Communication Design of Intercom System's Digital Audio

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Abstract

the article will introduce a high quality audio transmission and decoding circuit, the system is divided into main two parts, they are audio signal encoding module and audio signal decoding module, the audio encoding part is composed of A/D converter CS5340 and digital audio transmitter CS8406. The audio decoding part is composed of digital audio receiver CS8416 and audio D/A converter CS4344. The CODEC supports AES/EBU and S/PDIF audio data that is 24bits and 192KHz sampling rate, the hardware circuit is sample, reliable and flexible. Experimental proof that the designed system can improve the signal's transmission quality, and lengthen the distance of transmission.

Keywords

long-distance transmission, audio signal encoding and decoding, AES/EBU, S/PDIF.

1. Introduction

There are two main modes for audio signal transmission: analog signal transmission, digital signal transmission. when analog signal is used for long-distance transmission, due to environmental noise and power loss, there are always distortion and attenuation, so during the transmission, the analog signal power need to be amplified by repeaters [1], but at the same time the signal is amplified the noise and distortion is amplified too. Once the transmission distance is lengthened, the cascading repeaters will overlay the distortion, and it will make the wave distortion even bigger. The transmission of the digital signal, the data only represents the change of electrical level between "0" and "1", the digital pulse signal during transmission will not only be attenuated but also be distorted, but the amplifiers can be replaced by repeaters [2]. The repeaters can recover the mode that same at the original signal changes between "1" and "0", it regenerates and transfers a new digital signal that without distortion and attenuation, so that makes cascading construction doesn't overlay noise to cause waveform distortion.

2. System Function Design

There are two main parts of the system: audio encoding part is showed as figure 1. Audio decoding part is showed as figure 2.

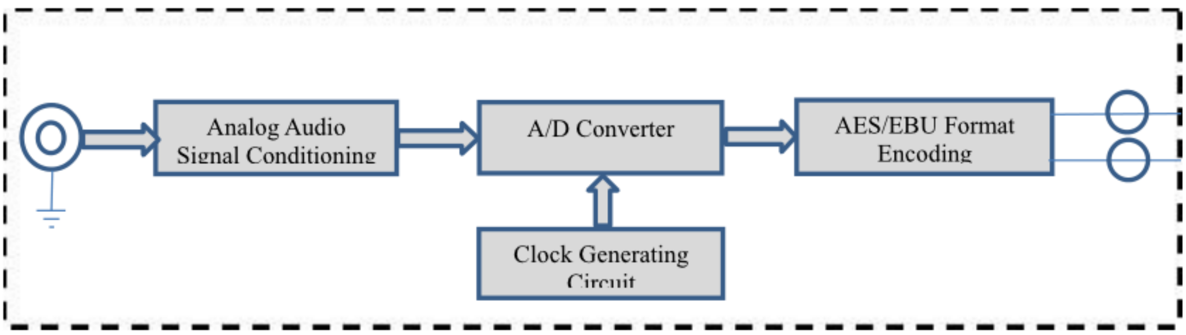


Figure 1 encoding circuit

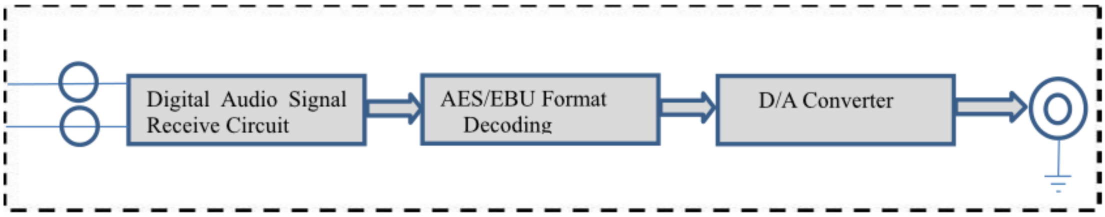


Figure 2 decoding circuit

The encoder and transmitter modules are processing the analog audio signal mainly, Converting by the D/A converter and then the I2S serial digital bus will transfer that to the digital audio encoder, the encoder will make the data flow AES/EBU or S/PDIF encoding and transmitting [3].

The decoder and receiver modules decode the AES/EBU or S/PDIF data flow, at the same time, it will generate a clean clock signal[4]. The decoded audio data will be transmitted to the stereo D/A converter, the high quality analog signal that filtered by start up & shutdown squelch circuit will be output by the RCA interface at last.

The full name of the AES/EBU is Audio Engineering Society/European Broadcast Union, now it has become a popular standard in the professional digital audio processing. It's a digital audio format without compressing. Though a serial bit transmitting protocol that based on a single bunch wire (24 bit quantification at maximum) . It can transmit data in the distance of 100m without equilibrium, but if it has equilibrium, the distance of transmission will be lengthened [5].

The full name of S/PDIF is Sony/Philips Digital Interconnect Format, it's a kind of civilian digital audio interface protocol that developed by the cooperation of Sony and Philips. Because of its widely usage, in fact it has already been the civilian digital audio format standard, There is a little difference between the structure of S/PDIF format and AES/EBU format. But the audio information get the same position in the data flow, make the two formats compatible in principle.

3. Circuit Design

Encoder Circuit Design. Before convert the analog audio signal with A/D converter, the signal has to be processed by the conditioning circuit first, he core device of the conditioning circuit is a high performance low noise dual-OP which the model is NE5532 [6], to process the alternating audio signal, by using R3,R8 and R20,R24 to divide the voltage, make the audio signal input range up to +1.67V.In Figure 3, C16 , C30 are the blocking condensers to decouple the voltage; R5, R10, C18 and R22, R26, C31 compose a low-pass filter with OP, the cutoff frequency of the filter is 500KHz, make the high-frequency component filtered in analog signal. Otherwise, it also plays a role of buffer when filtering the signal, reduces the resistance of the audio signal resource and improves the driving capacity.

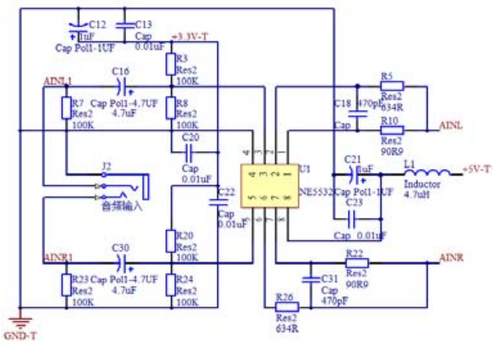


Figure 3 conditioning circuit

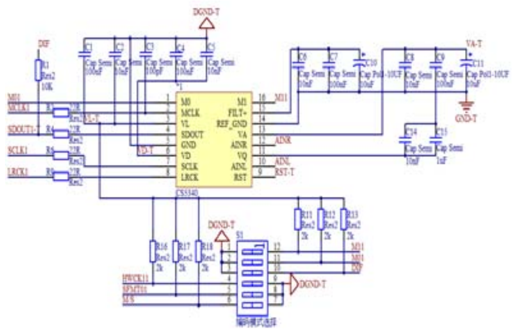
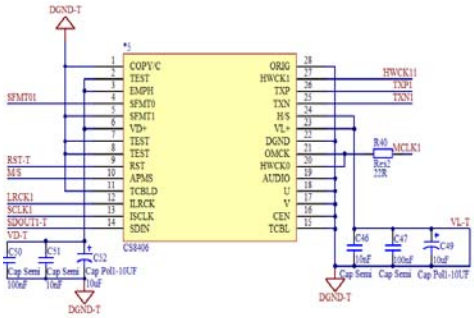


Figure 4 A/D converter

CS5340 is the digital audio system's A/D converter. It executes sampling , A/D converting ,filter of anti-aliasing , generating 24 bits serial data of left channel and right channel, the sampling rate of each channel can reach up to 200KHz. CS5340 adopts a 5th order multidigit Δ-Σ modulator, and then digital filtering and sampling. Thus, it doesn't need any external filter of anti-aliasing circuits. As for CS5340, as shown in Figure 4,the sampling rate is determined by Pin M0(Pin 1) and M1(Pin 16),for further transmission ,we use a lower sampling rate in the design, Pin M0(Pin 1) and Pin M1(Pin 16) are both connected to ground in the design, making the sampling rate to be the 512 frequency demultiplication of 16.384MHz that is 32KHz.Due to the Pin SDOUT of CS5340 is pulled up by a 10k resistor to VL, so the format of serial audio data transmission is I2S.

CS8406 is a single chip encoding CMOS device, According to AES3, IEC60958, S/PDIF or EIAJ CP1201 standard to transfer audio data. CS8406 receives audio and digital data, then reuse, encode, drive to a single bunch of wire. The audio encoder circuit is shown as Figure 5. Sampling rate of CS8406 and audio receive format have to match with CS5340, so the Pin HWCK0 (Pin 20) and Pin HWCK1(Pin 27) connect to ground and high level respectively. To guarantee the sampling rate is exactly 32KHz; Pin SFMT0(Pin 4) and Pin SFMT1(Pin 5) connect to high level and ground respectively. To guarantee the audio receive format is I2S. According to protocol AES3, the data flow output rate is 64*32KHz=2.048Mb/s.



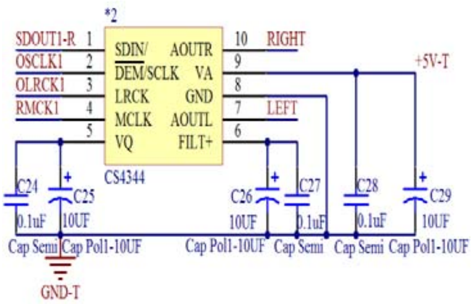


Figure 5 Encoder Circuit Figure 6 A/D Converter

CS8416 is a 8-channel stereo CMOS device that receive and decode the digital audio data , according to IEC60958 , S/PDIF , EIAJ CP1201 or AES3 interface standard to receive audio data. Audio decoder circuit is shown as Figure 7. CS8416 works under hardware mode, so SDOUT is pulled down to ground by a 47K resistor. To match with CS4344, the AUDIO Pin(Pin 15) , C Pin(Pin 19) of CS8416 are configured to low level and high level respectively, at this time ,the chip is working under the mode of 24bits I2S control, the digital signal that CS8416 receives will transfer to CS4344 in real time by I2S bus. Thus it recovers the analog audio signal from the left and right channel. Moreover, Because of the sampling rate of the system is 32KHz[7], to improve the quality of the signal that the Pin U(Pin 18)of CS8416 is connected to the ground. Regenerate the clock in order to 256 times of Fs, the frequency of recover main clock now is 256*Fs.

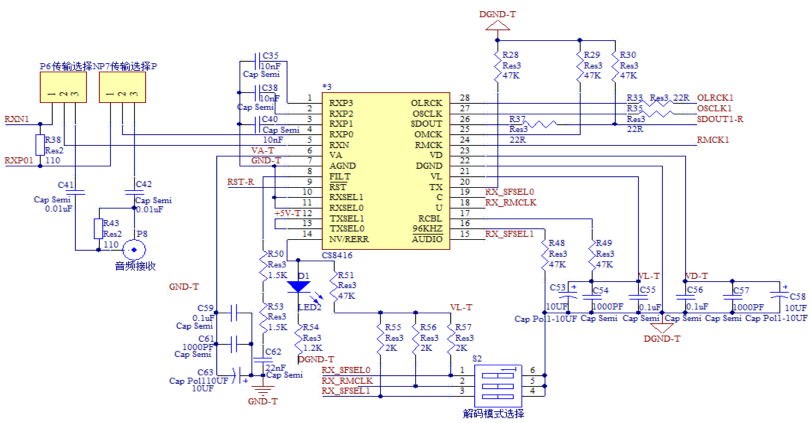


Figure 7 CS8416 interface circuit

C4344 stereo D/A converter, including run in multidigit D/A convert and output analog filter. CS4344/5/6/8 supports all kinds of audio data interface format, the only distinction of the single device is interface format, shown as Figure 6.

4. System Test

Using two devices that shown as Figure 6 to test the transmission distance, during the test we use two terminals to connect as transmitter and receiver. The result of the experiment is: without electrical level transition, transmit with coaxial line, transmission distance is longer than 5m;using 1mmPOF optical fiber for transmission, the distance is longer than 50m;when RS485 electrical level is transmitted by twisted-pair, the distance is shorter than 40m. So no matter adopting optical fiber or twisted-pair can both satisfy the need of long distance transmission of audio signal [7] on multi-load optical measuring vehicle. Otherwise the sampling rate of audio signal in the system is 32KHz, it can recover the voice of the talkers, meet the need of intercom system at the same time.

5. Conclusion

Aiming at the malpractice of the long distance analog signal transmission, we developed the digital audio transmission system. The system adopts AES3 CODEC technology, coaxial line, blocking twisted-pair, optical fiber or something else is applicative to transmit the digital audio signal, and applicate to multi-load optical measuring vehicle platform, improve the transmission quality and distance of the intercom system. Experimental proof that: the system features the hardware is simple and reliable, transmission quality is great and the application is flexible.

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