Design of a New Signal-generator Based on ASR-ONE

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Abstract: In recent years, with the continuous progress of science and technology and the expansion of application fields, the signal generator, as an important tool in the field of electronic testing and measurement, has played a key role in laboratory, communication system testing and radio spectrum analysis. In order to meet the increasing application requirements and technical challenges, this paper proposes a new design scheme of signal generator, which adopts the artificial intelligence chip ASR-ONE for speech processing, controls the DDS module output and displays the corresponding waveform parameters on the OLED screen. This paper aims to introduce the design principle and key technology of the new generation signal generator based on ASR-ONE. By combining the expertise in the field of artificial intelligence and signal generator, it is committed to providing a more flexible, intelligent and powerful signal generator to meet the needs of a variety of complex application scenarios.

Keywords: ASR-ONE; DDS; Speech Recognition; Artificial Intelligence

1. Introduction

With the rapid development of artificial intelligence and speech technology, voice signal processing is playing an increasingly important role in various fields. As one of the key devices, the design and function of speech signal generator have a profound impact on the performance and effect of speech recognition, speech synthesis and other applications. With the advantages of ASR-ONE, this design combines it with the signal generator to design a signal generator that can automatically generate the corresponding waveform according to the input voice commands. Through the signal generator, the user can flexibly control the frequency, amplitude, waveform and other parameters of the signal, to achieve a more accurate and diversified voice signal generation.[1]

2. System Implementation Principle

The whole system is divided into three modules, namely: voice control module, signal generation module and display module. The overall design structure of the system is shown in Figure 1.

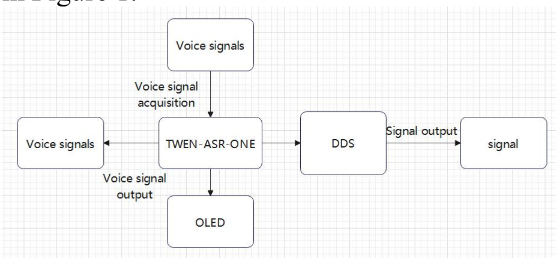

Figure 1. System Structure Diagram

In the figure, the voice control module is the core module of the whole design, which is implemented by the TWEN-ASR-ONE development board. The TWEN-ASR-ONE development board controls the OLED screen through the IIC communication protocol. Signal generation is controlled by the development board AD9910 high-speed DDS module to realize the role of signal output, the whole system by TWEN-ASR speech module to collect external voice signals for data conversion and processing, and use acoustic model and language model to identify voice commands, and control the AD9910 high-speed DDS module output corresponding parameters, while the output waveform parameters displayed on the OLED screen, these modules will be described in detail below.[2]

2.1 Voice Control Module

The voice control module is the TWEN ASR ONE V1.0 development board. The development board is equipped with the ASR-ONE chip, an artificial intelligence chip dedicated to voice processing. At the same time, the development board supports offline control function, with external horn and microphone, can not only collect voice signals, but also output voice signals.[3] The development board has a brain neural network processor BNPU, which supports local speech recognition within 200 command words. And TWEN ASR ONE V1.0 development board also integrated the built-in CPU core and high performance low power Audio Codec module, while using the port reuse technology, IIC, SPI, GPIO, UART and other peripheral control interface on the development board, and introduce the port, according to make it has to carry out a variety of cost-effective single chip intelligent voice product development scheme. The functional structure diagram of the development board is shown in Figure 2.

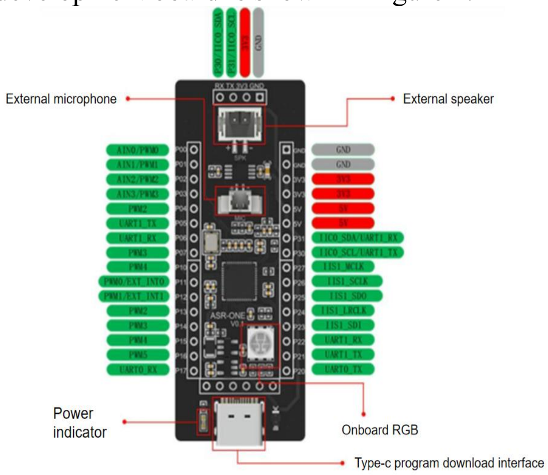

Figure 2. Functional Structure Diagram of TWEN-ASR-ONE

This design uses the ASR-ONE chip as the main control core of the design, The overall working principle is: when the microphone collects the analog signal of the voice command, From the analog signal to the digital signal into the BNPU neural network processor, Acoustic models and speech models have been preloaded in the neural networks, Will probabilistic match the sound information and text information with the model that has been learned, Combining text and sound information to identify external voice commands, When the recognized voice commands are already present in the information table, Give the speech ID the corresponding ID value,[4] The corresponding threads are performed simultaneously. The schematic diagram of speech recognition is shown in Figure 3.

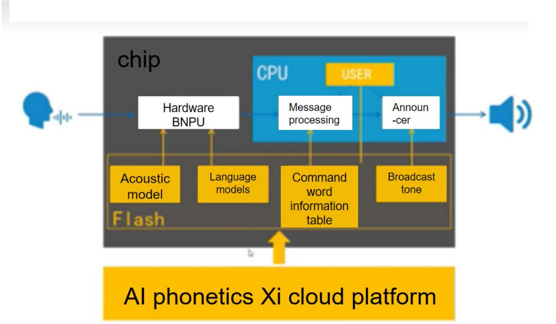

Figure 3. A Schematic Diagram of Speech Recognition based on Artificial Intelligence

When the waveform output is recognized, the thread Change is executed, and the Switch statement is executed according to the recognized speech ID to modify the different waveform parameters. When the change waveform is recognized, the speech ID is 10,11 when the change ID is 11, and the voice ID is 12. When the alignment ID is 10, the output waveform of the AD9910 high-speed DDS module is modified, and the OLED waveform type display is also modified. When the alignment ID is 11, the output peak value of the AD9910 high-speed DDS module is modified, and the peak peak size displayed by OLED is also modified. When the alignment ID is 12, the output frequency of the AD9910 high-speed DDS module is modified, and the frequency size of the OLED display will also be modified. As shown in Figure 4.

```
void Change(void){
  while (1) {
    delay(1);
    switch (snid) {//Speech recognition ID
     case 10:
      Wave = Temp;  //WAVE
      AD9910_Wave(Wave);
      OLED_Show_Wave(Wave);
      break;
     case 11:
      VPP = Temp;   //Vpp
      AD9910_AmpWrite(Vpp*16383/800);
      OLED_Show_Vpp(Vpp);
      break;
     case 12:
      Freq = Temp;  //frequency
      AD9910_FreWrite(Freq);
      OLED_Show_Freq(Freq);
      break;
     default:Temp =Temp;
      break;
    }
  }
  vTaskDelete(NULL);
}
```

Figure 4. Code program 1

For the storage recognition and storage part of the value, when the alignment ID is not 20 or 21, the alignment of the voice ID will be performed. When the alignment voice ID is 0~9, the value of the voice ID is stored in the Data array, only when the voice ID is 20 or 21, that is, when the hertz or volt is recognized.[5] When the alignment speech ID is 15 or 16, it indicates that the speech recognition is to the unit thousand or trillion, that is, assigning a value of K or M to 1 indicates the recognition to this unit. When the alignment speech ID is 22,23,24, it means that the speech recognition to sine wave, triangular wave, square wave, the data will be directly stored in Temp, jump out of the function. When the alignment ID is 20 or 21, the Data array and K and M units are integrated, and the value of Temp is the numerical part of the identified parameter. The specific code procedure is shown in Figure 5.

```
void Data_NUM()
{
  int i = 0;
  int K=0;
  int M=0;
  int Data[10] ={0};
  while(snid != 20||snid != 21){//hertz, volt
    switch (snid) {
     case 0:
    Data[i]=0;
      break;
     case 1:
     Data[i] = 1;
      break;
     case 2:
     Data[i] = 2;
      break;
     case 3:
     Data[i] = 3;
      break;
     case 4:
     Data[i] = 4;
      break;
     case 5:
     Data[i] = 5;
      break;
     case 6:
     Data[i] = 6;
      break;
     case 7:
     Data[i] = 7;
      break;
     case 8:
     Data[i] = 8;
      break;
     case 9:
     Data[i] = 9;
      break;
     case 15://thousand
     K = 1;
      break;
     case 16:
     M = 1;//omen
      break;
      case 22:
     Temp = 22;//sine wave
     return;
      break;
     case 23:
     Temp = 23;//Triangle waves
     return;
      break;
     case 24:
     Temp = 24;//square wave
     return;
      break;
     default:num = 0;
      break;
    }
       i++;
  }
  i--;
  Temp = GetValue(Data,i,K,M);

  prompt_play_by_cmd_id(1, -1, NULL, false)
}
```

Figure 5. Code Program 2

2.2 Signal Generation Module

In this design, the TWEN-ASR-ONE development board is used to control the AD9910 high-speed DDS module.[6] AD9910 Is a direct digital frequency synthesizer (DDS) with a built-in 14 bit DAC that supports a sampling frequency of up to 1 GSPS. Because AD9910 has advanced DDS patented technology, it can greatly reduce power consumption without sacrificing function. Its functional block diagram is shown in Figure 6.

Figure 6. AD9910 Functional Block Diagram

Theoretically, the output frequency of AD9910 high-speed DDS module is between 0 ~ 420 MHZ, and the output peak value is between 0 ~ 800 mV. The minimum resolution of the frequency is 1 HZ. The peak value is controlled by the 14-bit control word, that is, the control data 0~16383 corresponds to the output peak value of 0 ~ 800 mV. The minimum resolution of the peaks was approximately 0.0488 mV.[7]

2.3 Display Module

The display module used in this paper is 0.96 inch OLED screen. OLED screen, OLED screen is compared with LCD screen, which can achieve low power consumption, fast response and high price ratio based on the display output waveform parameters. In this design, the display is a 0.96-inch OLED screen controlled by IIC communication mode. It only needs four IO ports to drive the screen display, greatly saving the IO port resources. The memory of OLED is provided by SSD1306, and the relationship between SSD1306 memory and the screen is shown in Figure 7.

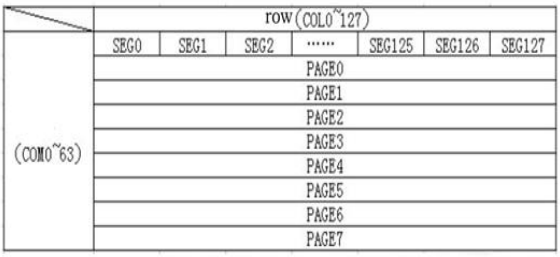

Figure 7. SSD1306 Video Memory and Screen Correspondence

3. System Test Results

After combining the modules of each part of the new generation signal generator of ASR-ONE, it is also necessary to test the real voice signal, and find the problems and adjust the software according to the functions and indicators of the digital signal generator. So that all aspects can meet the expected standards, even better than expectations. The test of this design focuses on whether it can quickly respond to the voice signal and external voice reply, the signal output interface to the oscilloscope, observe whether the waveform on the oscilloscope is smooth without distortion, whether there is a burr, frequency and peak peak parameters are within the expected error range, and the size of the error.[8]

3.1 Voice-controlled Waveform Output Test

In Test 1, the human body said "waveform output mode, change the waveform to triangular wave, peak value of 365 mv and frequency of 244.14 kHz". The test result is the voice reply "already set", the OLED displays correctly, the output waveform is triangular, the waveform is smooth and burr, no distortion, the output frequency is 244.14kHZ, and the output peak value is 368 mv. The test result plots is shown in Figure 8.

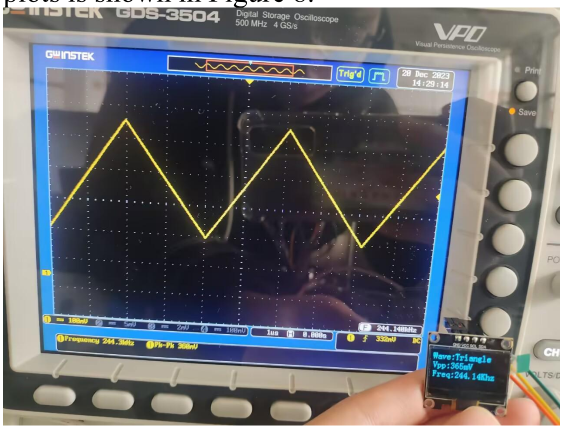

Figure 8. Voice Control Waveform Output Test 1 Test Result Diagram

In Test 2, the human body issued the voice "waveform output mode, change the waveform to sine wave, change the peak value to 141 millivolts, and change the frequency to 10.236 kHz". [9]The test result is the voice reply "already set", the OLED display is correct, the output waveform is sine wave, the waveform is smooth without burrs, no distortion, the output frequency is 10.2360kHZ, the oscilloscope shows the output peak value is 140 mv. The test result plots is shown in Figure 9.

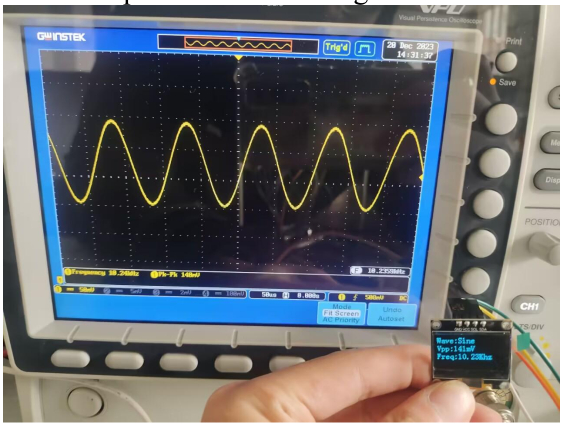

Figure 9. Voice Control Waveform Output Test 2 Test Result Diagram

Test 3, keep the result of test 2 unchanged, and the human body said "change peak to 500 mv". The test result is the voice reply "already set", the oscilloscope shows the peak value of 496 mV, the other waveform parameters remain unchanged, and the OLED shows the peak value parameter changed to 500 mV. The test results diagram is shown in Figure 10.

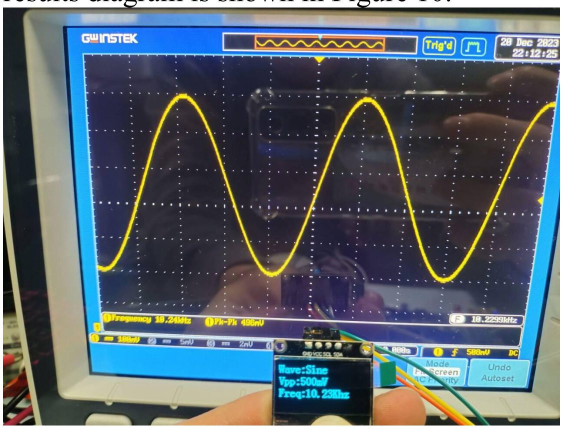

Figure 10. Voice Control Waveform Output Test 3 Test Result Diagram

Test 4, keep the results of test 3 unchanged, and send from the human voice "change the frequency to 10 MHz". The test result was the voice response "set", and the OLED display frequency parameter changed to 10 Mhz. The frequency of the output waveform is changed, and the remaining waveform parameters are unchanged.[10] The test results plots are shown in Figure 11.

3.2 Data Test

Finally, the data is tested more carefully. According to the measured data, the output of the designed frequency is more accurate when it is greater than 1 HZ, and the error is less than 1% and negligible. When the frequency was 1 HZ, the frequency error reached 7%. The output has 420 Mhz low pass filter, its amplitude decreases with increasing frequency. The test data are shown in Table 1. The total process time from the end of the human voice to the completion of the change of the OLED display result, the change of the output waveform, and the start of the voice response is less than 1s.

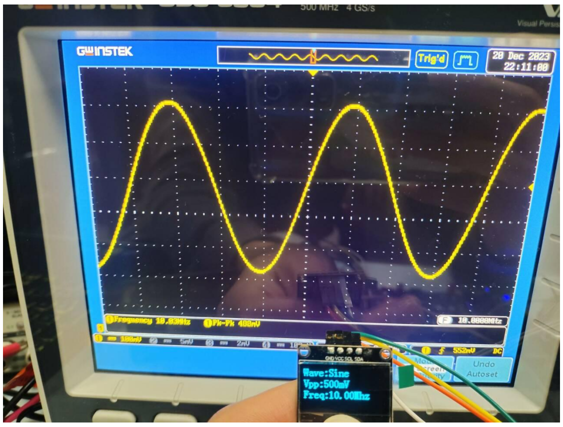

Figure 11. Voice Control Waveform Output Test 4 Test Result Diagram

Table 1. Test data

NO.	Voice control frequency	Voice control Peak peak value	Whether the OLED display is correct	Oscilloscope test frequency	Oscilloscope tests the Peak peak value
1	1Hz	250mV	yes	931.5mHz	250Mv
2	55Hz	300mV	yes	54.9587Hz	300Mv
3	100Hz	350mV	yes	99.8792Hz	344Mv
4	3kHz	400mV	yes	2.9998Hz	392Mv
5	10kHz	400mV	yes	9.99990kHz	380Mv
6	125kHz	500mV	yes	125.000kHz	460Mv
7	1MHz	550mV	yes	1.00000MHz	508Mv
8	15MHz	600mV	yes	15.0001MHz	548Mv
9	75MHz	650mV	yes	75.0003MHz	524Mv
10	100MHz	700mV	yes	100.000MHz	520Mv
11	350MHz	750mV	yes	350.001MHz	658Mv
12	420MHz	600mV	yes	420.002MHz	392Mv

4. Conclusion

The new generation of signal generator based on ASR-ONE has the advantages of portability, small size, low power consumption, available voice control, high price ratio and high accuracy. This design can allow the engineer to free his hands in the process of use, so that the engineer in the process of circuit debugging more convenient and fast. After the optimization of the software, the accuracy of the new generation signal generator based on ASR-ONE has been improved, and the output waveform is also more beautiful. In the frequency range, the peak value output error is within 5%, while the frequency output error is below 0.1%. However, in the waveform display part, the output of ordinary sine wave, triangular wave and square wave is relatively simple. If the subsequent development continues, this work can generate various modulation waves through the external adder, multiplier and other constituent circuits.

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