AN HYBRID APPROACH OF LOW FREQUENCY ROOM EQUALIZATION: NOTCH FILTERS BASED ON COMMON ACOUSTICAL POLE MODELING

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ABSTRACT

In this paper, a new method of low frequency multiple-point equalization for room acoustics is presented. This approach is theoretically based on Common Acoustical Pole (CAP) modeling of the room transfer functions developed previously by Haneda et al. [1]. With the hybrid approach, the multiple-point equalizer is made up with cascaded notch filters which coefficients are directly determined from the CAP model. This new method reduces the complexity of the equalizing structure in addition to an automatisation of notch filters design which should be very attractive to many applications such as home theater systems by intelligently controlling room modes.

1. INTRODUCTION

Sound transmission between a source and a receiver in a listening space is well characterized by its Room Transfer Function (RTF) given by physical laws. RTF indicates the distortion undergone by the signal during its propagation in the listening space. Since RTF depends on source and receiver positions [2], many studies were devoted to finding multiple-point equalization solutions over the entire audible spectrum extending from 20 *Hz* to 20 *kHz* [3, 4]. However, other researches [5, 6, 7, 8] limit the equalization process to the low frequency part of the audio range. This simplification is advised by the good separation of room modes at low frequencies in addition to the quasi omnidirectional radiation of dedicated acoustic sources over this frequency band.

This work presents an hybrid method of multiple-point room equalization at low frequency. The technique proposed will benefit from the CAPZ-modeling of RTF by well estimating the room resonances, common to all the receiver positions [1].

In order to reduce the complexity of the equalizer which means minimizing the number of the notch filters needed, we are going to compensate for the most audible modes over the frequency band of interest. From resonance parameters (central frequency, maximum gain and Q-factor), we will design using White method [9] digital biquadratic (or notch) filters which will correct the selected peaks.

In the next section, we will discuss the relevance of limiting the room equalization process to low frequencies. We will present concepts like source directivity and Schroeder frequency which will help us to well understand why equalizing the whole response is almost hard to achieve. In section three, we will talk about the new method. We will present its principles and show its originality. The fourth section will focus on the practical tests and results obtained. After having described the experimental conditions of measuring room impulse responses, we are going to show the performance obtained in two cases: the CAP equalization method described in [5] and the new hybrid technique. The last section concludes the paper.

2. RELEVANCE OF LOW FREQUENCY EQUALIZATION

Traditional room equalization techniques focus on compensating RTF irregularities over the whole audio range extending from 20 *Hz* to 20 *kHz*. Such process can be called into question for reasons that will be later discussed.

2.1 RTF variability at high frequencies: the influence of source directivity

If the acoustic source used to measure a RTF is omnidirectional, its dimensions are negligible compared to the wave length, it reproduces all the frequencies with the same sound pressure level over the 4π solid angle. However, real-life transducers can approach the behavior of simple sources just over the low-frequency range of the audible spectrum. Thus, a RTF measured out-off the loudspeaker central axis is highly influenced by the source directivity especially at high frequencies.

Figure 1 shows the frequency response of a typical full-bandwidth loudspeaker system[1] measured at different angles from its central axis.

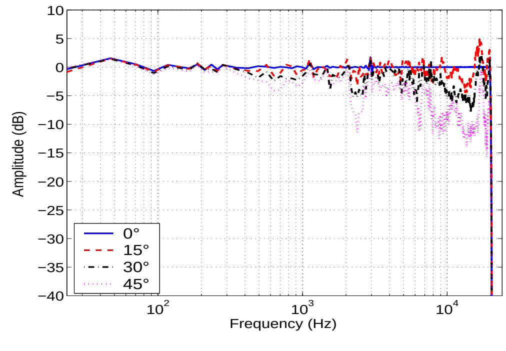


Figure 1: Frequency responses of a full-bandwidth loudspeaker system.

As we can see from this figure, the sound source radiates almost in the same way for all directions only for low frequencies, that is for frequencies under 300 *Hz*. Above almost this frequency, speaker response variations from one angle to another become more and more important.

[1]These measurements are relative to a Cabasse full-range source measured in anechoic room.

2.2 RTF peaks separability at low frequencies

The sound propagation in an enclosure is governed by physical laws. Through room acoustics theory, we know that sound transmission corresponds to a superposition of normal modes which are characteristic solutions of the wave equation satisfying some boundary conditions. We also know that the number of room modes N_f excited over a $[0, f]$ frequency range, can be approximated by the following formula [10]:

$$N_f = \frac{4\pi}{3} V \left(\frac{f}{c}\right)^3 \tag{1}$$

where V is the room volume and c is the sound velocity.

It is obvious that the number of modes N_f increases rapidly when f gets higher. For low frequencies, the peaks are well separated. The more frequency increases, the more the RTF changes rapidly with the receiver position [2]. At a certain frequency f_{Sch}, the modal density becomes so high that resonances will increasingly overlap, creating therefore a diffuse field. The Schroeder frequency is then defined as the limit between the frequency range of negligible and significant modal overlap. This frequency is given by [10]:

$$f_{Sch} = 2000\sqrt{\frac{T_{60}}{V}} \tag{2}$$

where T_{60} is the reverberation time of the room.

2.3 Interpretations

Exceeding the Schroeder frequency, the combined effects of room modes and source directivity lead to a very complex multiple-point RTF equalization process. However, at low frequencies, room modes are well separated and independent from receivers' location in the room. Additionally, the speaker is omnidirectionnal and does not influence RTF measurements. It's then more suitable to limit correction to the low frequency part of the audio range.

3. DESCRIPTION OF THE HYBRID APPROACH

3.1 Multiple-point RTF equalization by using Common Acoustical Pole Modeling

Haneda et al. [1] have proposed an ARMA modeling of RTF whose AR part represents the room modes or resonances. These poles are acoustically independent from the source-receiver position in the room. Such model is called CAPZ model (for **C**ommon **A**coustical **P**ole and **Z**ero) and has the structure given by figure 2. The signal received at a given listening position (i, j) in the room can be written:

$$Y_{(i,j)}(z) = H(z)\frac{B_{(i,j)}(z)}{A_{CAP}(z)}X(z) = H(z)\frac{B_{(i,j)}(z)}{1+\sum_{l=1}^{P} a_l z^{-l}}X(z) \tag{3}$$

where $A_{CAP}(z)$ is the AR common part, P is the number of the modeled poles, $B_{(i,j)}(z)$ is a residue function related to the receiver (i, j) and $H(z)$ is an equalizing filter. This modeling allows therefore a multiple-point equalization that compensates for room modes effects [5]. To do that, we have to apply a multiple-point all zeros filter which is the AR part of the CAPZ model, so that the equalizer $H(z)$ of figure 2 is given by:

$$H(z) = A_{CAP}(z) = 1 + \sum_{i=1}^{P} a_i z^{-i} = \prod_{i=1}^{P}(1 - p_i z^{-1}) \tag{4}$$

where a_i are the coefficients of the equalizer and p_i the poles of the ARMA model: independent from the receiver position in the room.

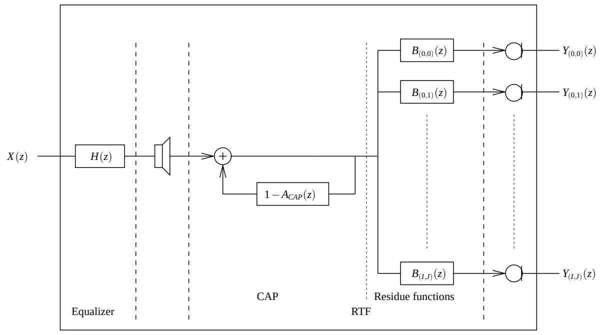


Figure 2: CAPZ Modeling of RTF and hybrid equalization method.

3.2 Principles of the hybrid method

The strong point of the CAP-based equalization technique is that it takes into consideration the physical properties of the room for modeling resonances by common acoustical poles at different receiver positions. However, compensating for room modes by applying the FIR filter $A_{CAP}(z)$ presents some drawbacks. In fact, the wider is the frequency band to control, the longer is the equalizer (equation 1). This problem becomes more critical at high sampling rates with high order FIR filters. Additionally, through the direct use of the all-zero equalizing filter $A_{CAP}(z)$ obtained from the CAPZ modeling, some undesirable boundary affects appear out of the frequency band of interest. Such behavior induces amplifier saturation and creates band interferences for a multi-way loudspeaker system. We require then some shaping like band-pass filtering or out of band flattening the frequency response of the equalizer [6]. As we know, these solutions and their combination bring to other problems like phase matching or hard implementation cost.

In practice, the control of room modes is often achieved by cascaded notch filters [7, 11]. For the design of each biquadratic cell, an operator has to specify several parameters as the central frequency of the notch, the corresponding maximum gain and the Q factor or the bandwidth at $-3\ dB$ to the maximum gain.

The idea here is to automate the design of the parametric filters needed by using the CAPZ model which, as shown before, well estimates the room resonances. By doing so, we reduce the complexity and facilitate the implementation of the equalizer. Moreover, boundary effects will arise since out of the notching band, a biquadratic filter have a unity gain.

The new approach can be summarized as follow:

Step 1: CAPZ model and extraction of resonance parameters

1. First, we make a CAPZ modeling of the RTF, where the AR part estimates well the resonant modes of the room.

Each resonance is modeled by a couple of complex conjugate poles. The estimation can be improved by adjusting the order P of the poles [1]. From equation 3, the multiple-point equalization filter $A_{CAP}(z)$ is given by:

$$A_{CAP}(z) = 1 + \sum_{i=1}^{P} a_i z^{-i} = \prod_{i=1}^{P}(1 - p_i z^{-1}) \qquad (5)$$

Each complex pole p_i can be written as:

$$p_i = g_i e^{j\phi_i} \qquad (6)$$

where g_i and ϕ_i are the amplitude and the phase of the pole p_i.

2. From the CAPZ model we can calculate the parameters of each resonance. These parameters are the central frequency, the maximum gain and the bandwidth at $-3\ dB$ to the maximum gain.
   - **The central frequency** f_i and the relative pulsation ω_i of each resonance can be obtained from the phase ϕ_i as follow:

     $$\omega_i = 2\pi f_i = \frac{F_s}{2\pi}\phi_i \qquad (7)$$

     where F_s is the sampling frequency.
   - **The gain** G_i of the resonance i is therefore:

     $$G_i = \left|(1 - p_i z^{-1})(1 - p_i^* z^{-1})\right|_{z=e^{j\omega_i}} \qquad (8)$$

   - **The bandwidth** $\Delta\omega_i$ of the resonance is given by:

     $$\Delta\omega_i = |\omega_{i,1} - \omega_{i,2}| \qquad (9)$$

     where $\omega_{i,1}$ and $\omega_{i,2}$ are the pulsations corresponding to an amplitude level of $-3\ dB$ from G_i. Theses pulsations are solution of the following equation:

     $$\left|(1 - p_i z^{-1})(1 - p_i^* z^{-1})\right|_{z=e^{j\omega_i}} = \sqrt{2}G_i \qquad (10)$$

3. From the estimated poles we will compensate just for the most audible [12] P' resonances ($P' < P$). This step reduces the complexity of the equalizer by limiting the number of notch filters needed to correct the measured RTF.

Step 2: Design of notch filters

Each room resonance i ($i = 1..P'$) can be controlled by using a biquadratic filter $H_i(z)$ which transfer function is given by:

$$H_i(z) = \frac{b_0^i - b_1^i z^{-1} + b_2^i z^{-2}}{1 - a_1^i z^{-1} + a_2^i z^{-2}} \qquad (11)$$

The coefficients b_0, b_1, b_2, a_1 and a_2 of the filter $H_i(z)$ are related to the selected resonance parameters through the following formulas [9]:

$$\begin{aligned} b_0^i &= (1 + \lambda_i^2 + X_i)/D_i \\ b_1^i &= 2(1 - \lambda_i^2)/D_i \\ b_2^i &= (1 + \lambda_i^2 - X_i)/D_i \\ a_1^i &= b_1^i \\ a_2^i &= (1 + \lambda_i^2 - Y_i)/D_i; \end{aligned} \qquad (12)$$

where $\lambda_i = tan(\omega_i/2F_s)$, $X_i = G_i Y_i$, $Y_i = (S_i(1 + \lambda^2)/G_i)tan(\Delta\omega_i/2F_s)$, $S_i = \sqrt{1 - 2G_i^2}$ and $D_i = 1 + \lambda_i^2 + Y_i$.

The equalizer $H(z)$ of figure 2 is then a cascade of $P^{'}$ biquadratic filter:

$$H(z) = A_{BIQ}(z) = \prod_{i=1}^{P^{'}} H_i(z) \qquad (13)$$

We recall that $P^{'}$ is the number of the most audibles resonances selected by using the criterion given in [12].

3.3 Novelty and strong points of the proposed method

The new approach brings some innovation in the field of low frequency multiple-point equalization of RTF:

- It has a good robustness since it is based on CAPZ modeling RTF.
- It reduces the complexity of the equalizer. Indeed with the new method, the equalizer requires just cascaded biquadratic cells compared to an equivalent high order FIR filter.
- The user needs no more to define for each notch filter the parameters f_i, G_i and $\Delta\omega_i$. These characteristics are automatically determined thanks to the CAPZ modeling of the measured RTF.
- With a cascade of notch filters we do not have the problem of high increase of the equalizer transfer function in the vicinity of the band of interest.

4. EXPERIMENTAL RESULTS

4.1 Measurements procedure

The room impulse responses were all measured in a small living room [2]. The enclosure size was of $(6.5 \times 5.5 \times 2.7)m^3$. It has a reverberation time $T_{60} = 0.52\ s$. The Schroeder frequency f_{Sch} of this room approached by equation 2 is equal to 145 Hz. The loudspeaker we used was a woofer, which is quasi omnidirectionnal over the $[20Hz - 145Hz]$ frequency range.

As it is shown in figure 3, RTF measurements were taken at 25 positions. A microphone is so displaced from one position to another. Each position is referenced with its (i, j) coordinates. The sampling frequency F_s was set to 48 kHz. Only 12 RTF, those related to the gray area, were taken into consideration for the equalization process. This is due to the fact that the receivers' probability of presence is more important in this area than in the others.

Figure 4 shows at 3 positions ($(2,4)$, $(3,4)$ and $(4,4)$), the measured room responses over the $[20Hz, 145Hz]$ frequency band. From this figure, we can notice essentially two points. The first one concerning the variability of RTF with the receiver position. The second point is relative to the stability of the room resonances over the considered frequency range. Indeed, the most dominant peaks 26 Hz, 29 Hz, 53 Hz, 65 Hz, 82 Hz, 107 Hz and 132 Hz are well separated and common to the different positions.

[2] Measurements were done in the commercial auditorium of Cabasse Acoustic Center.

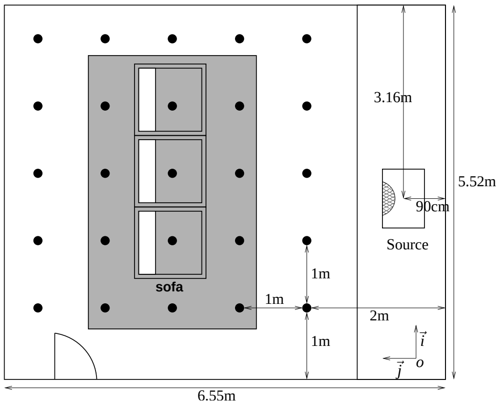


Figure 3: Experimental protocol.

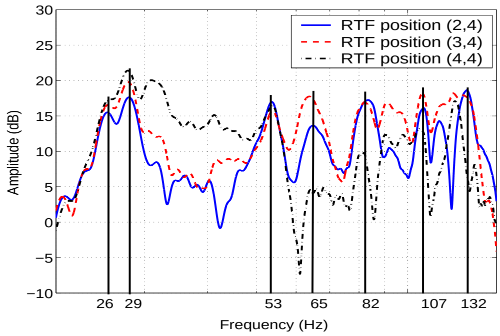


Figure 4: Room Frequency Responses at three positions.

4.2 Results

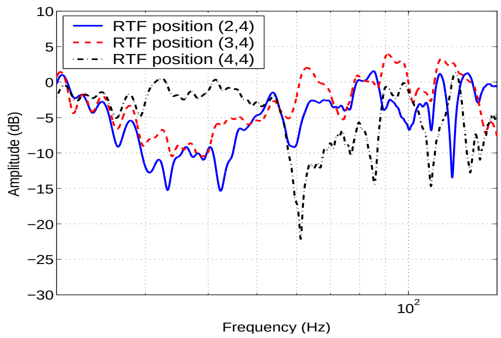
Figure 5 (a) presents the performance of the equalization obtained with the CAP filter described in [5]. In the other side, figure 5 (b) illustrates the performance of the new hybrid method described in this paper. All the results are related to the three positions in the sofa.

The first results were obtained by applying the CAP equalizer $A_{CAP}(z)$ to the RTF decimated to $F_s = 500\,Hz$. The estimation process was done by the Least Squares Method. So we calculated the poles' theoretical order $P = 2N_f$ (equation 1). In our case, the frequency range was from 20 Hz to 145 Hz. We therefore obtained 64 ($c = 343\ ms^{-1}$) as the theoretical P order value. This order was then reduced to 36 by minimizing the normalized mean-squared output error criterion introduced in [1].

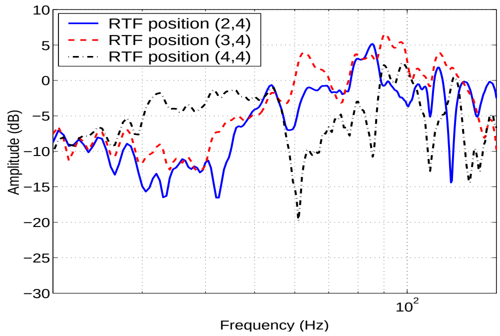
The second results were obtained by applying the equalizer $A_{BIQ}(z)$ (equation 13). By comparing the parameters of the estimated resonances (eighteen in this case) with the detection thresholds given in [12], the eleven most audible resonances ($P' = 11$) have been selected. The numerical parameters of these peaks are summarized in table 1. These parameters are the central frequency f_i, the central gain G_i and the Q-factor: Q_i of each resonance given by:

$$Q_i = \frac{2\pi f_i}{\Delta\omega_i} \tag{14}$$

where $i = 1..P'$.



(a) Using the CAP equalizer [5].



(b) Using the new hybrid method.

Figure 5: Low frequency multiple-point room equalization.

The compensation of the peaks corresponding to the selected resonances specified by table 1, is shown by figure 5 (b). Indeed, by cascading the designed digital notch filters, we reduce averagely the amount of the common resonant peaks.

Table 2, evaluates the complexity of the two equalizers $A_{CAP}(z)$ and $A_{BIQ}(z)$. The first one is interpolated to 48 kHz. However, the cascade biquadratic sections, used in the $A_{BIQ}(z)$ equalizer, are directly designed at $F_s = 48\,kHz$. From the values given in this table we conclude that the new hybrid approach reduces the data memory occupation with about 98%. Additionally, when using the CAP based equalizer, almost 7104 simple instructions (multiplication or accumulation) are needed by the DSP versus only 99 with the new hybrid method.

resonance n°	f_i (Hz)	G_i (dB)	Q_i
1	26	15	8
2	29	14	4
3	40	7	3
4	53	11	8
5	66	8	9
6	82	9	12
7	94	5	7
8	107	12	20
9	121	7	9
10	131	10	11
11	137	7	9

Table 1: Parameters of the detected resonances.

filter	$A_{CAP}(z)$	$A_{BIQ}(z)$
number of coefficients	3552	55
number of instructions	7104	99
σ (dB)	5.25	5.75

Table 2: Comparison between the two equalizing methods.

Another way used to evaluate the performance of the equalization algorithms over the frequency band of interest, was the the standard deviation of the room frequency responses. For a given listening position (i, j), this criterion is given by:

$$\sigma_{(i,j)} = \sqrt{\frac{1}{N}\sum_{f=f_{inf}}^{f_{sup}} (20log\left|H_{(i,j)}(f)\right| - AVG_{(i,j)})^2} \quad (15)$$

where $H_{(i,j)}(f)$ is the frequency response of the room between the source and the listening point (i, j) and $AVG_{(i,j)}$ is the mean value given by:

$$AVG_{(i,j)} = \frac{1}{N}\sum_{f=f_{inf}}^{f_{sup}} (20log\left|H_{(i,j)}(f)\right|) \quad (16)$$

f_{inf} corresponds to 20 Hz, f_{sup} corresponds to 145 Hz and N is the number of the frequency samples between f_{inf} and f_{sup}.

The average standard deviation σ over all the considered listening points is then equal to:

$$\sigma = \frac{1}{M}\sum_{i,j}\sigma_{(i,j)} \quad (17)$$

where M is the number of listening points, equal in our case to 12.

In the last line of table 2, we give the average standard deviations of the measured RTF after CAP-based equalization and when using the new hybrid approach. The average standard deviation of the unprocessed RTF is equal to 6.06 dB

The significant reduction of the equalizing structure complexity, combined with the arising of boundary effects, allow us to say that the proposed equalizer is more effective then the CAP-based one. This conclusion needs to be confirmed by subjective listening tests since the average standard deviation of the room responses over the considered listening area is 0.5 dB more important than the CAP-based equalization.

5. CONCLUSIONS

We have presented a new method for low frequency equalization at multiple listening positions. The new model is based on the CAPZ technique. After having estimated the common poles by least squares method, we select, some peaks which need correction. From the parameters (given by the AR part of the CAPZ model) of each resonance, which are, the central frequency, the maximum gain and the bandwidth, we later design digital notch filters which exactly compensate the desired resonances. The effectiveness of the proposed equalization is compared with that of a conventional CAPZ technique. According to the results obtained, the new approach allows a substantial complexity reduction in addition to arising some technical problems as the boundary effects.

6. ACKNOWLEDGMENT

This work was supported by Cabasse Acoustic Center. The authors would like to express special gratitude to Yvon Kerneis, R&D director of the Cabasse Acoustics Center, Bernard Debail, acoustic engineer and Pierre Yves Diquèlou, project manager in the supporting company.

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