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Comparative study of measured acoustic parameters in concert halls in the city of São Paulo

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Abstract: During the last two years we have been working in a project at the University of São Paulo, Brazil, devoted to the investigation of acoustic behavior of rooms designed for music performance and music production (AcMus Project). This paper describes some results related to the determination of subjective acoustic parameters for some concert halls in São Paulo. First we describe the implementation of a program for acquiring the impulse response. The system is based on MLS and logarithmic sweep methods and offers tools for estimating different acoustic parameters. Then, we introduce some questions related to subjective acoustic parameters such as liveness, warmth, brilliance, loudness, clarity and definition. These parameters can be related to objective parameters such as reverberation time, energy variation, and ITGD. The last section of this paper deals with comparative results of measurements made in some concert halls in São Paulo. We will show our conclusions about their different acoustic behaviors, and we also discuss the alternative use of a directional source to measure those parameters.

1. INTRODUCTION

In 2003, we initiate at the University of São Paulo, Brazil, a research project on room acoustics with emphasis on musical questions. The core of the work, entitled AcMus [1], is concentrated on the development of computational tools for project, measurement and simulation of the acoustic behavior of rooms designed for music performance and production. This paper discusses the development and application of the technology used in the acoustic measurements using a software developed by our group. The system can be used to calculate the values for different parameters that would serve as criteria for the determination of the acoustical quality of rooms. Although many acoustic parameters can be determined from measurements and specific calculations, we are interested in parameters that present correlation [2] with individuals' sensitivity and perception in a specific environment, particularly in rooms destined to music performance and listening. The basic experimental procedure consists of emitting specific signals inside of a room and recording the resultant signals at different points.

To obtain the impulse response of the room the recorded signal is mathematically compared with the generated signal. Then, we extract the necessary acoustic parameters from a series of computational processing. We have made measurements in different concert halls in São Paulo that carry regular performances of chamber and symphonic ensembles and compared the results. For the same positions where the measurements have been done, we have also recorded anechoic samples of music for future subjective evaluations. Since our analysis aims at investigating rooms projected for musical performance, we have decided to take into consideration the fact of that musical instruments project the sound in a directional way. For this reason the measurements were carried out with the use of a directional source to stimulate the room. Although the directivity of the source is not properly the same one could observe in a real instrument or in a instrumental ensemble, we had the chance investigate how much this option have intervened with the results in comparison with the use of a omnidirecional source as indicated in the ISO 3382 [3]. For this purpose, we have compared the results we have obtained with the results of other measurements achieved by using ominidirectional sources as extensively published in a recent work [4].

2. CHARACTERIZATION OF ACOUSTIC PARAMETERS

We call acoustic subjective parameters the values established according to the psychoacoustic sensitivity of listeners. Such parameters play a distinctive role in the artistic quality of a musical event. Each of them is related the specific physical phenomena that provoke the acoustic impressions that define the characteristics of a music room. The most important parameters are represented by mathematical expressions that generate objective values. They represent measurable physical values, as shown in Table 1, and can be defined as follow:

- *RT 60* is the *Reverberation Time*. RT_{125} , RT_{250} , RT_{500} are the reverberation time for the frequency bands of 125Hz, 250Hz 500HZ and so on. Another parameter related to the reverberation is the *Early Decay Time* (*EDT*).
- *BR* is the bass ratio and *TR* the treble ratio.
- t_d is the instant that direct sound arrives at the recording device and t_r is the instant at which the first reflection arrives. The difference between these two moments is the *Initial Time Delay Gap* (*ITDG*).
- *Ed* and *Er* are the total energy of direct and reverberant sound, respectively.
- p*(t)* is the instantaneous value of the acoustic pressure. The physical expression *Clarity* (C_{80}) is the logarithmic ratio between the energy that arrives in first the 80 milliseconds and the remaining energy. This ratio can also be taken for first the 50 milliseconds (C_{50}). The expression for *Definition* (D_{50}) is the linear ratio between the energy that arrives in first the 50 milliseconds and the total energy. This ratio can also be taken for the first the 80 milliseconds (D_{80}).
- *Interaural Cross Correlation* (*IACC*) is the maximum value of the coefficient of correlation between the signals at left and right ears. *Lateral Fraction* (*LF*) is the ratio between the energy captured in a lateral plane and the omnidirecional energy at the same point.

Table 1: Acoustic Parameters

Subjective Parameter	Objective Index	Physical Expression
Liveness	RT 60	Schroeder (1965)
Warmth	BR	$\dfrac{RT_{125}+RT_{250}}{RT_{500}+RT_{1000}}$
Brilliance	TR	$\dfrac{RT_{2000}+RT_{4000}}{RT_{500}+RT_{1000}}$
Ratio D/R	RDR	10 log [Ed / Er]
Intimacy	ITDG	$t_d - t_r$
Clarity	C80	$10\log\left[\dfrac{\int_0^{80ms} p^2(t)dt}{\int_{80ms}^{t_f} p^2(t)dt}\right]$
Definition	D50	$\dfrac{\int_0^{50_{ms}} p^2(t)dt}{\int_0^{t_f} p^2(t)dt}$
Spatial Impression	IACC	$\max\left\|\left[\int p_e(t)p_d(t+\alpha)dt\right]\left[\int p_e^2(t)dt\int p_d^2(t)dt\right]^{-1/2}\right\|$
	LF	$\dfrac{\int_{5ms}^{80ms} p_{lat}^2(t)dt}{\int_0^{80ms} p_{omni}^2(t)dt}$

3. THEORY AND PROCEDURES FOR OBTAINING ACOUSTICAL PARAMETERS

In terms of acoustics, a room designed for musical purposes can be seen as a linear system. One of the main characteristics of a linear system is its impulse response (IR), that is, the reaction of the system to an impulsive signal. The IR can be understood as the "acoustical signature " of a room, and all the acoustic parameters can be extracted from it. The Fourier Transform of the IR h (t) is the frequency response H (w), also called *Transfer Function*. From the signal theory we know that the output y (t) of the linear time invariant system is the convolution between the input x (t) with the impulse response h (t). In the frequency domain it is equivalent to the multiplication of both X (w) and H (w) transfer functions:

$$Y(w) = X(x) \,.\, H(w) \quad (1)$$

3.1 Obtaining the IR

The literature indicates two ways for obtaining the IR. The first one is known as MLS

(*Maximum Length Sequences*) a periodic sequence of unitary pulses (positive and negative) systematically constructed to satisfy some mathematical properties [5]. Although its deterministic characteristics, the MLS signal sounds as a white noise, being classified a pseudo-random signal.

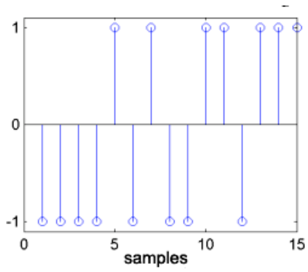

Figure 1: representation of a MLS sequence

According to the signal theory, we can say that the cross-correlation between the input x (t) and the output y (t) is related to the auto-correlation of the input by a convolution with the impulse response:

$$R_{xy}(t) = \int_0^{\infty} h(\tau) R_{xx}(\tau + t) d\tau = h(t) * R_{xx}(t) \qquad (2)$$

The MLS presents two important properties: its Fourier Transform has the same magnitude for all frequency components, so its auto-correlation function is a Dirac delta function.

$$R_{xx}(t) = \delta(t) \qquad (3)$$

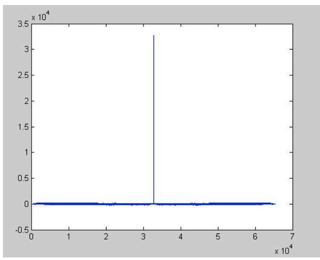

Figure 2: auto-correlation of a MLS sequence

The graphic at Fig. 3 was obtained in program implemented in MatLab by our colleagues in the AcMus Project. The result of convolving h (t) with a Dirac delta function is the impulse response h (t) itself. Thus, the impulse response can be found by cross-correlating the MLS input x (t) with the registered output y (t):

$$R_{xy}(t) = \int_0^{\infty} h(\tau) \delta(\tau + t) d\tau = h(t) \qquad (4)$$

This calculation is made by using the Fast Hadamard Transform, that optimizes the processing of the matrices generated by cross-correlation [6].

Another way of obtaining the IR is by using sine sweeps [7]. The sine sweeps are sinusoids that have their instantaneous frequency varying in the time. Sweeps can be linear or logarithmic. In our case, we use a logarithmic sweep that exhibits a pink spectrum, that is, its amplitude decays at a rate of 3dB/octave. It means that the signal has the same energy per octave. The frequency values double in time in a fixed rate.

$$\frac{\log\left(\frac{f_2}{f_1}\right)}{T_2 - T_1} = cte \qquad (5)$$

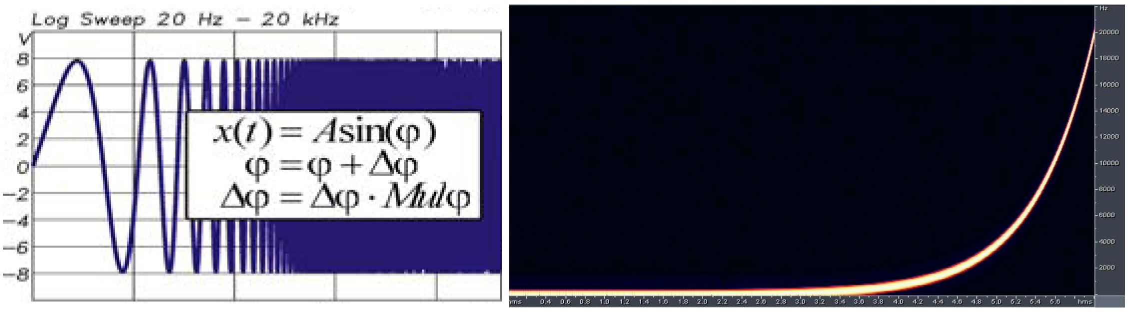

Figure 3: Representation of a log sweep in time (left) and a real spectrogram of a log sweep (right)

Be N the number of samples, the multiplying factor is given as follow:

$$Mul\varphi = 2^{\frac{\log_2\left(f_{final} - f_{inicial}\right)}{N}} \qquad (6)$$

To obtain the IR one should apply the IR as follow:

$$\text{Y (w)} = \text{X (w). [G (w). L (w)]} \qquad (7)$$

In the equation above, Y (w) is the transfer function of the signal recorded in the room, X (w) is the transfer function of the emitted signal (log sweep), G (w) and L (w) are the transfer functions of the room and the loudspeaker respectively. The sweep technique shows the advantage of eliminating the influence of the loudspeaker (L) in the response of the room (G), which is attained by means of a simple operation of division in the frequency domain (deconvolution):

$$G(w) = \frac{Y(w)}{X(w).L(w)} \qquad (8)$$

The deconvolving function was implemented in MatLab as follow:

```
function ir = dechirp(rec,B,A,n)
load ir_loudspeaker
ir = real(ifft(fft(rec(:,1),n)./fft(rec(:,2),n)./fft(ir_loudspeaker)));
ir = filter(B,A,ir);
```

where ‘`rec 1`’ and ‘`rec 2`’ are the recorded and emitted signals respectively. The ‘`ir_loudspeaker`’ is the IR for the loudspeaker obtained from the processing of measurements done in free field. The ‘`filter`’ function [6] generates octave band filters according to the IEC 1260.

3.2 Processing the IR

The energy decay curve is obtained from the room’s IR by using Schroeder Integration method [3]:

$$E(t) = \int_{t}^{\infty} p^2(\tau)d\tau = \int_{0}^{\infty} p^2(\tau)d\tau - \int_{0}^{t} p^2(\tau)d\tau \tag{9}$$

where ‘*p’* is the IR.

By processing this integral, one can calculate all the parameters shown on Table 1. In the graphics below we show some examples of this processing in MatLab, for some measurements made in music halls in São Paulo. The yellow point in the graphic shows the truncation point obtained by Lundeby’s method [6].

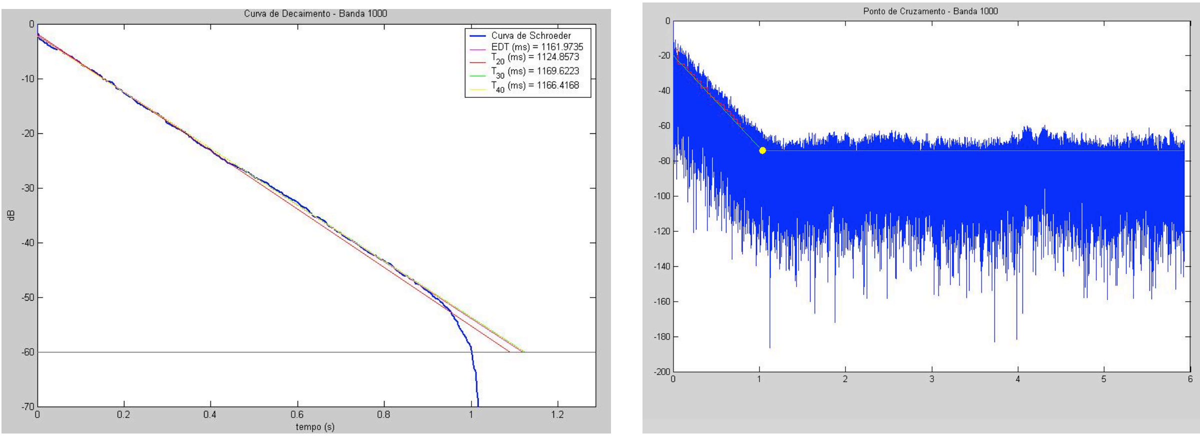

Figure 4: Schroeder Integration (left) and respective dB decay curve (right)

4. MEASUREMENTS

4.1 Measurement setup

Fig 6 shows the signal’s flowchart during the measurements.

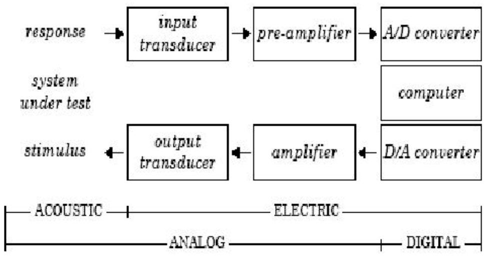

Figure 5: measurement scheme

For the measurements we have used a PC laptop computer, an *Edirol FA-101* digital audio interface, *Behringer ECM 8000* and *Neumann U-87* microphones, and *Electro-Voice EV Sx200a* active loudspeakers. The audio interface is connected to the computer by a firewire connection. The signal is sent through one of the interface’s output channels to the loudspeaker. Another channel sends the signal back to one of the interface’s input. This signal is taken as a reference to sync the signal that is recorded by the microphones. An example of acquired and emitted signals is shown bellow:

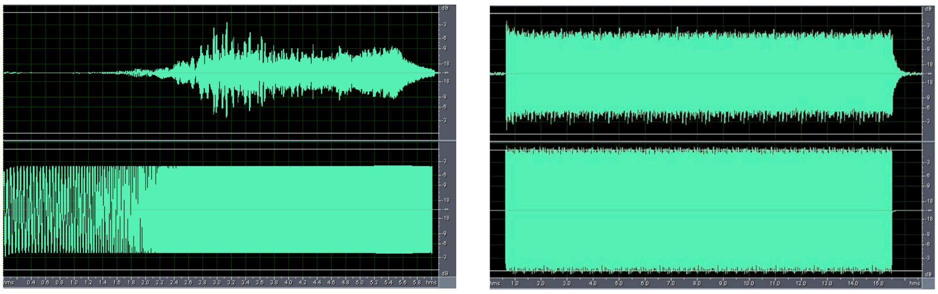

Figure 6: log sweep (left) and MLS (right), original signal (bottom) and registered room signal (top)

After recording the signals we apply a cross-correlation (MLS) or deconvolution (log sweep) between the registered and original signals to obtain the IR. Then we process the Schroeder's integral to extract the acoustic parameters using functions created in MatLab [6].

4.2 Results and Conclusions

We have carried out measurements in some of halls of the city of São Paulo. Here we are going to analyze three of them: the Teatro Municipal de São Paulo, which hosts the Orquestra Sinfônica Municipal, the Teatro São Pedro, former headquarters of the Orquestra Sinfônica Estadual, and the Anfiteatro Camargo Guarnieri, which hosts the Orquestra Sinfônica da USP. All the measurements were made without audience. Some of the preliminary results are shown below:

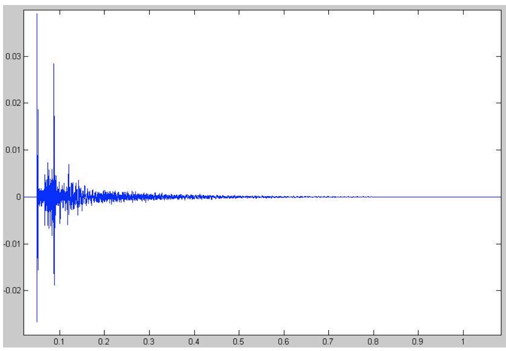

Figure 7: lateral main floor IR Municipal

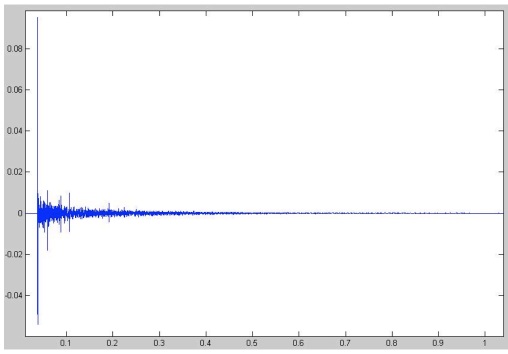

Figure 8: central main floor IR Municipal

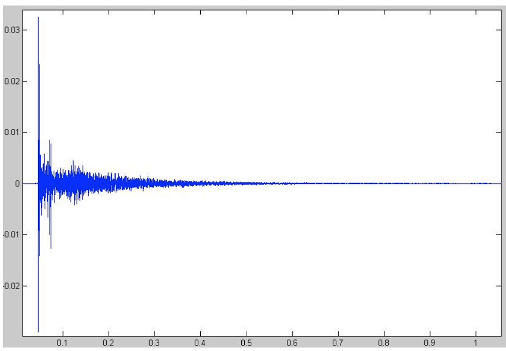

Figure 9: balcony IR São Pedro

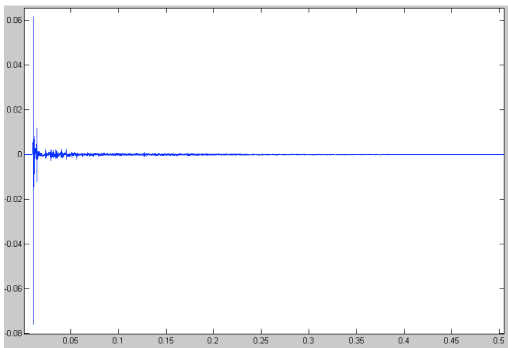

Figure 10: stage IR Guarnieri

We have made many measurements for each position, and calculated both the parameters for the average IR and the average of these parameters for every IR (Fig. 18). The results for all the parameters are close enough with a maximum standard deviation of 4.2% above 125 Hz.

The log sweep and MLS techniques have presented very similar IR and practically the same values for the calculated parameters (Fig. 12).

In comparison with the results gotten from similar rooms, one can observe that the use of the directional source produces differences for some parameters, specially for the *Clarity*, a problem already detected in the measurements of lecture halls [8]. For the low (diffuse) frequencies, the loudspeakers behave almost as an omnidirectional source, but as the frequency increases they become more directional. This can explain the relative high values of *Clarity* in the high frequencies (specular), since in this frequency band the room as a whole is less excited than when it is exposed to the diffusion of an omnidirecional source. Thus, one can expect that in this case the microphone captures less reverberant energy in the high frequencies. The values of RT 60 had presented less sensitivity to the directivity of the source.

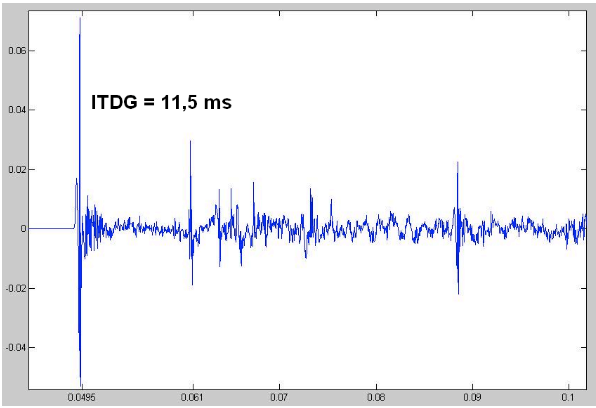

Figure 11: IR Guarnieri detail (first 100 ms)

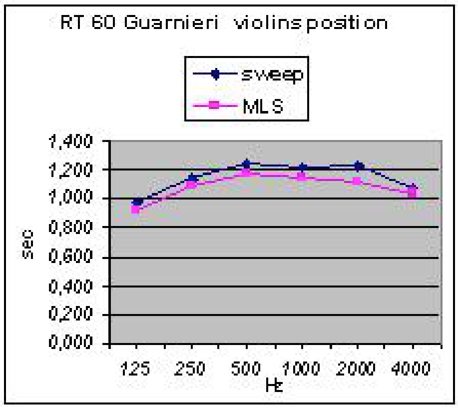

Figure 12: comparing MLS with sweep

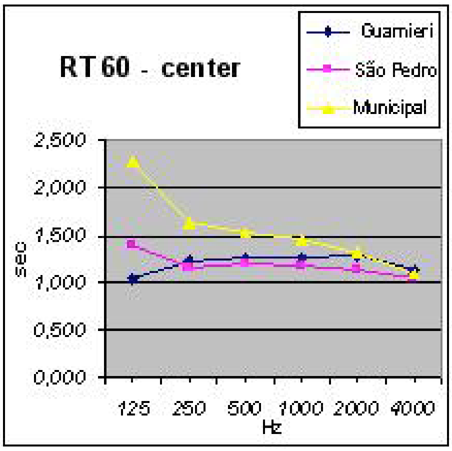

Figure 13: main floor RT60

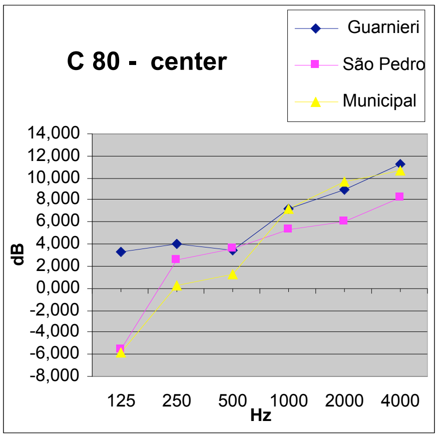

Figure 14: main floor C 80

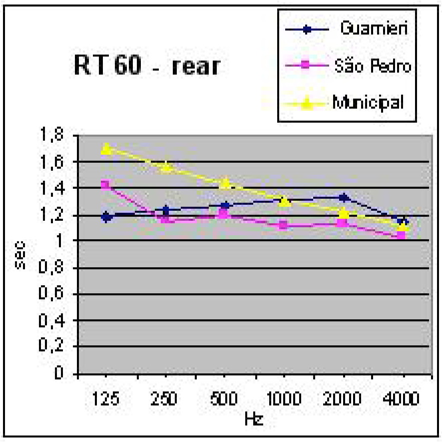

Figure 15: rear audience RT 60

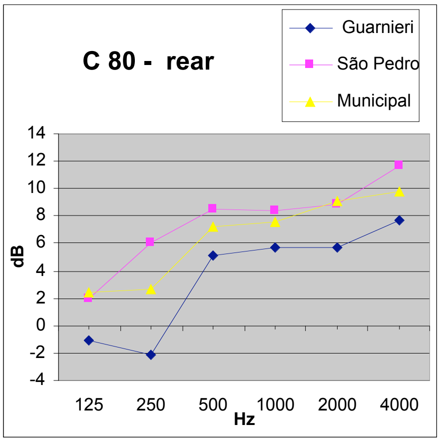

Figure 16: rear audience C 80

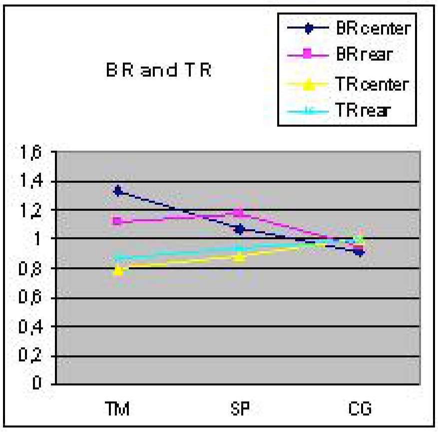

Figure 17: bass ratios and treble ratios

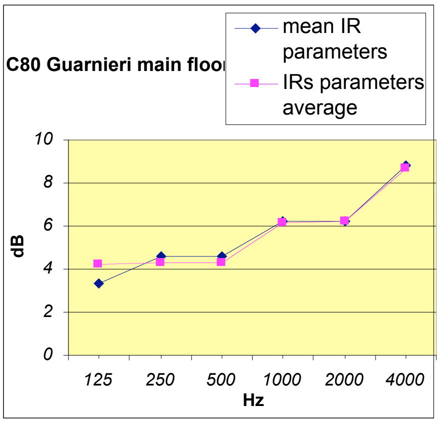

Figure 18: comparing statistics ways

We have noticed that the patterns of the IRs have changed considerably for the different positions of measurement. Generally we have got short and dry IRs for stage measurements (in part because of the directivity of the source) when compared with the measurements made in the center of the audience, partly due to the higher amount of reflection. It was observed – and it is a desirable characteristic in the work of the conductor – that the values of *Clarity* and *definition* in the stage are bigger than those for the rest of the theater. The ITDG values, as one should expect, are lower in the stage (average of 4 ms) than in the audience (average of 13 ms). The measurements in the back of the audience, the IRs indicate a higher diffusion and lower values for the direct/reverberant energy ratio. We could also notice distinct timber colorations, but we will return to that later in this text. As we changed the measurement positions we could clearly perceive a "natural equalization" of the sounds, both in the calculated parameters and in the listening of the anechoic samples.

The architectural characteristic of the theaters can help to understand the graphs above. The Teatro Municipal holds 1500 people in a Italian shaped audience, has a significant part of the floor covered by carpets, and offers 4 floors for the audience, having therefore a much higher ceiling than the others. The Teatro São Pedro, as well as the Anfiteatro Camargo Guarnieri, holds about 700 people and both of them are also used for dramatic arts. The São Pedro has a shell shaped audience while de Guarnieri is basically rectangular. The superficies are significantly different: predominantly masonry in the São Pedro, and wood in the Camargo

Guarnieri. Consequently, both the Teatro Municipal and The Teatro São Pedro presented a higher reverberation time for the lower frequencies in the audience (Figs. 13 and 17) – and consequently less *Clarity* (Fig. 14) – than the Anfiteatro Camargo Guarnieri (in which the wooden walls help in the sound diffusion). For higher frequencies the values of RT60 tend to converge (Figs. 13 and 15), as if the rooms played a less important role in the acoustic behavior of the sounds (again, probably due to directivity of the source for higher frequencies). The large carpet surfaces in the Teatro Municipal collaborate in the reduction of the brightness (Fig. 17) and in the descending curve of reverberation (Figs. 13 and 15). In terms of frequency ratios, the Camargo Guarnieri showed to be more balanced (Fig. 17). Also we can observe an interesting fact when we compare the values of *Clarity* in the center and the back of the halls. In the center, the Camargo Guarnieri presents greater *Clarity* for the majority of the frequencies (Fig. 14), but in back it shows lower values of *Clarity* for all frequency ranges (Fig. 16). It is not by chance: the measured back seats of the Teatro Municipal and the Teatro São Pedro are balconies, while in the Camargo Guarnieri, which does not have balconies, the back seats are part of the main floor. As Beranek observes [4], depending on the height and the depth of the balcony, it can act as a barrier, specially for the reflections coming from above, and in this case it can generating higher values of *Clarity*. The values of RT60 of the Teatro Municipal, as well as its architectural design, would characterize it more as an Opera House than a symphonic Concert Hall. By analyzing the values measured for the Anfiteatro Camargo Guarnieri and the Teatro São Pedro, specially if we consider their size and acoustic treatment, we can conclude that they present a series of acoustic characteristics that are far from international standards for Concert halls, specially if we compare with values obtained from the measurements done on acknowledge great concert halls.

5. ACKNOWLEDGEMENTS

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6. REFERENCES

[1] F. Iazzetta, F. Kon and F. Silva, AcMus: Design and Simulation of Music Listening Enviroments, *Anais do XXI Congresso da Sociedade Brasileira de Computação*, Fortaleza, Brazil, 2001

[2] A. Farina, Acoustic Quality of Theaters: Correlation Between Experimental Measures and Subjective Evaluation, *Applied Acoustics* **62** , 889 – 916, 2001

[3] ISO 3382 *Acoustics – Measurement of the reverberation time of rooms with reference to other acoustical parameters*, 1997

[4] L. Beranek, *Concert halls and opera houses: music, acoustics, and architecture*, Springer-Verlag, New York, 2004

[5] H. Kuttruff, *Room Acoustics,* Elsevier Applied Science, London, 1991

[6] B. Masiero, *Estudo e Implementação de Métodos de Medição e Resposta Impulsiva em Salas de Pequeno Porte*, Relatório Técnico - Fapesp, at http://gsd.ime.usp.br/acmus/public.html , 2004

[7] S. Müller and P. Massarani, Transfer Function Measurements with Sweeps, *JAES* **49**(6), 443, 2001

[8] F. Arruda, R. Guedes et al. A influência da direcionalidade de fontes sonoras na avaliação experimental da qualidade acústica de salas*, XIX Meeting of the Brazilian Society of Acoustics*, Belo Horizonte, 2000