A Power Spectrum Density Model for 5.1 Audio Cinema System Using the Fast Fourier Transform

Iván Luna-Sánchez, Gerardo M. Chávez-Campos, Adriana C. Téllez-Anguiano, Enrique Reyes-Archundia, and Arturo Méndez-Patiño

***Abstract*—Digitalization in cinema projection has promoted a more efficient monitoring and projection systems, this allows improving supervision and projection status worldwide. Nevertheless, the monitoring systems do not include the ability to detect audio failures. In this paper a power spectrum density model for a 5.1 audio cinema system is proposed. The model was determined by using data from a DSP system located into a demo projection hall. The DSP algorithm performs the white-noise emission, data acquisition, feature extraction and failure classification routines. The FFT and power spectrum density algorithms allow characterizing non-failure and failure sound system behaviors. The model is able to classify low and high levels of sound pressure, the absence of specific channels, as well as the missing of low, mid and high drivers at frontal and sub-woofer channels.**

***Index Terms*—FFT, audio system, power spectrum density, digital signal processor.**

I. Introduction

Cinema industry has become economically attractive for motion pictures production and projection. Actually worldwide cinema incomes have surpass the USD\$33,000 millions, only in the projection hall rooms [1], [2]. Specifically, Mexico reports incomes close to MXN\$10,000 millions [2], [3].

Digitalization in cinema projection has promoted a more efficient monitoring and projection system, known as *Network Operation Center* (NOC) [4], [5]. The NOC is implemented in cinemas, allowing to verify the status of the projection in every hall by several test processes [6] and reporting in real-time when a failure occurs. The NOC also allows to perform preventive maintenance tasks, such as lamp changes.

Nevertheless, the sound performance is not considered in this monitoring system, due to its own analog nature [7], [8], which is why sound failures still need to be reported by audience or staff, consequently, some failures remain unnoticed for a long period of time [9]-[13].

This document reports the implementation and results of a spectrum-density-based system applied in a 5.1 cinema sound system hall room. This system uses a DSP card and conditioning circuitry to process a white-noise audio pattern. A *Fast Fourier Transform* (FFT) algorithm was implemented in order to propose a *Power Spectrum Density* model to evaluate the status and failures in the sound system.

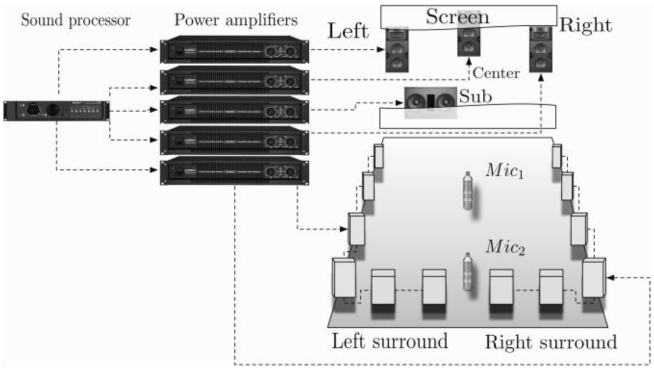

Fig. 1. The 5.1 Sound System configuration and microphone positions: Mic_1 for equalization according to *SMPTE 202* standard [17] and Mic_2 for data acquisition.

II. Modeling the Cinema Sound System

Cinema Sound System Considerations

The 5.1 sound system, shown in Fig. 1, is conformed by three frontal channels (left, center and right with X-Curve function activated) [14], two surround channels (left and right) [15] and one sub-channel with 1/10 of the total audible spectrum (sub-woofer) [16].

Each frontal or surround channel can be bi-amplified or tri-amplified, known as 2-ways and 3-ways amplification. In this case, a 3-way amplification system was used; this system is conformed by low, high and midrange-band drivers for frontal and surround speakers.

In order to equalize the 5.1 sound-system the SMPTE-202 standard was used due to its wide acceptance [17], [18], this equalization is known as *X-Curve*. The *SMPTE-202* allows using minimal equalization to produce a response within the *X-Curve* window, as a consequence the room reverberation is mitigated in the measurement, reflecting only the acoustical coupling problems (speaker, screen, and room) [19].

The 5.1 sound-system was installed in a demo hall room (7m length and 5.5m width) to perform sound measurements. These measurements were performed without seats or screen; one Sound Processor CP650 [20] and one professional QSC amplifier ISA 750 system [21] were used to compare patterns and amplify the white noise emission. The equalization was done using a CM-10 omnidirectional microphone by Audiocontrol [22] and a *Real Time Analyzer* (RTA), according to the *SMPT-202* standard and the Dolby Installation Manual. The microphone (*Mic_1*) was located at 2/3 of room's length from the screen and at 5ft height, as shown in Fig. 1.

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TABLE I: SPL LEVELS ACCORDING TO X-CURVE

SOUND ZONE	SPL(dB)
FRONTAL	85
SURROUND	82
SUBWOOFER	90

The Sound Pressure Levels for each channel in the 5.1 sound-system are shown in Table I.

III. SOUND PRESSURE, POWER SPECTRUM DENSITY AND FFT

The Sound Pressure Level (SPL) is defined in (1). The SPL determines the ratio of the pressure of the sound P and a reference pressure P_{ref}, unless otherwise is specified the reference pressure is 0.00002 N/m^2 (20 µPa) [23], [24].

$$SPL = 20 \cdot \log_{10}(\frac{P}{P_{ref}})dB \tag{1}$$

The SPL is a single threshold rule commonly used in audio pattern recognition problems [25]-[27], nevertheless (1) offers low identification margins; consequently, methods that extract more features from signals are preferred.

The *Power Spectral Density* (PSD) is a transformation method defined in (2).

$$PSD(f,T) = \frac{1}{T}\left|\int_{-T/2}^{T/2} x(t)\exp(\frac{-j2\pi}{f}t)dt\right|^2 \tag{2}$$

where $x(t)$ is the time variant signal during a time window of length T. However, (2) remains as a single value rule of the spectrum density [26], making difficult the appropriate failure identification. Therefore, a method to extract the power spectral density by ranges, similar to frequency bands, is proposed.

IV. THE POWER SPECTRUM DENSITY MODELS

A. Non-Failure Model

In order to obtain the PSD model for the 5.1 sound system, a Digital Signal Processor (DSP) System and an own-designed pre-amplifier PCB were implemented as shown in Fig. 2. Before each test the 5.1 sound was equalized according to Table I. The DSP system is based in the DSK TMS320C6713 [28].

The DSP was programmed according the algorithm shown in Fig. 3. This algorithm is basically divided in three parts*: I) white-noise emission \& data acquisition, II) feature extraction and III) failure classification* [26]. The routine *I)* was used to obtain data, the routine *II)* to calculate the average models (dashed line Fig. 3), this routine uses equations (3), (4), (5) and (6).

Equation (3) represents the total power spectrum density, i.e., the sum of all frequency contributions of $X(k)$ defined in (7). The term $X(k)$ is the Fast Fourier transform of a digital signal $x(t)$, with sampling period T (implied in $X(nT)$), N is the frame length and *exp* terms represent the phase factors as functions of N [24], [29].

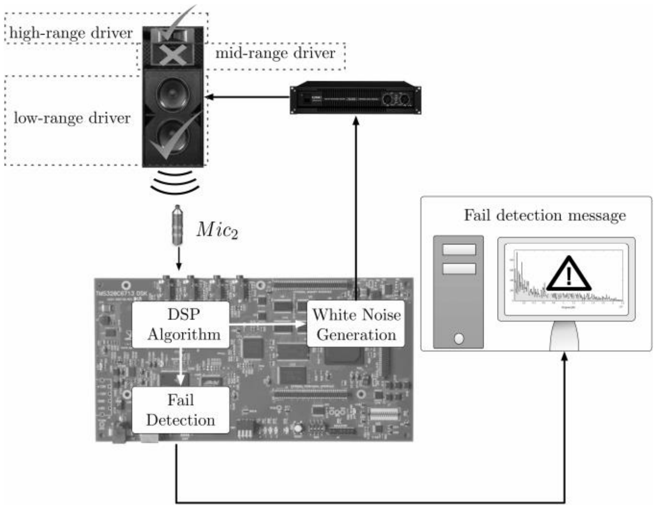

Fig. 2. DSP used to determine PSD model.

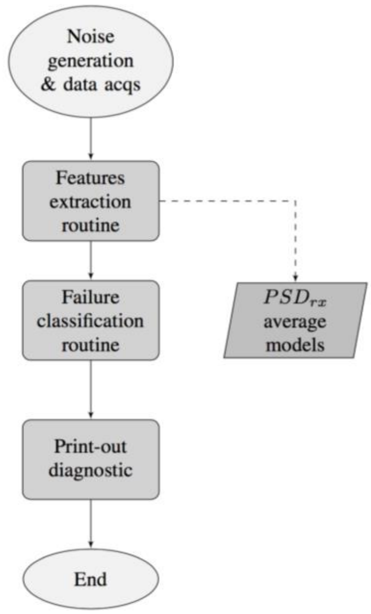

Fig. 3. General algorithm for features extractions and failure classification modes for 5.1 audio cinema system.

$$PSD_T = \sum_{0}^{255} G_1 \cdot X(k) \tag{3}$$

$$PSD_{r1} = \sum_{1}^{6} G_2 \cdot X(k) \tag{4}$$

$$PSD_{r2} = \sum_{7}^{57} G_3 \cdot X(k) \tag{5}$$

$$PSD_{r3} = \sum_{58}^{255} G_4 \cdot X(k) \tag{6}$$

$$X(k) = \sum_{n=0}^{\frac{N}{2}-1} X(nT)\cdot\exp(\frac{-j2k\pi}{N}) \quad k = 0,1,...,N-1 \tag{7}$$

On the other hand, equations (4), (5) and (6) are proposed to divide PSD in frequency ranges similar to frequency bands. In order to obtain the non-failure PSD_{rx} models for the 5.1 sound system, 30 experiments were performed at each channel (frontals, left surround, right surround and sub-woofer). The obtained data was statistically analyzed in order to define maximum and minimum values in each range.

B. Failure Model

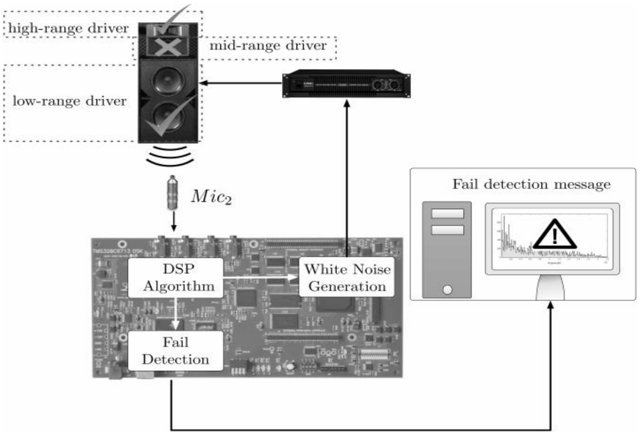

Fig. 4. Experimental procedure for failure induced experiments at central channel.

Eventually experiments with induced failures were done, Fig. 4 depicts the induced failure into the center channel at mid-range driver. Several fails were induced into each channel, for central speakers: *low SPL, high SPL, non-low-range, non-mid-range and non-high-range drivers*. In the surround channels, the induced failures were provoked by the absence of an specific speaker; failures in the Sub-woofer channel were limited to the complete absence of the channel.

The routine *III)* for frontal speakers is shown in Fig. 5; different routines were developed for surround and sub-woofer channels. A nested algorithm inspects minimum and maximum threshold values for PSD_{r1}, PSD_{r2} and PSD_{r3}. Thus, the routine was able to detect the absence of low, mid and high range drivers, as well as changes in the sound pressure levels. The routine ends by sending a failure message.

V. Results and Discussion

Table II presents the total and sectioned *PSD* data for the frontal speakers upon the conditions mentioned above. In every case the approximate (≈) SPL in dB is reported at the left corner. Fig. 6 depicts data from Table II for the central channel, comparing ideal and failing behaviors.

Table III presents total and sectioned sums for surround channels under ideal conditions and considering the absence of a specific speaker. Note that PSD_T is enough to determine the absence of an speaker and indicate which speaker is missing.

In contrast, PSD_{rx} are not enough to diagnose a driver missing failure (low, mid or high), this may be attributable to the surround speakers array. When a driver or speaker is damaged, the PSD_{rx} results are different of the normal behavior range due to the impedance compensation levels by the sound processor. Hence, data should be sectioned in more narrow frequency ranges for surround channels.

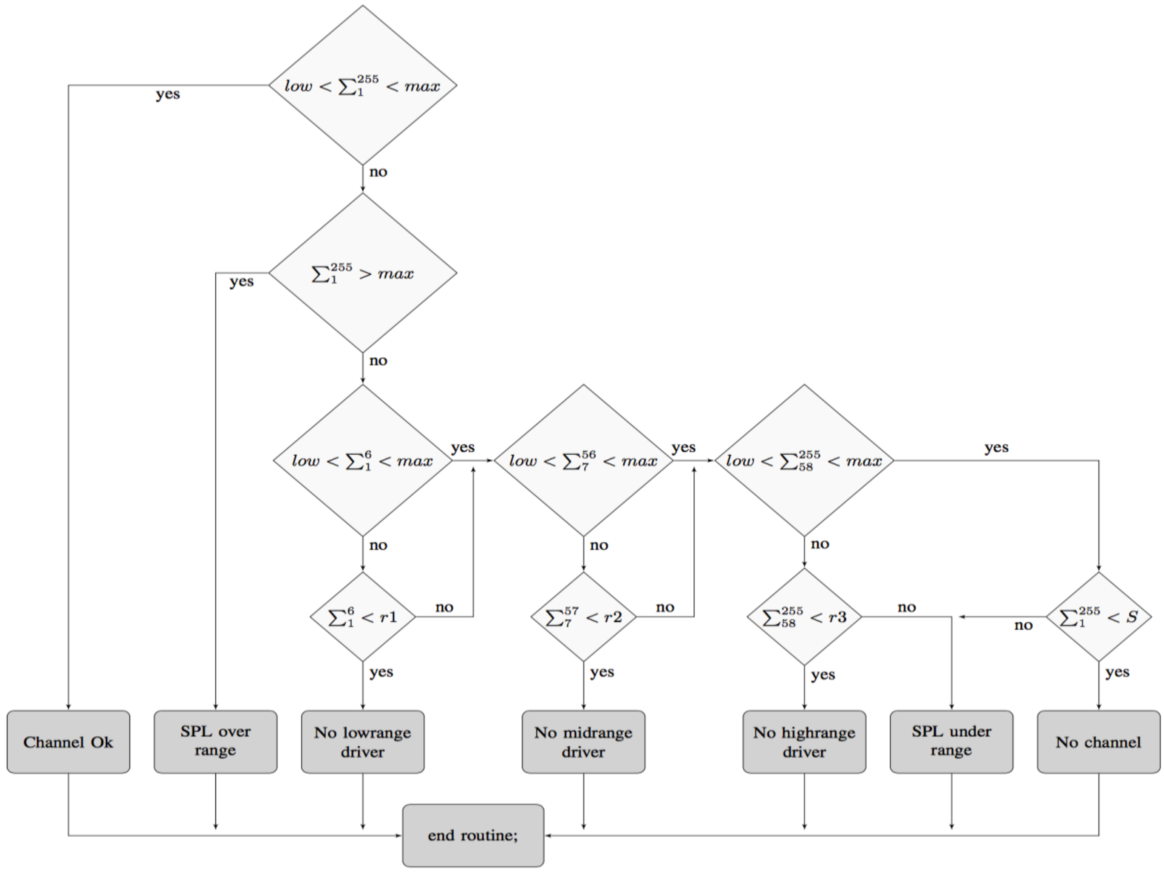

Fig. 5. Failure classification routine implemented in the DSP system for frontal speakers.

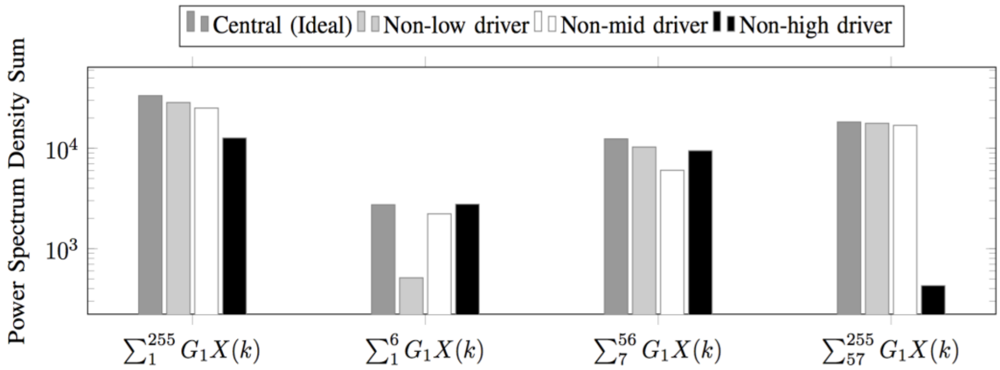

Fig. 6. Frontal-Central Speaker Power Spectrum Density Sums for Ideal conditions, No lowrange driver, No midrange driver, No highrange driver.

TABLE II: AVERAGE VALUES AT IDEAL CONDITIONS AND FAILURES INDUCED FOR FRONTAL SPEAKERS

	PSD_T $\sum_{k=1}^{255} G_1 \cdot X(k)$	PSD_{r1} $\sum_{k=1}^{6} G_2 \cdot X(k)$	PSD_{r2} $\sum_{k=7}^{57} G_3 \cdot X(k)$	PSD_{r3} $\sum_{k=58}^{255} G_4 \cdot X(k)$
SPL[dB]≈ $85db$		IDEAL CONDITIONS		
LEFT	33391	2737	6572	9872
CENTER	28422	2484	10280	15658
RIGHT	32826	2513	11080	19233
SPL[dB]≈ $83dB$		LOW SPL INDUCED		
LEFT	25070	1958	9449	13663
CENTER	22164	1891	7646	12627
RIGHT	25543	1948	8524	15072
SPL[dB]≈ $86dB$		HIGH SPL INDUCED		
LEFT	37309	2918	13418	20973
CENTER	31816	2778	11883	17155
RIGHT	35608	2738	11512	21358
SPL[dB]≈ $84dB$		NO LOWRANGE DRIVER		
LEFT	28513	513	10285	17715
CENTER	19666	338	7143	12186
RIGHT	29211	462	9442	19307
SPL[dB]≈ $83.6dB$		NO MIDRANGE DRIVER		
LEFT	25135	2223	6031	16881
CENTER	18179	2185	4100	11894
RIGHT	29115	2635	7268	19212
SPL[dB]≈ $82.3dB$		NO HIGHRANGE DRIVER		
LEFT	12638	2768	9442	428
CENTER	11652	2500	7380	1772
RIGHT	12517	2979	9170	369

TABLE III: SURROUND CHANNELS AVERAGE PAD VALUES

	PSD_{r1} $\sum_{k=1}^{6} G_2 \cdot X(k)$	PSD_{r2} $\sum_{k=7}^{57} G_3 \cdot X(k)$	PSD_{r3} $\sum_{k=58}^{255} G_4 \cdot X(k)$	PSD_T $\sum_{k=1}^{255} G_1 \cdot X(k)$
		LEFT SURROUND		
IDEAL	673	6224	12208	19105
NO SPK_1	735	5566	10490	16791
NO SPK_2	572	5106	10272	15950
NO SPK_3	875	5487	9423	15785
NO SPK_4	692	5156	12340	18189
		RIGHT SURROUND		
IDEAL	568	7332	12490	20390
NO SPK_1	839	6183	12823	19845
NO SPK_2	664	5303	9913	15879
NO SPK_3	851	5256	7669	13776
NO SPK_4	505	4242	12054	16801

VI. CONCLUSIONS

A PSD model for a 5.1 audio system was presented. The white noise pattern and feature extraction supported by FFT technique provide substantial data to propose threshold rules to detect failures. Although the developed model is very efficient to detect critical failures in cinema exhibition by using defined-range sums, when the surround channels fails, the algorithm did not recognize damaged drivers, only complete absence of speakers. Therefore, improvements must be done such as using another noise patterns, specific tones or sound incidence angle detection routines.

Finally, a more complex failure detection technique can be implemented, such as neural network or failure detection and isolation techniques.

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