Low Cost Embedded Chairman/Delegate Units Design for Digital Conference System


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Abstract—For enhancing our existing conference systems product of embedded chairman/delegate units, a low cost embedded chairman/delegate units design is proposed in this paper, which is capable of digital audio processing for digital conference system. In addition, the functionality of an automatic gain control (AGC), a dynamic range compression (DRC), and a 2.5W embedded class-D speaker amplifier is also supported in this design.

Keywords—chairman unit; digital conference system; delegate unit; digital audio codec; digital signal processing (DSP)

I. INTRODUCTION

Recently, the related techniques of digital audio signal processing are widely studied, designed, and adopted in the professional conference systems, broadcasting systems, and public address (PA) systems [1]-[9]. These techniques overcome the disadvantage of traditional analog signal audio processing systems such as lower signal noise rate (SNR), hard to maintain and management, and lower quality of audio.

Compared to traditional analogue system, the digital system has many advantages, which can be ease of maintenance and debug, be easily expanded and modified, be higher quality of audio, and so on. Moreover, for the digital audio conference systems, the CAT.5 twisted pair or fiber optic cable can take instead of analog audio cable so that the cost would be reduced.

The related techniques of digital audio signal processing are adopted for developing embedded chairman/delegate units in our developing digital audio conference system as can be seen in Figure 1 in this paper. Moreover, the digital audio conference system also should support bidirectional digital control between chairman/delegate units and central control units (CCUs).

As a result, high quality audio is one of the key factors in a successfully conference. Therefore, this paper proposes a low-cost high-quality digital audio embedded chairman/delegate units design for the professional digital conference systems.

II. FRAMEWORK OF DIGITAL CONFERENCE SYSTEM

A. System Architecture

One of our developed digital conference systems [10] consists of central control units (CCUs), embedded chairman units (ECUs), embedded delegate units (EDUs), a monitor control keyboard (MCK), high-speed-dome type cameras, a multi-channels multiplexer, and TVs/projectors, is shown in Figure 1.

Figure 1. An example of the digital conference system.
speaker is finished speech according to the maximum capacity of speeches.

3) Override Mode: In this mode, the next speaker can override previous first speaker who exceeds the maximum capacity of speeches.

4) Free Discussion Mode (Normal Mode): All of the participants in the conference can freely discuss each other in this mode.

As a result, the modes above mentioned can be defined by our developed graphical control software.

III. EMBEDDED CHAIRMAN/DELEGATE UNIT DESIGN

An embedded chairman/delegate unit consists of an audio ADC, an audio codec, an automatic gain control (AGC), a dynamic range compression (DRC), an audio DAC, a 2.5W embedded class-D amplifier, a power management module (PMM), and microcontroller.

The system architecture of the proposed embedded chairman/delegate unit is shown in Figure 2. The inputs are from self-chairman/delegate-unit microphone input and CCU as a different reference.

These inputs can be suitably gained by using a microphone programmable gain amplifier (PGA) for ADC. The AGC is used to maintain nominal constant output signal amplitude for continuously entering speech or audio signals. The audio codec processes digital audio signal.

The DRC is adopted for monitoring the output of DAC digital volume control to detect its power level relative to 0 dBFS. The proposed embedded chairman/delegate unit will be introduced in details in this section below.

A. Audio ADC

The audio ADC consists of a microphone PGA, a delta-sigma modulator, and a digital decimation filter including a linear-phase infinite-impulse-response (IIR) filtering. The audio ADC data sample rate is set at 44.1 KHz for speech application.

The microphone PGA can support analog gain control from 0dB to 59.5dB in steps of 0.5dB. The gain level can be regulated by using a microcontroller and our developed graphical control software. In the other hand, user can adjust gain level for the requirement of microphone sensitivity. Moreover, the microphone PGA is also controlled by the AGC loop.

The digital decimation filter is used for processing the oversampled data from the delta-sigma modulator to produce the digital audio data to audio codec. Moreover, the audio data is transferred between audio ADC, audio codec, and audio DAC.

B. Automatic Gain Control (AGC)

AGC is an adaptive system, which can be found in many electronics applications. The mean output signal level is fed back to adjust the gain to an appropriate level for a range of input signal levels [11].

The AGC is adopted for our microphone inputs. The AGC is used to maintain nominal constant output signal amplitude for continuously entering speech or audio signals. The AGC can automatically adjust the microphone PGA gain, as the input signal is overly loud or very weak, given an example, when a person speaking into a microphone takes closer to or farther from the microphone.

For the AGC-based control, some settings [12] would be determined: 1) target level, 2) AGC low-pass filter, 3) attack time, 4) decay time, and 5) noise threshold. Also, the AGC operation is demonstrated in Figure 3.

1) Target Level: means output signal, which the AGC tries to keep the ADC output signal.

2) AGC Low-Pass Filter: is adopted for assisting to define the mean level of the input signal.

3) Attack Time: represents how quickly the AGC to reduce the PGA when the input signal is too loud.

4) Decay Time: represents how quickly the PGA is increased when the input signal is too low.
5) **Noise Threshold:** as a reference level, the AGC would be concerned it as a silence when the input audio mean value falls following the noise threshold.

C. **Audio Codec**

The main functionality of the audio codec is to provide clock generation and phase-locked loop (PLL) for the audio ADC and audio DAC as well as interface. Those clocks for the audio ADC and audio DAC require a source reference clock. For enhancing digital audio processing, the audio codec also provides some digital filters and some digital audio effects such as equalization (EQ), 3D, etc.

D. **Dynamic Range Compression (DRC)**

Typically, audio signals can be characterized by crest factors, thus the ratio of peak signal power to mean signal power of 12dB or more. The audible distortions have be avoided due to the clipping of peak audio signals, thus the gain of DAC have to be adjusted so as not to cause hard clipping of peak signals [12]. Consequently, the applied gain is low during the nominal periods, which could cause the perception that the signal is not loud enough [12].

The DRC can be solved such problem [12]-[14]. Hence, in this work, the DRC is adopted for monitoring the output of DAC digital volume control to detect its power level relative to 0 dBFS. The DRC will be increased the input signal gain to do sound louder when the power is lower. The DRC can also be autonomously reduced the applied gain to avoid clipping when a peaking signal is detected at the same time.

For the DRC functionality, the parameters of DRC need to determine as follows [12], [13]. An example of the DRC parameters settings by using TI control software is also shown in Figure 4.

1) **DRC Threshold:** determines a level value of the DAC playback signal, which the gain compression is driven.

2) **DRC Hysteresis:** avoids the rapid activation and deactivation of gain compression in the DRC operation. The current value is set to 3dB.

3) **DRC Hold Time:** is intended to slow the start of decay for a specified period in response to a decrease in energy level.

4) **DRC Attack Rate:** exceeds the programmed DRC threshold when the output of the DAC digital volume control, the gain applied in the DAC digital volume control is progressively reduced.

5) **DRC Decay Rate:** detects a reduction in output signal swing beyond the programmed DRC threshold, the DRC enters a decay state, where the applied gain in the digital-volume control is gradually increased to programmed values.

E. **Audio DAC**

The audio DAC consists of digital interpolation filter, multi-bit digital delta-sigma modulator, and an analog reconstruction filter. This audio DAC is to provide enhanced performance via increased oversampling and image filtering at the low sample rate.

The digital interpolation filter produces oversampled data for the multi-bit digital delta-sigma modulator. The audio outputs include a class-D speaker, a lineout, and a headphone. This audio DAC data sample rate is set at 44.1 KHz for speech application.

F. **Embedded Class-D Amplifier**

The class-D amplifier is widely adopted in consumer electronics, especially for continuing improvements in power semiconductors [16]. The advantages of class-D amplifier are listed as follows [17].

- The size and weight of an amplifier are reduced.
- Power waste is reduced as heat dissipation and hence smaller (or no) heat sinks.
- The cost is reduced due to smaller heat sink and compact circuitry.
Very high power conversion efficiency, usually better than 90% above one quarter of the amplifier's maximum power, and around 50% at low power levels.

Following the benefits of class-D amplifier mentioned above, a 2.5W embedded class-D speaker amplifier is adopted to the our design for the purpose of low cost and low power.

G. Power Management

A basic power management is implemented. The chairman/delegate unit detects input signal to judge the microphone is in use, if no, then turn-off the audio codec, audio ADC, and audio DAC for saving power consumption. Moreover, the turn-off devices information will be sent to graphical control software for managing chairman/delegate units.

H. Microcontroller

A PIC-like microcontroller is adopted to control the proposed chairman/delegate unit via I 2 C protocol. Moreover, CCU and graphical control software are communicated with the microcontroller of each chairman/delegate unit via CAT.5e with RJ-45 interface to transmit commands and audio signals to each chairman/delegate unit or CCU.

IV. IMPLEMENTATION RESULT

This design is implemented by using a TI TVL320AIC series audio IC [12] and a PIC-like microcontroller. Figure 5 shows prototyping board for this work. Two RJ-45 interfaces are given for communicating the CCU or other chairman/delegate unit as can be seen in Figure 2 and 5. The design of chairman/delegate unit can be successfully integrated with existing conference system product as shown in Figure 1.

Table I summaries the proposed low cost embedded chairman/delegate unit design. Audio inputs provide maximum 3 audio inputs simultaneously. One 2.5W class-D speaker and 2-channels audio lineouts are provided for audio outputs.

<table>
<thead>
<tr>
<th>Item</th>
<th>Description</th>
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<tbody>
<tr>
<td>Audio Inputs</td>
<td>Maximum 3 Audio Inputs Simultaneously.</td>
</tr>
<tr>
<td>Audio Outputs</td>
<td>One 2.5W Class-D Speaker; 2-channels Audio Lineouts.</td>
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<tr>
<td>Interfaces</td>
<td>RJ-45 Interfaces.</td>
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<td>Connection</td>
<td>CAT.5e Cable</td>
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<td>DSP</td>
<td>Including a Simplest DSP Processor.</td>
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<tr>
<td>Resolution</td>
<td>16-bits Audio ADC.</td>
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<tr>
<td>Sample Rate</td>
<td>Audio ADC: 44.1 KHz; Audio DAC: 44.1KHz</td>
</tr>
<tr>
<td>Special Features</td>
<td>Integrated with AGC and DRC.</td>
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</table>

Table I. SUMMARY OF CHAIRMAN/DELEGATE UNIT DESIGN

The digital audio signal is processed by a simplest DSP processor. The RJ-45 is adopted as an interface. The audio ADC is 16-bits resolution for speech application. The sample rate of audio ADC and audio DAC is 44.1 KHz, which achieve the requirement of speech application.

Fully switching power is adopted for low power design so that it can increase the maximum connected number of chairman/delegate units. Power management can turn off chairman/delegate units, which are not in use. The special features of the proposed design which are integrated with AGC and DRC that it can provide more stable audio quality in progress of conference.

V. CONCLUSION

In this paper, we have proposed a low cost design of embedded chairman/delegate units for digital conference systems. The design has high quality low cost digital audio processing for digital conference system. Moreover, the functionality of an automatic gain control (AGC), a dynamic range compression (DRC), and a 2.5W embedded class-D speaker amplifier was included in this design.

This design can be successfully integrated with existing conference system product. For further future work, dynamic power management [18] will be concerned, analyzed, and designed for low-power systems design.

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