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Real-Time Correction of the Frequency Response of a Public Address System

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Abstract **– Correcting the public address (PA) system during a concert event is one of the crucial tasks in ensuring acoustic comfort. However, the existing approaches to such correction do not allow for real-time adaptation to changes in the acoustic properties of the venue that occur during the event. To address this limitation, this article proposes the use of a multiband compressor. It is shown that a zero-latency VST plugin can serve as a multiband compressor. Pink noise can be used as a test signal for system calibration. The results of testing the proposed algorithm, conducted through model and real-world experiments, demonstrate the feasibility and effectiveness of the proposed approach.**

Keywords – multiband compressor; equalizer; PA system; dummy head; VST plugin; signal correction.

I. Introduction

The use of equalizers for the correction of acoustic systems is widely practiced and subject to further research [1]–[3]. In the cited works, authors focus on frequency correction of an acoustic system, which includes acoustic equipment and the room. However, the issue of correcting the frequency response of an acoustic system cannot be considered fully resolved. For instance, a drawback of the studies [4]–[6] is the failure to account for changes in the acoustic properties of the room resulting from variations in its occupancy, decorations, and other factors.

Significant contributions to the research on frequency correction have been made by the authors of works [7]–[9], where the use of filters for correcting the acoustic properties of rooms is proposed. However, the drawback of the systems proposed in [7]–[9] is their inability to operate in real-time.

In [10], an equalizer containing a comb of linear-phase octave filters was proposed, while [11] presented a quasi-linear phase octave graphic equalizer with low latency. The drawbacks of these works are that these equalizers were not used for correcting the acoustic properties of the room, and they did not dynamically adjust the signal amplitude. An equalizer whose frequency response varies over time is commonly referred to as a dynamic equalizer [12]. A multiband compressor is an equalizer that splits the input audio signal into multiple filters, with each filter passing through its own compressor [12]. The implementation of such a compressor is described in patent [13], and its structural diagram is shown in Fig. 1. Works [2], [3], [14], [15] describe devices similar to the device in patent [13]; however, they were used for sound frequency enhancement tasks. A dynamic compressor, whose parameters can be controlled by external sources (sidechain), is widely used in the audio industry [2], [3], [14], [15]. The structural diagram of such a compressor is shown in Fig. 2. As stated in [15], these

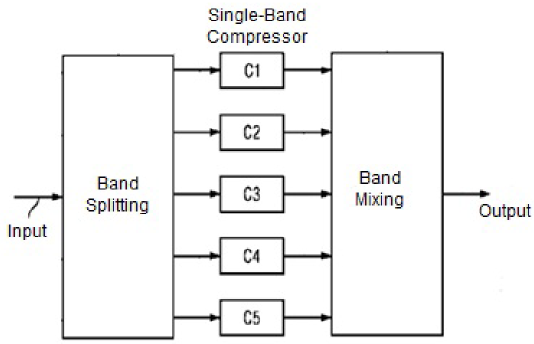

Fig. 1 Structural Diagram of a Dynamic Compressor

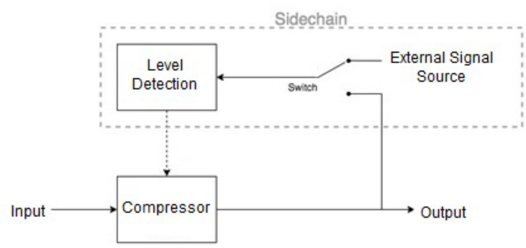

Fig. 2 Structural Diagram of a Compressor with a Control Signal

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compressors are commonly used in studio work. Additionally, compressors can be useful in resolving frequency conflicts within a certain frequency band [3]. An example of such a device is the DBX DriveRack PA2 [16], which has an auto-equalization function that allows for automatic adjustment of the acoustic system to the room. The working principle of such an equalizer is similar to that of a dynamic equalizer. However, a drawback of this approach is that the auto-equalizer does not adjust its parameters in response to changes in the occupancy of the room.

The aim of this study is to address the shortcomings identified in [2], [3], [14], [15], namely, the utilization of a multiband compressor for real-time correction of the acoustic system, taking into account the occupancy of the room.

II. Methods

A. Experimental Methodology

The purpose of this study is to investigate the possibility of real-time correction of the frequency response of the "loudspeaker-room-microphone" (LRM) system using a multiband compressor.

The structural diagram of the room's frequency response correction with the application of a multiband compressor is depicted in Fig. 3. According to this diagram, during system calibration, a test signal in the form of pink noise is applied to the input 1 of the multiband compressor. The signal is radiated by the loudspeaker, passes through the room, and is captured by a pair of microphones on an dummy head (DH) placed within the room. The electrical signal from the microphones' output is fed to the input 2 (sidechain) of the multiband compressor. When the signal level exceeds a certain threshold in a specific frequency channel, the compressor is intended to attenuate the signal, as shown in Fig. 4.

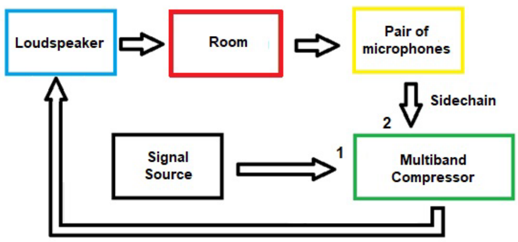

Fig. 3 Structural diagram of the room frequency response correction system using a multiband compressor

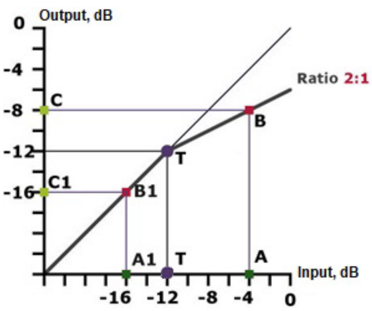

Fig. 4 Operation principle of the compressor[20]

In Fig. 4, the X-axis represents the level of the input signal reaching the compressor, and the Y-axis represents the level of the output signal. If the level of the input signal exceeds the threshold T, the gain of the signal will be adjusted.

The software equivalent of the multiband compressor presented in patent [13] is the VST plugin FabFilter Pro-MB [17], which allows real-time processing. It features a linear phase-frequency response and offers two signal level adjustment modes: compression and expansion. Additionally, it includes a sidechain mode. In this work, we focus only on the compression mode as it is more commonly used in real-world applications. The plugin is praised for its convenience and ease of configuration. The typical operating principle of the VST plugin is illustrated in Fig. 1 and Fig. 2.

The algorithm for signal processing using a multiband compressor consists of the following steps:

1. The signal X(t) is passed through a bank of bandpass filters, resulting in signals $X_1(t)…X_N(t)$ at the filter outputs.
2. The signals $X_1(t)…X_N(t)$ are fed into the inputs of compressors $C_1…C_N$.
3. The control signals for compressors $C_1…C_N$ are the signals $Y_1…Y_N$, which are formed by passing the signal from the microphone output through another bank of filters.
4. Compression of the signal *X(n)* in the n-th channel of the compressor occurs when the level of the signal *Y(n)* exceeds the threshold value T.
5. The parameter Ratio is recommended to be chosen between 1.5:1 and 4:1.
6. The signals from the compressor outputs are summed together.

B. Organization of experimental research

The experimental research of the multiband compressor (MBC) was conducted in two steps. The first step involved simulating the distorting effect of the room on the signal *X(t)* and testing the operation algorithm of the MBC. The signal *X(t)* was artificially distorted by amplifying the high frequencies, resulting in the formation of the signal *Y(t)*. Subsequently, the correction of the signal *Y(t)* was performed to restore the signal *X(t)*.

The second step of the experiment consisted of real-world tests in the room. The signal *X(t)* was distorted by the room, and then the correction of the signal *Y(t)* was

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carried out, taking into account the acoustic properties of the room, in order to restore the signal *X(t)*. The real-world experiments were conducted in the domestic room with dimensions of 12x5x1.5-3.5 m. The following equipment was used:

1. Behringer XR18 digital mixing console (MC1);
2. Shotgun microphone with two Sennheiser e614 microphone capsules;
3. Personal computer with software: X-Air-Edit for controlling the mixing console, Cockos Reaper as a VST host, FabFilter VST plugin package for signal processing;
4. Berhinger Xenyx 502 analog mixing console (MC2);
5. Mobile phone with TMSOFT Noise Generator software for pink noise generation;
6. Sound source in the form of a two-way bi-amped Sven Monitor 5 acoustic system.

III. Results

During the model testing of the MBC a test signal *X(t)* in the form of pink noise was fed into the input of the mixing console MC2, where it was intentionally distorted by amplifying the high-frequency components (Fig. 5). The distorted signal was then routed to stereo input 1 of the mixing console MC1, while the microphone signals from the stereo input 2 were also fed into it. An active speaker was connected to the output of MC1. To minimize the reverberation effect of the domestic room, the microphone was positioned 5 cm away from the loudspeaker.

Signal *Y(t)* from the microphones was used as the control signal for the MBC. Five channels of the MBC were utilized for the experimental researches. The signal processing was performed according to the aforementioned algorithm. The following parameter values were selected for the MBC:

1. Frequency bands: 20-110 Hz, 110-250 Hz, 250-600 Hz, 600-2000 Hz, 2000-7000 Hz, 7000-20000 Hz.
2. The threshold parameter T for each frequency band was set to -18 dB.
3. The parameter Ratio for each frequency band was set to 4:1.
4. The parameter Range for each frequency band was set to -15 dB.
5. Each frequency band was set to Sidechain mode with external microphone control.

The other parameters remained unchanged.

During the real-life experiment, the test signal in the form of pink noise was fed into stereo input 1 of the mixing console (MC1), which was connected to a pair of active speakers of the Sven Monitor 5 system. The signal *X(t)* was reproduced by the speakers, passed through the room, and captured by the pair of DH microphones that were connected as in the previous experiment. The spectrum of the received signal *Y(t)* is shown in Fig. 6. The equipment was placed in the room according to the diagram depicted in Fig. 7.

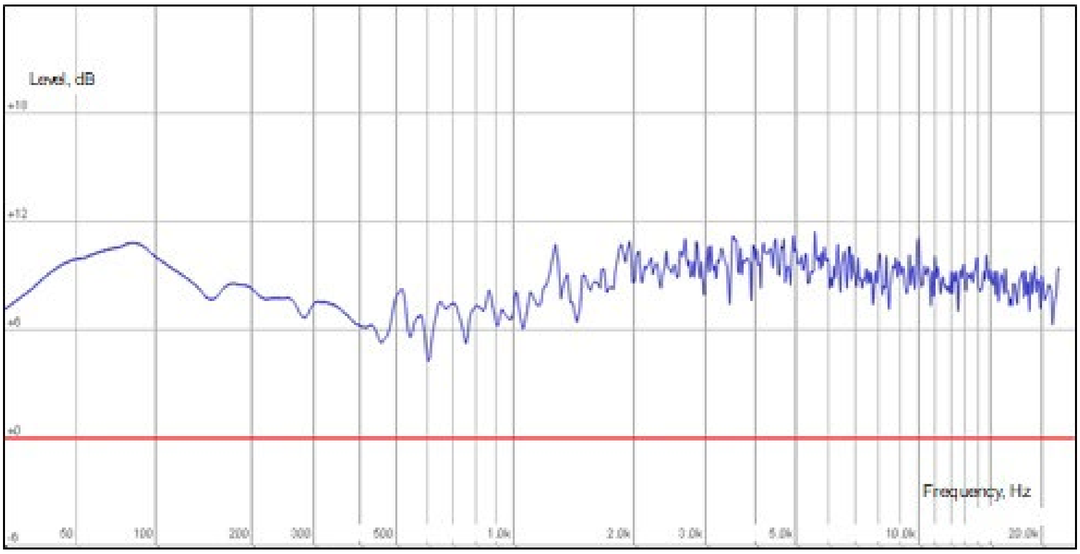

Fig. 5 Artificially distorted signal

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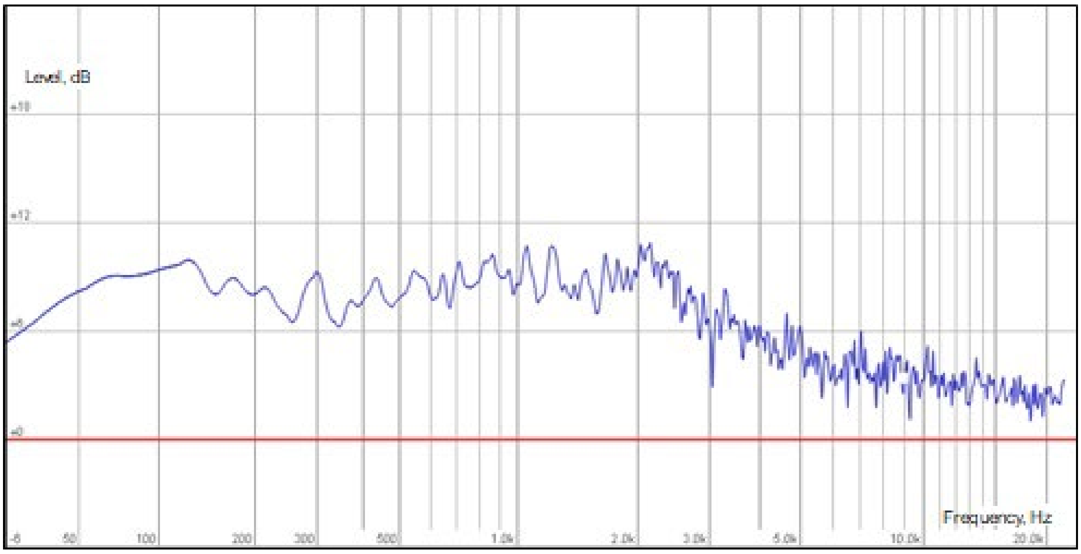

Fig. 6 Signal distorted by the room

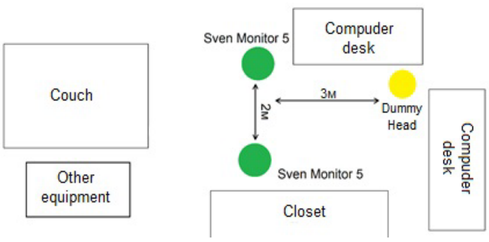

Fig. 7 Equipment placement in the room

IV. Discussion

After processing the signal with the aforementioned parameters of the MBC, the following results were obtained. Fig. 8 shows the spectrum of the corrected signal that was artificially distorted (a) and affected by the room (b).

The analysis of Figure 8 indicates that the chosen parameters of the MBC were able to restore the signal with an error of up to 3 dB, which can be considered acceptable. It should be noted that for more accurate correction, it is necessary to use a greater number of frequency bands and optimize the parameters of each band, which is planned to be implemented in future work.

During the experiments, a significant drawback was identified. Since signal processing is done using a computer, there is a time delay caused by the sum of Analog-to-Digital Converter (ADC) delay + processing + Digital-to-Analog Converter (DAC) delay.

During the experiment, the total delay was found to be 20 ms at a sampling rate of 48000 Hz. The processing delay was approximately 4 ms, and the ADC+DAC delay was 16 ms. Reducing the sampling rate can decrease the delay, but subjectively, the reproduced signal is perceived as degraded. Decreasing the buffer memory of the ADC can significantly reduce the delay, but it introduces artifacts similar to "clicking" sounds, which is also undesirable.

To reduce processing delay, live processors like Waves Multi Rack SGS1 Combo [18] can be used, which allow for zero-delay VST hosting. To minimize hardware delays, it is advisable to use the Dante data exchange protocol [19] in combination with mixing consoles such as Yamaha TF, Yamaha CL, Behringer x32 with Dante Card. The combination of Dante and Waves achieves a total delay of around 0,1 ms at a sampling rate of 44100 Hz. However, the drawback of this combination is its relatively high cost.

Conclusions

In this work examined the effectiveness of a MBC in addressing the task of correcting signals distorted by room reverberation. The results of both simulated and real experiments demonstrate that the use of a MBC is appropriate for correcting the frequency characteristics of the room. It has also been observed that the use of a VST host with a multi-channel VST compressor with zero-delay still introduces a delay (not less than 4 ms) in the processed signal. Future research will focus on research methods to minimize hardware and software delays, as well as optimizing the parameters of the MBC.

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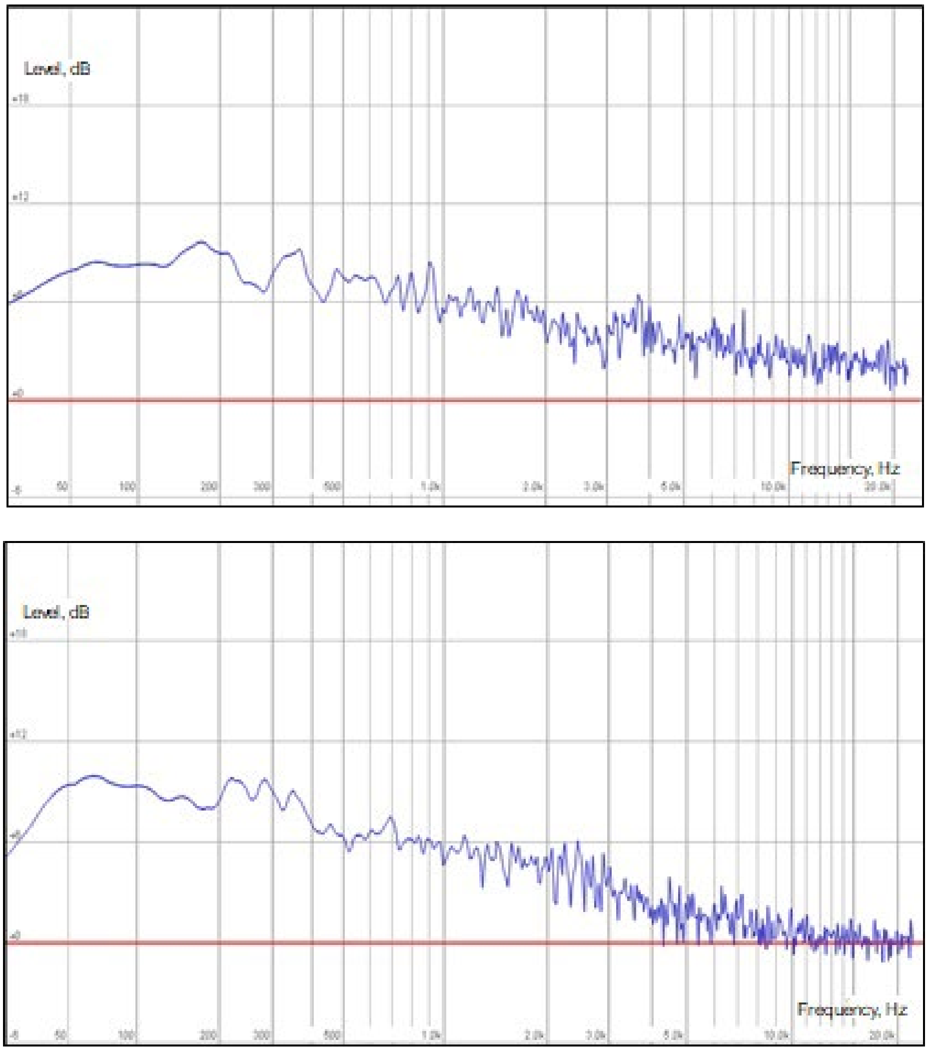

a)

b)

Fig. 8 Corrected spectra of pink noise: model (a) and real (b) experiments

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Корекція амплітудно-частотної характеристики акустичної системи у реальному часі

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Анотація – Корекція акустичної системи під час проведення концертного заходу є одним із найважливіших завдань забезпечення акустичного комфорту. Проте відомі нам підходи до такої корекції не дозволяють підлаштуватися в реальному часі під зміни акустичних властивостей приміщення, які відбуваються під час заходу. Для усунення цього недоліку, в даній роботі пропонується використовувати багатоканальний компресор, як альтернативу класичного еквалайзеру. В рамках даної робот перевіряється можливість коригування в реальному часі частотної характеристики системи «гучномовець – приміщення - мікрофон». Сигнал випромінюється гучномовцем, проходить через приміщення та сприймається парою мікрофонів штучної голови, розміщених у приміщенні. Електричний сигнал з виходу мікрофонів подається на додатковий вхід (вхід керування) багатоканального компресору. При перевищенні певного порогу рівнем керування сигналу в певному частотному каналі, сигнал, що проходить через компресор має послабитися. В роботі показано, що в ролі багатоканального компресору може бути використана зв'язка «комп'ютер – VST-хост - VST-плагін», слід зазначити, що використаний VST-плагін є плагіном з нульовою затримкою, має лінійну фазочастотну характеристику, дозволяє змінювати рівень сигналу у двох режимах, - стиснення (компресія) та розширення (експандія). В даній роботі обмежуємось лише режимом стиснення як таким, що частіше використовується у реальних задачах VST-хост вносить мінімальну затримку в обробку. При налаштуванні системи, в якості тестового сигналу можна використати рожевий шум. Обладнання використане для реалізації експерименту (окрім гучномовів) є типовим для проведення концертного заходу. Експериментальні дослідження багатоканального компресору виконувалися у два кроки. Першим кроком було моделювання спотворюючої дії приміщення на сигнал та перевірка алгоритму роботи багатоканального компресору. Другим кроком експерименту були натурні випробування у приміщенні Результати перевірки запропонованого алгоритму, виконаного шляхом проведення модельних та натурних експериментальних досліджень, свідчить про працездатність та ефективність запропонованого підходу. Такий підхід має недолік пов'язаний з обмеженнями апаратної частини. Для реалізації підходу під час проведення мовленевих заходів необхідно забезпечети високу якість обладнання, у свою чергу для прослуховування музики, або перегляду фільмів такий підхід може бути реалізованим на більш дешевій апаратній частині.

Ключові слова – багатоканальний компресор; еквалайзер; акустична система; штучна голова; VST-плагін; корекція сигналу.