bias to the experimental result. The most important of them were the inevitable big time intervals between the successive presentations. The experimenter had to change the amount of absorption in the room before each one of the presentations. That was presupposing the lifting and the precise positioning of many absorbers (especially in the first big presented differences), located outside of the room. Thus, the time intervals were proportional to the changes in RT. (15 min in the beginning of the series, and about 4 min at the critical region where the small differences in RT were judged). Therefore, the listeners had to rely on their "acoustic memory".

Another important factor is that, because there were visual effects, between the presentations, the listener was waiting in an adjacent listening room (without listening in to music), and he was being led to the control room with his eyes covered by a mask to avoid visual cues. Thus, the test was a "blind" test from all the points of view.

A possible source of bias was also the fact that since the experiment was conducted separately for each one of the listeners, the soundfields created in the listening position should have been exactly the same for each one of them. But the creation of each soundfield requires precise placement of the absorbers in a minimum, as possible, time. Although special attention had been given to this factor, we can't guarantee that all the listeners were subjects to "exactly" the same soundfields. The average variation between corresponding soundfields during the conduction of the experiment, was estimated, in terms of standard error, as 0.0037 sec, with a maximum of 0.0067sec and a minimum of 0.0023 sec.

The statistical analysis revealed that the factor Subject is non significant. For the sake of the statistics, that factor was considered "random", since it represents a sample from the whole population of listeners.

The factor Direction is significant at the 10% level. That is a weak level of significance but it still means that the jnnd value is significantly different from the value of jnd. The latter implies, that the error of expectation is greater than the error of habituation; but that is a product of the experimental method. As a result of that perhaps it would be dangerous to average the jnnd and jnd values in order to obtain a single figure for the DL, but as it has been mentioned above it is considered sensible for the true limen to lie between them. Its value is the average of the jnnd and jnd values.

The value of DL, for reverberated noise as it has been estimated by Seraphim is approximately, 0.024sec. Our estimated value, for reverberated music is 0.026sec varying between 0.0039sec and 0.048sec. The confidence limits (standard error), which are a measure of spread, are in the order of the DL itself and reflect the influence of all those mentioned before factors (big time intervals etc.) to the resulted variance. But this is not surprising and not far from what one would expect from that sort of experiment. Our results are very close to Seraphim considering the different source signals. It is well known that music consists of transient sounds and different frequency bands. The fact that the listeners had to rely on their "acoustic memory" in combination with the transient nature of the music is a mirror of the experimental result. It is possible that transient nature of the music to be interpreted by someone as a subject under continuous change. Consequently, and especially when small changes are judged, he might report that he perceived a change even if that change does not exists objectively. It is noteworthy that, in some cases, this was true even when big differences were judged. That was the reason why some of the subjects were rejected. In addition, doing comparisons between successive presentations, the subject's judgement should have been due to an overall impression; but that does not eliminate the possibility that their judgements may be influenced by a most prominent reverberant decay of a particular frequency band. In our experiment the presented reverberation curves were not flat over the frequency spectrum. Most of them had "humps" and "valleys". Consequently the reverberation decay was different for each individual frequency band. The latter, in combination with the human's limited acoustic memory and the evidence that the "humps" are more easily recognised than the "valleys" as Plenge pointed out, might lead the listener to report a change even though in the "absence" from his memory of the first presented member of the pair under comparison.

4.2 Binaural experiment

The Binaural experiment provided much better experimental control, allowing rapid subjective comparisons to be made as opposed to the full scale experiment. The time which was required for the completion of the binaural test for a single listener was much less than that to the real time test. That time was estimated to be 91 % less compared to the real time one.

Factor Direction was found to be significant at the 2.5% level, but this is a product of the experimental method.

Testing the significance of the (Subject x Direction) and (Subject x Program) interactions, some discrepancies were observed. Thus, as far as the interaction between Subjects and Direction is concerned, the calculated F ratio revealed that they are significantly different at the 2.5% level of significance. That is in contradiction to the result which is concluded from the graphic representation between these two factors.

A general procedure for drawing such a graph is to represent the levels of factor Subject as values on the x axis, making a separate curve for each level of factor Direction (Fig. 4). The graph revealed that even

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though the lines do not cross, there is some interaction due to one subject: PP. In a similar way, while the calculated F ratio revealed that there is no interaction between the factors Subject and Program, the graph shows the these two factors interact (Fig. 4). An explanation about these discrepancies has been given by Ferguson [9, pp.321]. He pointed out, that in a triple classification case for a "mixed" model with one observation per cell, no unbiased test of differences between Subjects, or interactions in which the factor Subject (random factor) is involved, can be made. Probably, by repeating the experiment for several times, these discrepancies would be eliminated and more accurate statistical results would be obtained. But apart from the point of view of statistics, the scientific rather than the statistical importance of the interaction is usually at issue.

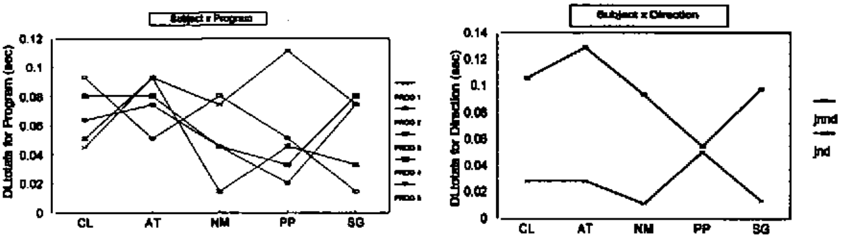

Figure 4:Graphic representation of the interactions Subject x Direction and Subject x Program

The meaning of an interaction between the factors Subject and Direction is, that the perception of a jnnd or jnd is different for different subjects. The error of expectation is also greater than the error of habituation, but as it has been stated above this is a product of the experimental method. Hence, even in the presence of a possible interaction between these two factors, the true limen lies between them. As one can see in Fig.4, even the results from PP verify the above argument.

There is a possible interaction between the factors Subject and Program which manifests that the effect of different programs, in the perception of small changes in RT, is different for different subjects. This is not surprising and it was reported by some of the subjects as well. They found it easier to hear differences in RT during the presentations of programs which were faster or more " impulsive" than the others where the masking effect of the longer decay was more obvious. These were the programs (1) and (4). Program (3), although it was a flute's solo, did not attract the listener's "sensitivity", probably because of its "legato" style. Finally, some listeners were more sensitive in the perception of a DL, to the equalized version of program 1, than the unequalized one. Nevertheless, there is no obvious direction of the listener's "sensitivity", to a particular program.

Although statistical analysis revealed that the factors Subject and Direction are not significant, unbiased conclusions are difficult to make about these factors, since both of them are involved in a possible interaction.

As we have seen so far, the perception of small changes in RT seems to be dependent on subject and music program. Furthermore, since the factor subject was considered random, the assignment of separate limens for each one of the subjects would be extreme. Consequently, the values of DL for each one of the programs, could be obtained by averaging across Direction and Subject (Table 1). As one can see from Table 1., the estimated DL values for each different program are close enough, so one could conclude that for practical purposes, that the effect of music program has little influence in the perception of small changes in RT. Hence, by averaging across all the programs, one could obtain a single figure for the DL. That is , **DL=0.057±0.005sec**

5. CONCLUSIONS

The limited "acoustic memory" and the transient nature of the ,source signal (Music), were the most decisive factors in the perception of small changes in Reverberation Time within a control room. Both of them are unfortunately bywords for "uncertainty" in judgement, for that sort of experiment. That is something that was expressed by the subjects as well. In a full scale experiment, uncertainty, is a factor which might lead not necessarily to higher Difference Limens, but to smaller DL as well. On the other hand, both of these factors are very apparent in the real world for every sound engineer and their influence is reflected through the variance in the experimental result.

The estimated DL from the full scale experiment, bearing in mind the transient nature of the music, lies between the expected limits while, in our opinion, the most reliable value for that sort of experiment, is towards to the 0.048 sec direction. The latter was also confirmed from the binaural experiment's results. The binaural experiment was a better controlled experiment, relieving the listener of the dependence on his acoustic memory. Nevertheless, the non-steady nature of the music seems to be crucial in the perception of small changes in Reverberation Time for both the experiments. The estimated Difference Limen was higher than that from Seraphim's experiment, but within the expected limits, taking in to account the different source signal. To summarize, the transient nature of the music reveals a degrading capability of the listener in the perception of small changes in Reverberation Time. Therefore, in order to portray the influence of all the aforementioned factors from both of the experiments to the final result, one could obtain a single figure for the DL, by averaging across the estimated DL values from these two experiments. Thus, the overall Difference Limen is, **0.042±0.015sec.**

REFERENCES

[1], H. P Seraphim, Acustica, 8 p280 (1958)

[2] G Plenge, Acoustica 1,6 p.269 (1965/66)

[3]F.E. Toole "Subjective measurements of loudspeaker sound quality and listener performance", JAES vol. 33,No. 1 / 2, (1985)

[4] Guilford, "Psychometric methods" McGraw-Hill 1954

[5] G. Ballou, "Handbook for sound engineers. The new audio cyclopedia" second edition, pp1144-1145 Howard W. Sams & Company (1991)

[6] F.E.Toole, "listening tests-Turning opinion in to fact," J. Audio Engineering Society.,vol. 30, pp. 431-445 (1982 June)

[7] M. D. Burkhard and R. M. Sachs, "Anthropometric Manikin for Acoustic Research," J. Acoust. Soc. Am., vol 58, pp. 214-222 (July 1975)

[8] D. Griesinger, "Equalization and Spatial Equalization of Dummy- Head Recordings for Loudspeaker Reproduction," J. Audio Eng. Soc., vol. 37, No. 1/ 2, (1989 Feb.)

[9] G.A. Ferguson "Statistical analysis in psychology and education" fifth edition" pp 261-262 McGraw-Hill (1981)