Improved Remote Control System for Analog Audio Mixers Featuring Internet of Things Elements

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Abstract. Music is often found in people's lives, as a form of relaxation and inspiration. Small or medium concert venues use analog audio mixers to process the audio signals produced by each instrument and by the singer to deliver a pleasant sound to the audience. The problem is that the position of the mixer is, in most cases, on the stage or in a corner of the hall. This way the sound that the sound engineer hears will not be the same as what the audience hears. His place should be in public for a qualitative assessment of sound. Analog mixers cannot be controlled remotely, and the current alternative involves the replacement with a new and expensive digital audio mixer. The paper presents a cost-effective system that can be attached to any analog audio mixer, allowing to remote control main parameters like the attenuation of each signal. The remote control is a smartphone application, allowing easy further development and connectivity. The system was implemented, tested and the results and performances are presented in the paper, along with the details about the developed custom remote-to-mixer synchronous communication.

Keywords: Audio signal · Analog mixer · Synchronous communication · IoT

1 Introduction

At any musical events, the optimum place of the team adjusting the sound should be in front of the stage. This place is also called Front of House (FOH). Unfortunately, installing the mixer and the other audio equipment in front of the stage in small or medium venues is avoided because the equipment will take most of the place and the people would not be able to enjoy the artistic performance properly. Consequently, the sound engineer is usually placed in a corner of the hall or even on stage. The placement of the sound engineer will not allow him to correctly hear the sound that will be delivered to the audience. An improvement would be if the mixer and the other audio equipment could be placed where is convenient in the venue and the audio engineer could control the mixer from various places using his smartphone or tablet while making the necessary adjustment and monitoring the results from the audience perspective. In most of the small or medium venues an analog audio equipment is available.

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In the recent years, as the music technology began to evolve, the digital audio mixers began to feature more advanced capabilities than the analog audio mixers [1]. One important capability of the digital audio mixer is the remote control [2]. Comparing the costs between the analog and the digital version, the latter is much more expensive. Unfortunately, investing in a digital audio mixer is not always considered by venue owners because of the supplementary cost and the fact that they already own an analog one with can still organize concerts even if it cannot be remotely controlled and the audition quality will not be the best.

The system proposed in this paper uses the dedicated "Insert" connector available on most analog audio mixers to allow to remote control the levels of the audio signals using and Android application which can be installed on a smartphone or on a tablet [3]. Therefore, the sound engineer now has the possibility to set the important parameters from the desired position within the venue. The proposed equipment has several advantages: it is compatible with most analog audio mixers because total cost is affordable, allowing the sound engineer (and not the venue owner) to own it and use it in any location that he works in.

Several weighted audio mixing algorithms were proposed, some of which being able to increase the voice quality of the mixer output. But the voice quality cannot be maintained by these algorithms if the background has high noise levels, therefore leading to lower mean opinion scores. A new weighted audio mixing algorithm [4] was introduced, including enhancement algorithms such as noise reduction, automatic level control and voice activity detection. The weighted factor is calculated by the proposed algorithm based on the root mean square values of the input streams of the participants of the conference. Thus, the algorithm is able to adaptively smoothen the input streams and provide a scaled mixer output which is better in perceived speech quality. Perceptual Evaluation of Speech Quality (PESQ) and Perceived Audio Level (PLL) measures are used to compare the results of this new algorithm with earlier work in different background noise levels. Better and consistent speech quality was demonstrated by this new algorithm in all background noise levels.

Another research group [5] developed a standalone audio mixer either for wireless microphones or for working cooperatively with audio consoles. The universal serial bus (USB)-based Software Defined Radio hardware component employs an SDR to perform personal computer (PC)-based computations, and the free open-source software (OSS) GNU radio companion (GRC) is chosen to build the block-diagram of the SDR application. Using Software Defined Radio to perform computations, GNU radio companion to build block-diagram of the SDR and Open Source Software as Jitsi for the XMPP client in a PC, Empathy for a client in a laptop and Openfire for the XMPP server, the authors tested these three technologies and the hardware and software components in several scenarios and demonstrated the efficiency of the proposed solution.

In a previous paper written by our research group [6], a system was designed that allows the remote control to perform an attenuation of the audio signals entering an analog mixer using an Android application which can run on a phone or tablet. This system allows a critical upgrade to analog audio mixers having much lower costs than an upgrade to a digital mixer.

The paper is organized as follows: Sect. 2 comprises the system architecture, whereas Sect. 3 describes the mobile application designed for controlling the mixer. Section 4 was dedicated to the presentation of the technical characteristics of the system. In Sect. 5, the analog audio mixer connections and signals processing are described. Finally, the conclusions are drawn in Sect. 6.

2 System Architecture

The proposed system (Fig. 1) consists of two subsystems. The first subsystem is called the effects processor and contains a power block, an attenuation block, a command and communication control block, and the communication radio interface. The second subsystem is a remote controller implemented as a smartphone application allowing multiple communication means like Bluetooth or Wi-Fi [7].

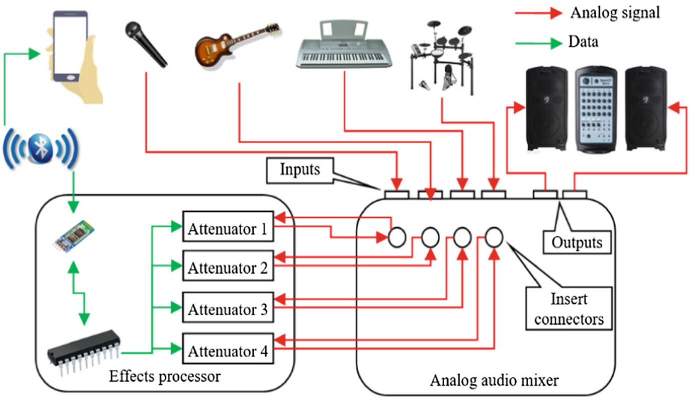

Fig. 1. The functional diagram of the proposed system

3 Mobile Application for Controlling the Mixer

The developed mobile application (Fig. 2) is described onwards. The sound engineer pairs his smartphone running the developed application with the effects processor by pressing the “Connect” button, selecting the Bluetooth transceiver of the effects processor from the list and entering the password. Each effects processor has a unique password, so other users cannot connect to it. In this way, the security of the system is increased.

Next, the channel to be controlled can be selected by pressing on the button illustrating the channel’s number (from 1 to 4 in this implementation). After this, the user can drag the displayed fader like it would use a real fader on the analog mixer, to set the desired attenuation on the selected channel. At each movement of the slider, the selected channel and the current attenuation is sent to the Bluetooth transceiver which is part of the effects processor.

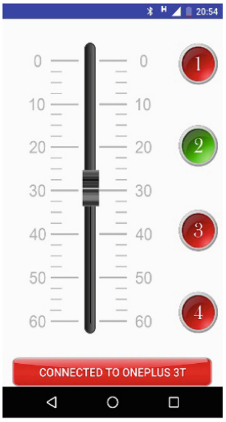

Fig. 2. The developed smartphone application.

The Bluetooth communication can be considered a serial communication tunnel, sending ASCII characters. This communication contains synchronization symbols, very important because it allows correct interpreting of the code word even if some characters are lost, allowing robust resynchronization with the data flow. The structure of the code word is "*CCAAA#" where "*" and "#" are the start and end of code word markers, "CC" is a two-digit number representing the selected channel (e.g., 00 would mean the first channel, 01 the second, 02 the third etc.), and "AAA" is the set value of the attenuation, expressed in dB. When a channel is selected, the last set value for the attenuation on that channel is automatically set on the slider, so no accidentally changes in the attenuation will occur. Figure 1 shows the interface of the developed application and Fig. 2 illustrates the functional diagram of the proposed system in the context of a live performance situation.

4 Technical Characteristics of the Control System

The system was designed keeping portability as a priority. The primary power supply is a 5 V battery power bank, having great capacities nowadays and being compact.

The attenuators need a larger supply voltage to accommodate the whole dynamic range that is typical for signals passing through an analog audio mixer. To obtain the needed 9 V, a step-up miniature switching power supply is used-not described in detail as it is not the main subject of this paper. Also, the microcontroller and the Bluetooth transceiver require a 3.3 V supply voltage, obtained using a LM317 regulator [8].

The voltage obtained using the switching power supply should be thoroughly filtered since it powers the attenuators which process the audio signal, and noise contamination should be avoided. It can be observed in Fig. 3 that each LM1971 attenuator [9] and the operational amplifiers [10] which participate in the audio signal processing are accompanied by a pair of power supply filtering capacitors.

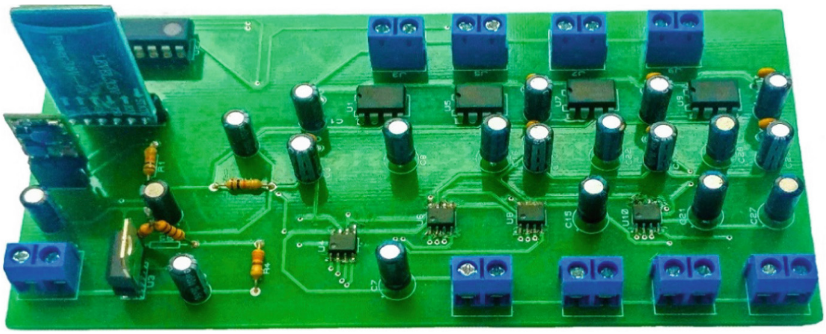

Fig. 3. The developed remote-controlled effects processor featuring four audio channels.

Since the analog audio signal processing blocks in the developed effects processor are powered by a single supply source, and not using a symmetrical split power supply that could deliver positive and negative supply voltage, DC coupling capacitors must be used because audio signals have no DC components.

The capacitance was chosen to be 1μF. In this way, the lower cutoff frequency of the system is smaller than 20 Hz, outside the audio bandwidth. For correctly biasing the input and output, the attenuator circuit needs a voltage reference that is provided using a resistive divider made by R1 and R2 resistors, as shown in Fig. 3. The reference voltage is filtered using the capacitor labeled with C4 and furthermore the equivalent reference voltage source made using R1, R2, and C1 is improved by using an operational amplifier-based buffer, to decrease its output impedance.

A buffer is also used at the output of the attenuators to assure a large load impedance for the attenuator, no matter the load connected at the output of the system.

The data connection between the microcontroller and the attenuators is similar to the standard SPI communication and uses 3 lines: LOAD, DATA and CLOCK. The LOAD line of the destination chip is kept low during the communication. The DATA line will contain the desired value of the attenuation (in dB) delivered bit by bit, one bit in each clock period. The clock signal is delivered on the CLOCK line [9].

The commands to be sent to the attenuators are received by the microcontroller from the Bluetooth transceiver through serial communication. The decoding process of a code word is started when the "*" character is received. Then five more characters are received, and then it is tested that the next character is "#". If it is, then the selected channel and the value of the attenuation are extracted from the code and sent to the corresponding attenuator as it was described above. It must be observed that the attenuation value is received as a decimal number and must be sent to the attenuators as a binary value, so a decimal to binary conversion was implemented in the software code running on the microcontroller.

The whole code word can be treated like an integer number, processed as described further. Because C programming language was used, the channel can be identified by dividing the number by 1000 and the attenuation value would be the remainder after division. The initialization of the attenuators should be done when the system is powered up by setting the attenuation value to 0 dB (no attenuation).

The proposed system involves the use of a smartphone and this represents an open door to accessing many Internet of Things features and advantages [11]. For example, the smartphone is equipped with multiple communications means, like Wi-Fi, Bluetooth, and communications over a mobile network (in the form of data or voice) and, in this application, could be considered an IoT node. With these features, the system could be easily controlled from the Internet, simplifying the work of the sound engineer.

The proposed effects processor was implemented and tested. It is illustrated in Fig. 3. The developed hardware module can process four audio channels and it is a fully working prototype. The design could be improved by making the solution scalable with easy adding or removing audio channel modules and by redesigning the layout to occupy less space. The use of SMD components could also bring space and cost improvements.

5 Analog Audio Mixer Connections and Signals Processing

The musical instruments and the microphones are connected to the analog audio mixer, each on a separate channel, as shown in Fig. 2. Each channel on almost any analog audio mixer is featured with a connector named "Insert". This connector allows the connection of external audio signal processors. The connections are made so the external processors are cascaded in a way that allows the audio signal from a channel to be sent to the external effects processors and received back after it was processed using a single TRS (Tip-Ring-Sleeve) jack connector which is very convenient. Thanks to this connectivity type, the developed system is compatible with any analog audio mixer, but also with any other audio processor designed to work in this configuration.

After being processed using the external audio processors, the signals are summed and then sent to the main output of the mixer which is connected to the speakers, delivering the sound to the audience. A very important element of this proposes system is that the command terminal is a smartphone. This allows the commands to be sent not only from operating the developed smartphone application, but also using SMS or the data connection, behaving like an IoT node, greatly increasing the flexibility of the system. The circuit diagram of the proposed remote-controlled signal processor is shown in Fig. 4.

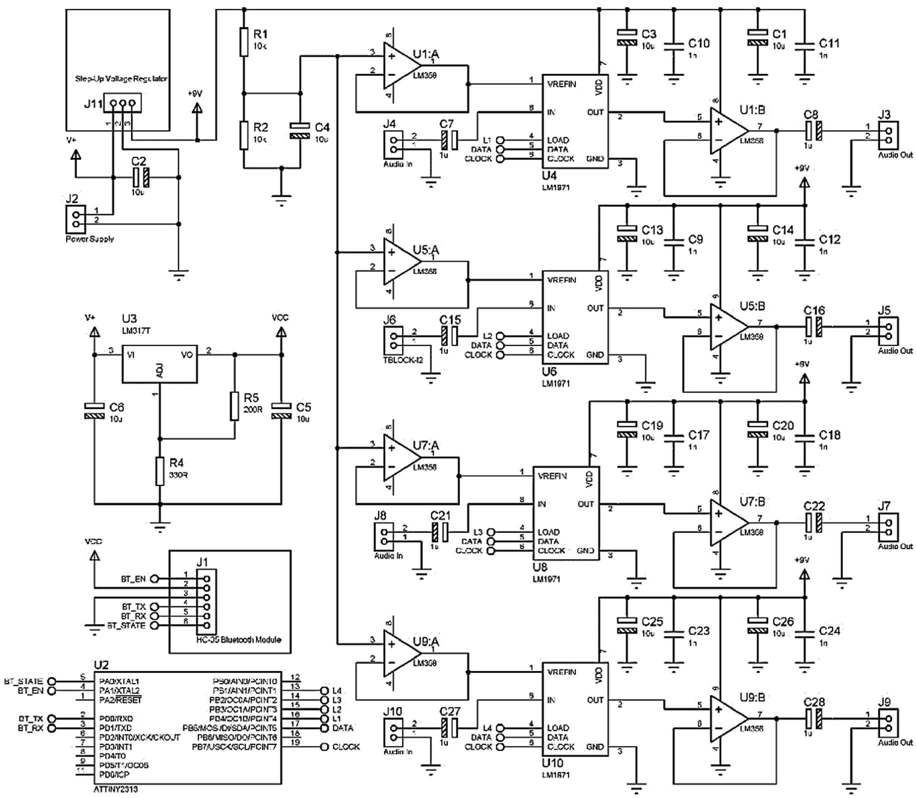

Fig. 4. The schematic diagram of the proposed remote-controlled signal processor.

6 Conclusion

The paper has proposed a system that allows the remote control of the level of the signals in an analogue audio mixer. The system consists of two subsystems: a remote controller implemented as a smartphone application which transmits selected channel and current desired value for the attenuation, and an effects processor that receives this information, decodes it and sends the attenuation information to the correct attenuator. The system allows communication via Bluetooth or Wi-Fi.

In this implementation, a Bluetooth communication between the smartphone and the effects processor was chosen. This is a serial communication tunnel. The transmitted code contains synchronization symbols, which are useful for correctly interpreting code words, even if some characters are lost during the transmission.

The proposed system has several advantages: it is secured with passwords to prevent access by other unknown users, it is compatible with most analog audio mixers because it can be inserted in the signal's path using the "Insert" connector that is available on the majority of the analog audio mixers, it is cost effective (especially compared to digital audio mixers which have this feature), and it is flexible due highlighted IoT capabilities.

The system was tested on a Soundcraft Spirit Rac Pac analog audio mixer and its correct functioning was confirmed. The range of the Bluetooth communication of around 10 m is enough for the targeted application, given the dimensions of small and medium concert venues. The system could be further improved by considering a scalable solution, because the audio channels on an analog audio mixer varies greatly from model to model, improving also the economical aspect of the solution.

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