Design of Digital Voice Storage and Playback System

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Abstract: The traditional tape voice recording and reproducing system has many limitations due to its large volume and inconvenient use in the use of electronic and information processing. This paper presents a compact, low-power digital voice storage and playback system will be able to replace it. The basic principle of digital voice storage and playback system is the digital control of voice recording and playback. Among them, the key technology is that in order to increase the voice storage time and improve the utilization of memory, the use of non-distortion compression algorithm to compress the voice signal and then stored in the playback and then decompression while the input voice signal Filtering to suppress noise and interference, thus ensuring the reliable quality of voice playback. The system uses three methods, namely data acquisition, straight sampling, sample sampling and self-similar incremental modulation to complete the storage and playback of voice signals 32.7s, 65.5s and 147.4s respectively. The front straight AGC controls the speech signal within the controllable range of the A / D converter to ensure that the voice signal is not distorted. Bandpass filter a reasonable range eff ectively, filters out-of-band noise and reduces aliasing distortion. Through the post-level compensation circuit on the output of the voice signal correction, playback voice clear. The system has the functions of automatic recording, manual recording, recording and playing time display and protection of voice signal after power failure.

Keywords: Digital storage, playback, digital filtering, sampling, analog / digital conversion, calibration

1. Introduction

This paper expounds the research background, current situation and development direction of digital voice storage, playback system, points out the defects and problems of traditional voice storage, playback system, the advantages and prospects of digital voice storage and playback system.

1.1 Background

Digital voice storage and playback system to micro-processing chip as the core, with voice control, flexible, no wear, reliable and simple features. Thus in all kinds of public facilities, intelligent instruments, household electronics and other fields have a wide range of applications[1]. The system is currently a variety of programs can be achieved, which collected into a voice chip is a relatively simple and common program but the program is less intelligent such as the volume cannot be amplified and recording time is fixed. The system uses a different program to AT89C51 microcontroller as the core and designed a set of flexibility to achieve the recording, playback, volume control of the new voice storage and playback system.

1.2 Purpose and significance of research

At present, widely circulated voice storage means for tape recording, its large volume, the use of inconvenience, in the use of electronic and information processing by many restrictions. Hence the digital storage is the future development trend. Here we build the voice signal on the basis of digital, while in order to reduce the noise to improve the quality of voice and

volume stability using a band-pass filter and automatic gain control circuit[2]

1.3 Overview at home and abroad

Since Edison invented the phonograph in 1877, audio technology has been more than 100 years of development history, during which the record storage of various sound carrier, transmission and playback of voice technology development can be described as ever-changing. The system uses the microcontroller to control the recording, playback, fast forward, pause and other functions to control, with DSP technology for voice information processing with Flash ROM technology for storage, improve the voice playback quality and extended storage time, and box tape recorder compared to avoid the mechanical transmission noise, good sound quality, low power consumption with a clock function and friendly man-machine interface. It also the interrupt with control of recording and playback process to achieve the voice storage and playback of digital [3]

2. Comparison and demonstration of the program

2.1 Scenario 1

Voice coding scheme:

The human ear can hear the sound is a frequency range of 20 Hz - 20000 Hz, while the general voice frequency up to 3400 Hz. Voice acquisition refers to the voice of the sound signal by the microphone and high-frequency amplifier into a certain degree of amplitude of the analog signal and then converted into the entire process of digital. According to the Nyquist sampling theorem, the sampling frequency must be greater than twice the maximum frequency of the analog signal. Since the speech signal frequency is 300 to 3 400 Hz, the sampling frequency of the speech acquisition is set to 8 kHz. From the perspective of the storage and compression of speech, the model parameter representation is superior to the signal waveform representation [4]. However to be applied to the microcontroller, it is clear that the signal waveform representation is relatively easy to achieve. Based on this idea of the algorithm, in addition to some of the traditional pulse code modulation, the current has been

used VQ technology and some transform coding and neural network technology but the algorithm is complex, the current speed of the microcontroller, it is diffi cult to achieve. Combined with the actual situation, put forward the following several possible solutions.

(1) Short-term average cross-zero counting method program by determining the signal across the zero, the voice signal encoded as a digital signal, commonly used in speech recognition. But for the microcontroller, due to the ability to deal with data at the end of the method is not easy to achieve.

(2) Real-time subtraction sample sampling process shown in **Figure 2.1**.

Figure 1; Sampling process**.**

The concrete realization includes three kinds of basic methods, such as direct access method, under sampling method and self-similar incremental modulation method. The third method is the most unique, the method can make the data pressure 1: 4.5, both the advantages of modulation and also both the advantages of PCM coding error is small, the coding error does not spread backwards.

A / D, D / A and memory chip selection

Single-chip voice generation process can be seen as the reverse process of voice acquisition process but not intact to restore the original voice and the original voice can be controlled and reorganized real-time recovery. In the playback, as long as the original sampling directly through the D/A interface, you can make the original sound to reproduce.

(1) A / D conversion chip selection According to the requirements of the sampling frequency fs = 8KHZ, word length = 8, select the conversion time of not more than 125μs eight A / D converter chip. A / D conversion is currently used to achieve a variety of methods, in view of the conversion speed requirements, we use A / D converter chip AD574. The chip is a high-speed 12-bit successive comparison A / D converter, built-in bipolar circuit composed of hybrid integrated conversion film, with external components, low power consumption, high precision, and has automatic zero and automatic polarity Conversion function, just a small amount of external RC component can constitute a complete A/D conversion [5].

(2) D / A conversion chip selection D / A conversion chip is the role of the stored digital voice signal into analog voice signals, the general analog converter can achieve 1μs conversion rate, enough to meet the requirements of the subject, So we have chosen the general-purpose D / A converter DAC0832.

(3) Data memory selection when the sampling frequency fs = 8KHZ, the word length is 8 bits, one second of the voice needs 8K bytes of storage space, the memory needs at least 80k capacity. Here we use the flash memory AT29C040 as a memory, a chip can store 60 seconds of language.

2.2 Option II

The system uses MCS-51 series microcontrollers, 256kB external RAM data storage area (using paging storage technology), using DPCM way to compress data, and the use of two stereo microphone for input, the differential amplification, with good performance of the five The use of the Schleswig bandpass filter, and the correction circuit[6].

Control mode

Controller using single-chip implementation, single-chip man-machine interface is good, and has a certain programmable capacity

For the speech signal (the highest frequency is about 3.4kHz, 8kHz sampling frequency), the 6MHz crystal frequency of 8031 is sufficient (each sampling period 125, equivalent to 125/2 = 62 machine cycles, averaging 31 instructions).

Amplifier and A / D, D / A chip selection

To reduce the system noise level, increase the dynamic range of the system to prevent blocking distortion and set the automatic gain control circuit in this amplifier. It has both analog and digital implementations. Digital high precision, large control range than the simulation test complex, so the program uses the traditional simulation test AGC to achieve. With TDA2030A as a power amplifier can drive the speaker sound, and has a certain power margin[7].

A / D, D / A and memory chip selection: As the subject requires the highest frequency of the voice signal is 4kHz, according to Nyquist theorem, the sampling frequency = 8kHz (cycle = 125), can be no distortion to restore the original voice. In the absence of special requirements, the word length can be selected eight, but

taking into account the scalability of the system so the conversion time for this 35 AD574. According to the same analysis, the conversion frequency is chosen to be 8kHz using the DAC0832. Memory using 256Kb RAM available 628256 implementation.

2.3 Program III

The program uses the microcontroller 8031 as the core device and the 128kBRAM array is the data memory. 8031 typical clock for the 6MHz instruction cycle can be in the required 125 sampling interval to perform the system work can also be on the A / D converter output digital voice signal incremental modulation or diff erential pulse code modulation [8]. And DPCM are two voice compression coding techniques that compress the speech rate from 64kb / s to 8kb / s and 32kb / s, respectively. In addition, in order to increase the recording and playback time, we use four 622526 composition RAM array, with 8031 P1 port to participate in address selection, using paging storage mode, the system can store data storage space to 128kB, 128kB space to store PCM code, DPCM code and voice playback time up to 16s, 32s and 128s to achieve the subject requirements.

The above three programs have their advantages and disadvantages, taking into account its ease of use, simplicity and other factors decided to take the first program.

3. System overall structure

Digital voice storage and playback system, the basic idea is through the pickup to the sound signal into electrical signals, and then amplified by the amplifier, and then through the bandpass filter, analog voice signal through analog-to-digital conversion (A / D) into digital signals, And then through the microcontroller control data from the memory read out, and then through the digital-analog conversion (D / A) into analog signals, amplified by the speaker or headphone output.

System components shown in the figure by the input channel, AT89C51 microcontroller and output channel composed of three parts. The input channel consists of a pickup, a preamplifier circuit and a band-pass filter. The output channel consists of a band-pass filter and a post-stage amplifier circuit [9]. The millivolt signal output from the pickup is measured

in the range of about 20 to 25 mV. This signal is too small to be sampled. The dynamic range of the A / D conversion input signal is 0 to 5V and the range of the speech signal is compared with the sampling range. The amplifier magnification should be about 200 times, whereby the signal through a gain of 46dB amplifier, amplified to the volts of magnitude, the output stage amplifier circuit also uses this circuit, two-stage amplifier circuit are used to tune the typical circuit. Taking into account the inherent characteristics of the voice signal, will be less than 300Hz and higher than 3.4kHz component filtering after the voice quality is still good. Here it is through a gain of 46dB amplifier, therefore, the bandpass filter is designed as a typical 300Hz - 3.4kHz, output stage bandpass filter is 300Hz - 3.4kHz, so that can filter out the low frequency components can Filter out the D / A conversion to bring the high-frequency components, a good filter out the noise. According to the Nyquist sampling theorem to make the sampling signal without distortion, the sampling frequency of the lowest 6.8kHZ, taking into account a certain margin, so that is sufficient to ensure voice quality.

4. Circuit design

4.1 Pickup

The pickup is an acoustic sensor, which is a sensor that converts an acoustic signal from an external sound field into an electrical signal. It has a wide range of applications in communication, noise control, environmental testing, sound quality evaluation, culture and entertainment, ultrasonic testing, underwater detection and biomedical engineering and medicine [10]. It is a lot of its type and frequency. It is divided into ultrasonic sensors, sound pressure sensors and surface acoustic wave sensors. The electrical principle of a simple magnetic pickup is that when the sound is vibrated in the coil wound around the copper wire, the magnetic induction line generated by the core wound by the coil is induced and the output signal is induced out of the coil. The magnitude of the induced current depends on the magnitude (amplitude), the cutting frequency (vibration frequency) and the strength of the magnetic induction line.

The pickup includes a pickup (transducer, stylus)

and a sound arm. The transducer device mainly piezoelectric, electromagnetic, capacitive and semiconductor [11]. Electromagnetic pickup head, with electromagnetic induction principle, the mechanical vibration into the electrical signal amplitude response pickup head. Mainly by the coil and magnet and other components. The stylus is coupled to the coil and is called a dynamic type. In addition, there are also coupled with the needle on the armature called the movable iron, also known as variable magneto-resistive. In this design decided to use moving coil pickup

4.2 Design of the amplifier

Gain amplifier pickup output millivolt signal measured its range is about 20 - 25Mv

This signal is too small cannot be sampled, after the A / D conversion input signal dynamic range of 0 - 5V, the scope of the voice signal and the sampling range of the amplifier to get the magnification of about 200 times, here will be The signal is amplified by a gain of 46dB, amplifying it to the volts, and the output stage amplifier circuit also uses this circuit, both of which use a typical circuit with gain adjustable [12]. In order to amplify the weak speech signal obtained from the pickup, the bipolar high input impedance of the co-amplifier, the circuit shown in **Figure**, each amplifier magnification calculated by the following formula:

$$A_{V1} = 1 + R_{P1} / R_3$$
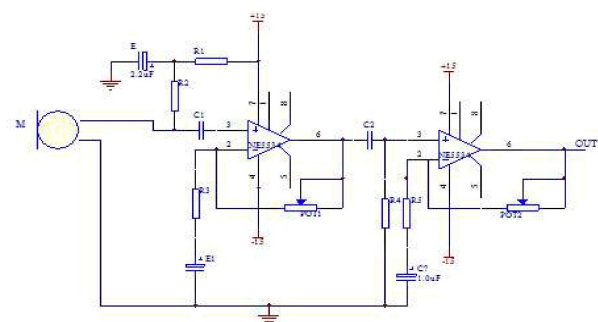
$$A_{V2} = 1 + R_{P2} / R_5$$

Figure 2; Gain Amplifier**.**

(2) Output amplifier through the bandpass filter output sound playback signal, the amplitude of 0 - 5V, enough to use the headset to listen to and cannot take any amplifier. However in view of the actual reuse of the speaker to the outside, so in this system to increase the extroversion function, front-end amplifier using general-purpose audio power amplifier LM386 to

complete[13]. The circuit is shown in **Figure 4.1**. The circuit gain of 50 to 200, continuously adjustable, the maximum distortion power of 325mW. Output termination C4, R9 series circuit to correct the frequency characteristics of the speaker to prevent high frequency self-excited. Feet 7 then 220uF to even capacitor to eliminate low frequency self-excited. In order to facilitate the work of the amplifier in the case of high gain, here will not use the input pin 2 to ground short circuit.
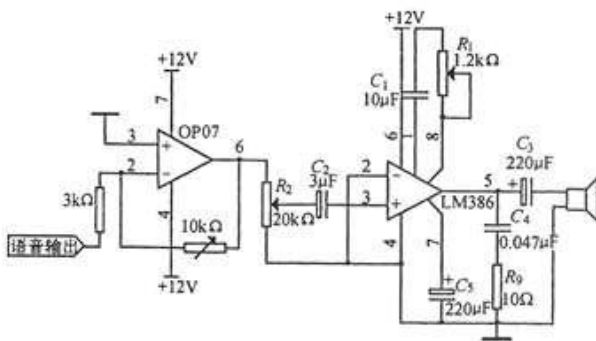

Figure 3; Output Amplifier.

4.3 Active bandpass filter design

A filter is an electronic device that enables a useful frequency signal to be simultaneously suppressed (or largely attenuated) unwanted frequency signals. Engineering commonly used for signal processing, data transmission and suppression of interference and so on. Here mainly discuss the analog filter. In the past, this kind of filter circuit is mainly composed of passive components R, L and C. Since the 1960s, the integrated operational amplifier has developed rapidly. It is composed of active filter circuit composed of R and C, which has the advantages of no inductance, small volume, Light and other advantages [14]. In addition, due to the integrated op amp open-loop voltage gain and input impedance are high, the output impedance at the end, constitute the active filter circuit also has a certain voltage amplification and buffering. However, the integrated op amp bandwidth is limited, so the current active filter circuit frequency is difficult to make very high, which is its shortcomings.

For the amplitude-frequency response, the frequency range of the signal that can pass is usually defined as a passband, and the frequency range of the blocked and attenuated signal is defined as a stopband. The ideal filter circuit should have zero-amplitude amplitude response and linear phase Response, and in the stopband should have infinite amplitude attenuation. According to the different position of the passband and the stopband, the filter can be divided into low-pass filter, high-pass filter, band-pass filter and band-stop filter.

The amplitude and frequency response is usually used to characterize the characteristics of a filter. To make the signal through the filter very small distortion, the phase and delay response should also be considered. When the phase response is linearly changed, that is, the delay response is constant, the output signal is possible to avoid distortion.

Here is the function of the transfer function for its phase angle.

The sound signal is converted into a voltage signal by a moving coil pickup, which is amplified by the pre-stage. Before the data acquisition, it is necessary to pass the bandpass filter in addition to the band clutter, and select the passband of the filter. The range is 300Hz - 3.4KHz. Its role is:

1. To ensure that 300 - 3400Hz voice signal without distortion through the filter;
2. Filter out the band low frequency signal, in order to reduce the external frequency and other components of the interference, greatly reducing the noise impact, the lower limit frequency can be extended to 270Hz or so;

Easy to filter out the band of high harmonics, in order to reduce the 8kHz sampling rate caused by aliasing distortion,

4.4 MCU AT89C51 and AD574 interface principle

AD574 and the basic components of the microcontroller system mainly SCM, A / D converter and computer interface. One microcontroller is the core part of the system, the host through the interface to start the microcontroller work, so that CPU resources to other requests open. The microcontroller sends a control signal to initiate the A / D converter for sampling, and then stores the conversion result in the dual port SRAM[22]. When the data in the RAM reaches a certain number, the microcontroller sends an interrupt request to the computer. After receiving the request, the host enters the interrupt service routine and issues a command to the

microcontroller to decide whether to continue sampling, while reading the data in the SRAM into memory. The hardware design of the system should mainly consider the connection of the three bus (control bus, address bus, data bus) on the connection. **Figure 2** shows an A / D converter and microcontroller AT89C51 interface circuit. The AD574 is a fully monolithic 12-bit successive A / D converter with a tri-state buffer that can interface directly with an 8-bit or 16-bit bus, eliminating the need for additional latches. As the AD574 chip comes with high precision reference voltage and clock, so no external circuit and clock can work at full speed, is a more commonly used in the speed A / D converter chip. The AD574 completes a full 12-bit conversion with a maximum of 35μs and is suitable for applications where the conversion rate is less than 30kb /s.

AT89C51 microcontroller is a typical representative of MCS - 51. As the interface system requires each signal measurement synchronization, that is, start the A / D converter to convert[23]. Therefore, 8031 must be completed at the same time start, respectively, read out the conversion results of the task.

Now the main tasks of AT89C51 are as follows:

(1) Receive the host sampling command. That is, the use of P1. 7 and the use of the query waiting for the host to issue a sampling command, when it is low, start the sampling process.

(2) Start sampling. The AT89C51 uses P2. 7 to control the AD574 read and start the conversion control line R / C after inverting and then control the chip select line CS (active low) via the NAND gate and the inverter. When P2.7 is high, all AD574 are in the start state, that is, set the AD start address are # FFFFH.

AT89C51 the WR, RD through the NAND gate connected to the AD574 enabled side, any valid signal will enable AD.

(3) Read and store the conversion results. All AD conversion is judged by the lower 4 bits of the P1 port. When the low 4 bits are low, it means that all conversions have ended. You need to assign addresses to the converter to read the conversion results one by one. In the case of a read operation, the address should be the address of the corresponding memory cell, since the last 2 digits

of the memory cell address are 00,01,10,11, and therefore, the operation of the cell means that the corresponding number of A / D The converter has read the operation. This can make the system can operate as many as 10 A / D number.

(4) To the host to issue an interrupt request. After the current storage area is full, an interrupt request is sent to the host to transfer data to the host. Since each memory area is 1kB, the state of P2.2 can be used to determine whether it has been completely occupied when storing data. In **Figure 2**, because AD574 chip with a clock, so no need to add clock signal. The circuit uses a unipolar input mode, can be 0 - 10V or 0 - 20V analog signal conversion. The upper 8 bits of the conversion result are output from D11 to D4, the lower four bits are output from D3 to D0, and are directly connected to the number bus of the microcontroller. Conversion to follow the left alignment principle, D3 - D0 should be connected to the microcontroller data bus high nibble. In order to

To achieve the start A / D conversion and conversion results read, AD574 chip select signal CS by the address bus secondary low A1 (P0. 1) to provide, read and write, A1 should be set to low. The CE signal of the AD574 is generated by the L1 and A7 (P0.7) of the microcontroller. R / C by RD and A7 by a level or non-door. Visible in reading and writing, A7 should also be low. The output status signal STS is connected to P3. 2-terminal for single-chip inquiry to determine whether the A / D conversion is over. The A0 of AD574 is controlled by the lowest bit A0 (P0. 0) of the address bus and can be used to implement full 12-bit conversion and send 12 bits of data into the data bus twice

4.5 Memory

Program memory

Program memory is used to store compiled programs, tables and constants. Program memory addressing range can have 64KB corresponding to this, the program memory address from 0000H start, the largest to FFFFH. Program memory addressing the law for the first film, after the film, on-chip and off-chip continuous, both generally do not overlap. For the chip with the program memory chip, the CPU controller

specifically provides a control signal EA to distinguish, when EA is no high level, the reset after the first implementation of the chip program memory program, when the program counter content over OFFFH, It will automatically go to the program to execute the off -chip program memory while the instruction is executed, and when EA is low, the program in the off-chip program memory will be forcibly executed. At this time more in the off-chip program memory stored in the debugger, the computer work in the debug state. It should be noted that the off-chip program memory stored in the debugger part of the addressing and on-chip program memory address can be overlapped, through the EA can be achieved by the exchange of access.

In the program memory, there are seven units with special purpose.

0000H-0002H: is the entry address of all the implementation of the program, 89C51 microcontroller reset, the CPU always from the 0000H unit to start the program.

0003H: External interrupt 0 entry.

000BH: Timer / Event Counter 0 overflow interrupt entry.

0013H: External interrupt 1 entry.

001BH: Timer / Event Counter 1 overflow interrupt entry.

0023H: Serial port interrupt entry.

002BH: Timer / Counter 2 overflow or T2EX negative transition.

When used, an absolute jump instruction is normally stored at these entry addresses so that the program jumps to the users' scheduled interrupt start address or jumps from the 0000H start address to the user-programmed initial program.

The data memory on-chip data memory has 16 bits and the addressing range is up to 64KB. So the capacity of the off-chip data memory can be as large as the program memory, the address from 0000H start, the largest to FFFFH. 89C51 microcontroller data memory with on-chip data memory RAM and special function register SFR: the former has 128 bytes, the address is 00H-FFH, can read and write any data; the latter also accounted for 128 bytes, the address Bit 80H-FFH; both are continuous without overlapping. On-chip data memory capacity is very small, often need to expand the off -chip data memory. Such as the expansion of a small amount of off-chip data memory, the capacity of not more than 256 units, it can also be 8-bit binary number addressing, starting from 00H, the maximum to FFH.

On-chip data memory can be divided into work register, bit address area, data buff er area and other three regions.

Working register area

In the low 128B internal RAM, the first 32 units (address 00H-1FH) for the general working register area, is divided into four groups (register 0 group, 1 group, 2 groups, 3 groups), each group of 8 work The register consists of R0-R7, which occupies 32 units. Which set of parameters selected by the program state word PSW RS1, RS0 these two bits of the decision, if the program does not require four 4 working registers, then the rest of the working register can be used for general memory to use.

Bit address area

The 16 cells of the 20H-2FH are bit-addressed areas, each of which is assigned a bit address, 8 bits per bit, 128 bits each. Its bit addressing range is 00H-7FH. Each bit of the bit addressing area can be treated as a software trigger and processed directly by the program. Procedures are usually a variety of program status flags, bit control variables located in the bit address area. Similarly, the RAM unit of the bit address area can also be used as a normal data memory by byte unit.

Data buffer

30H - 7FH is the data buff er, the user RAM area, a total of 80 units.

(3) Special function register

Accumulator A- Accumulator A is one of the most commonly used 8-bit special function registers, which can be used to store operands or store intermediate results. Most of the operands of a single operand instruction are taken from the accumulator. Use ACC to represent the symbolic address of A.

Register B Register B is an 8-bit register that is used primarily for multiplication and division operations. Multiplication operation, B stored in the multiplication, multiplication operation, the product of the high 8 bits and stored in the B; division operation, B stored in the divisor, after the operation, B and stored in the remainder. In other instructions, register B can be used as a general register for temporary data.

4.6 Memory selection

In the digital voice storage and playback design can be used AT29C040 Flash memory to store time, AT29C040 has the advantages of online rewritable, nonvolatile, reliable information storage, storage capacity and each piece of the capacity of 512K bytes. It is the latest foreign products, the product read and write in general and RAM the same, as 89C51 general Addressing 64K bytes[24]. (P1.1, P1.2, P1.3), for the address line, so that the address space extended to 512K bytes, and the use of points, respectively, the use of P1 port for address widening, the system plus three lines (P1.1, P1.2, P1.3) The management of the allocation of memory, that is, before the bus output address, the first three additional high address page selection, and then in the selected page for input and output operations. AT29C040 read time is only 70ns, a single +5 V power supply, dual 8k bytes of the boot sector, internal program-controlled timer, hardware and software data protection, fast fan-out program cycle 10ms, low power: standby 100uA, start work For 50mA, 10000 times the number of erase. The input and output are fully CMOS and TTL circuits[25].

4.7 Correction of digitized voice storage and playback system

First, the function in the frequency domain is analyzed, and the curves in the frequency range of 30 - 4030H are shown in the figure. As can be seen from the figure, it is similar to the change in the gain in the stopband is extremely slow near the constant high-pass filter.
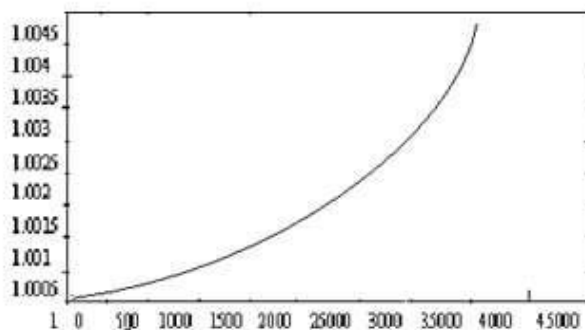

Figure 4; Function analysis chart.

The filter network is implemented using a simple passive filter network, which constitutes a network with a nearly constant magnification. It is observed that there is a significant attenuation at higher frequencies, so the network should also be at a higher frequency Of the attenuation characteristics, taking into account the sound signal, too much high-frequency components can only increase the noise, so then, constitute a low-pass filter, cut-off frequency is set at 3.4KHz100Hz. The network in the frequency domain from 30Hz, the gain slowly increased to 3.4kHz amplitude.

5. Software design

The microcontroller AT89C51 reads the A / D conversion data by chip select mode, writes the data memory, and sends the data to the D / A converter. The AT89C51 controls the sampling frequency through the timer when recording. Timed A / D conversion data is stored in the data memory. Playback, the microcontroller AT89C51 through the timer control, the timing of the data from the data memory removed to the D / A converter for digital touch conversion.

6. Conclusions

After more than two months of study, thinking and information access, and finally completed this graduation thesis. Through a wide range of reviews, through my theoretical analysis, can be considered in the hardware and software design is feasible, should be able to achieve the desired goal.

However, due to the rush of time, the system still has a lot of shortcomings, although the system mainly by the MCS-51 single-chip control based on the completion of the basic requirements to meet the basic requirements of digital voice storage and playback system, the system has voice sampling and / Touch, touch / digital conversion function, but also for a certain period of time the voice signal for real-time storage and continuous playback, this more sophisticated instrument requires the function of the system expansion, due to the limitations of the microcontroller pin, if you want to expand Function, it must be combined with a large number of combinational logic circuits and control chip, but can also be used to achieve multi-chip microcontroller, but if this design will make its volume increases. With the development of the electronics industry, a variety of modern electrical appliances are moving towards miniaturization, generally do not use this method. But in the circuit to change the hardware structure on the basis

of re-programming, so that the entire design process becomes relatively complex, due to time and level constraints, do not consider this situation.

In this design, I also note that the other is the lack of precision is the problem, because the DAC0832 sampling speed is not very high problem, and the system in the wiring, the device and the device between the mutual interference, making the system unstable. The output waveform has a glitch and other issues, so this simple digital oscilloscope there are some imperfections.

The last point is the stability of the microcontroller power supply, reset the circuit is too simple which makes the system anti-jamming capability is not strong, so that the system cannot be stable operation.

The above three shortcomings is that I can think of, certainly there are other aspects of the problem. I also hope that the teacher can guide and point out that the design of the short-comings.

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