Data and reports of the speech recognition system with memory function based on the server side

(Non-scientific report)

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In my statement,

Wu Xiang
Study data and report date of the server-based speech recognition system: July 21,2023

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abstract

The research objective of this project is to explore whether the speech recognition system can conduct memory cache processing on the server side and propose a better processing method. The project adopted python, developed for the base script, at the same time with the local computer as the hypothesis, write a common speech recognition system and another called segment processing speech recognition system code, and use large-scale data simulation technology for analysis, obtained the two different cpu usage, finally under the comparison named segment processing speech recognition system code under the same conditions and ordinary speech recognition system, can better memory cache and processing on the server.

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Enter the wav algorithm

Import the necessary libraries: The code first imports the modules and libraries that need to be used, including the os, pyaudio, and wave modules.

Set recording parameters: at the beginning of the code, define some recording parameters, such as audio block size, sampling number, number of sound channel, sampling rate, recording duration, etc. You can adjust these parameters as needed.

Create an PyAudio object: via the pyaudio. PyAudio () Create PyAudio objects for audio stream processing.

Turn on the microphone for recording: use the audio. The open () opens the microphone for recording, and sets the corresponding parameters, such as sampling format, sound channel number, sampling rate, etc.

Recording process: Read audio data from the microphone and add each

audio block to the frames list. Stop the recording flow: After the

recording ends, stop the recording flow, close the microphone, and

terminate the PyAudio object.

Merge audio data: merge the audio blocks in the frames list into complete audio data.

Create the output and backup directory: check and create the output and backup directory (if they do not exist).

Process duplicate files: If named "record" already exists in the output directory. Ffile for wav ", which the code will renaming to" record _1.wav "," record _2.wav ", etc. in the backup directory.

Save as a WAV file: save the audio data as a WAV file and set the corresponding number of sampling channels, sampling width, and frame rate.

Exception handling: Using try-except, block capture keyboard interrupt (press Ctrl + C) to stop the recording and turn off the stream and PyAudio objects correctly.

matters need attention:

Before running the code, make sure that the pyaudio library is installed and that the microphone is available.

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The Wav transformation algorithm

1. Import the necessary libraries: The code first imports the modules and libraries to be used, including os, shuti l and speech _recognition modules.

2. Create a speech recognizer object: use the ` sr. Recognizer ()` Create a speech recognizer object r for subsequent audio recognition.

3. Set the directory path: define the path for the input directory (INPUT _ DIR), output directory (OUTPUT _ DIR), and backup directory (BACKUP _ DIR).

4. recognize _audio _file function: This function is used for speech recognition of individual audio files. It uses the ` sr. AudioFile ()` Open the audio file and then pass the ` r.record ()` Load the audio data. Then proceed with Google's voice recognition service (` r.recognize _google ()`) Convert the audio to text. If the identification is successful, the identified text is returned, otherwise print the relevant error message and return None.

5. Move _ existing _ files function: This function is used to move the existing output files to the backup directory. If a file with the same name already exists in the output directory, the function renames it and moves it to the backup directory.

6. Get the list of files in the input directory: using ` os.l istdir ()` Get all files in the input directory and trathrough them.

7. For to. The file ending with wav, performs speech recognition: for each WAV audio file, call the `recognize _audio _file ()` function for speech recognition to obtain the recognized text.

8. If the text is successfully identified, save it as a text file: if the recognition is successful, the identified text is saved to the output directory, using the same file name as the input audio file, but replaced with. The txt file suffix.

9. If the identification fails, the corresponding output file will not be created, and the corresponding error message will be printed.

10. Notes:
-Before running the code, make sure that you have the speech _recognit ion library installed and an audio file in WAV format under the input directory (INPUT _ DIR).
-The voice recognition service in the code uses Google's service and therefore requires a connection to the Internet. If you have no access to Google services in the domestic network environment, you can try to use other voice

recognition services, such as iFlytek, and adjust the code to accommodate the service accordingly.
The- `generate _unique _filename ()` function is used to generate a unique file name to avoid heavy name files in the output directory.

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Sectional processing

1. Define the path of two script files: `script 1_path ` and `script 2_path ` point to two Python script files respectively.

2. Execute the first script: " 2023_ latest entry. The py ": using the `subprocess. The run ()` runs the `python ` command to start the Python interpreter and pass the `script 1_path ` as an argument. This performs the first script text
pi。

3. Execute the second script: "2023" _ the latest output.py ": After the first script is executed, the code continues to use `subprocess again. The run ()` runs the `python ` command to start the Python interpreter and pass the `script 2_path ` as an argument. This performs a second script file.

matters need attention:
-Before running the code, make sure that you have correctly specified the paths to both script files (`script 1_path ` and `script 2_path `).
-Ensure that Python is already installed in your system and can be run on the command line.
-Ensure that both script files exist and contain a valid Python code. Before running, it is recommended that you run both script files manually to ensure that they work properly when executed independently.
- `subprocess . The run ()` function waits for the child process to execute before executing subsequent code, so there may be a waiting time when running the script, depending on how long the script runs. If you want to execute two scripts in parallel, please refer to `subprocess. The Popen ()` function to implement the parallel execution

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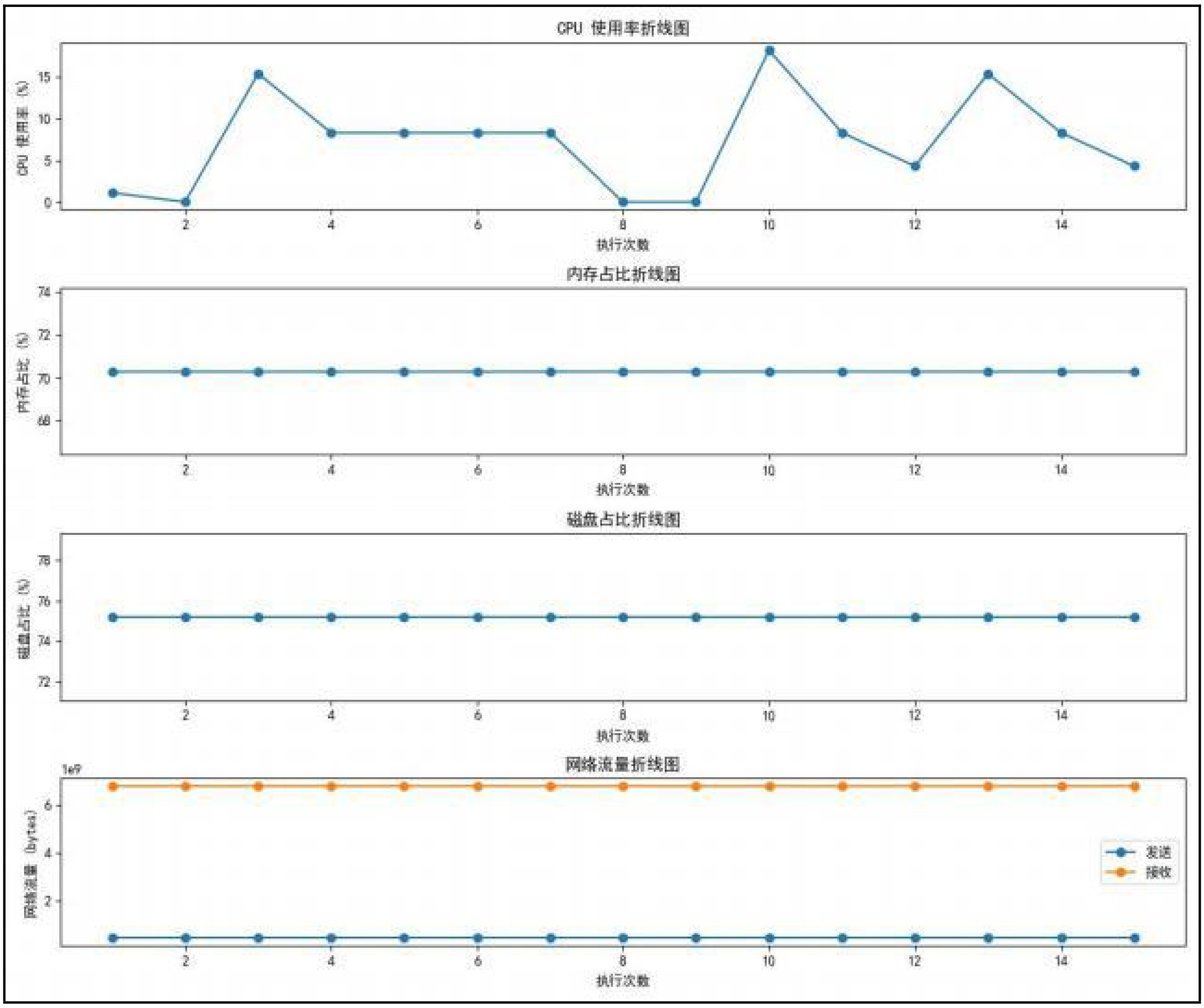
Data comparison: _ ordinary speech recognition



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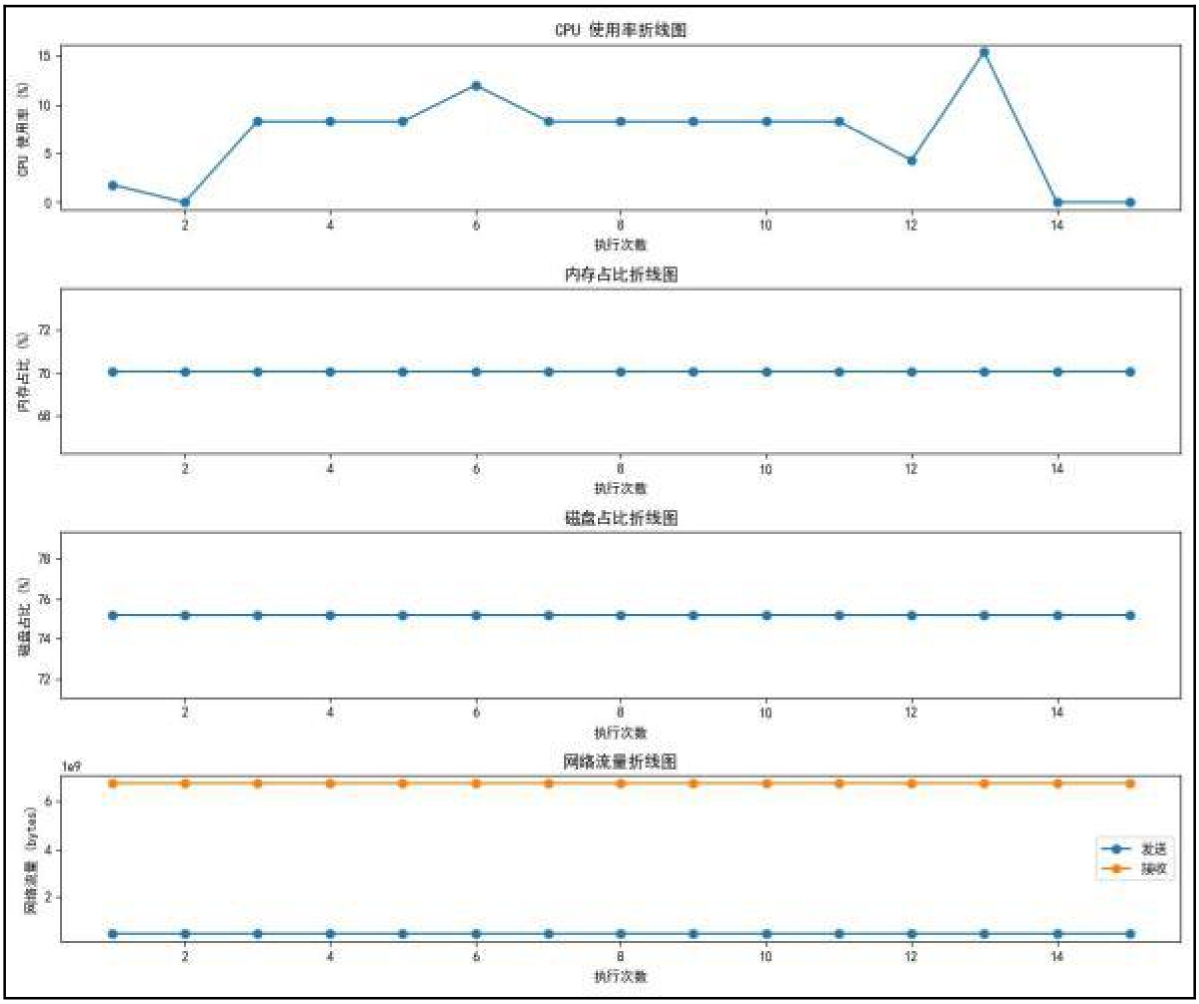
The segmentation processing of speech recognition



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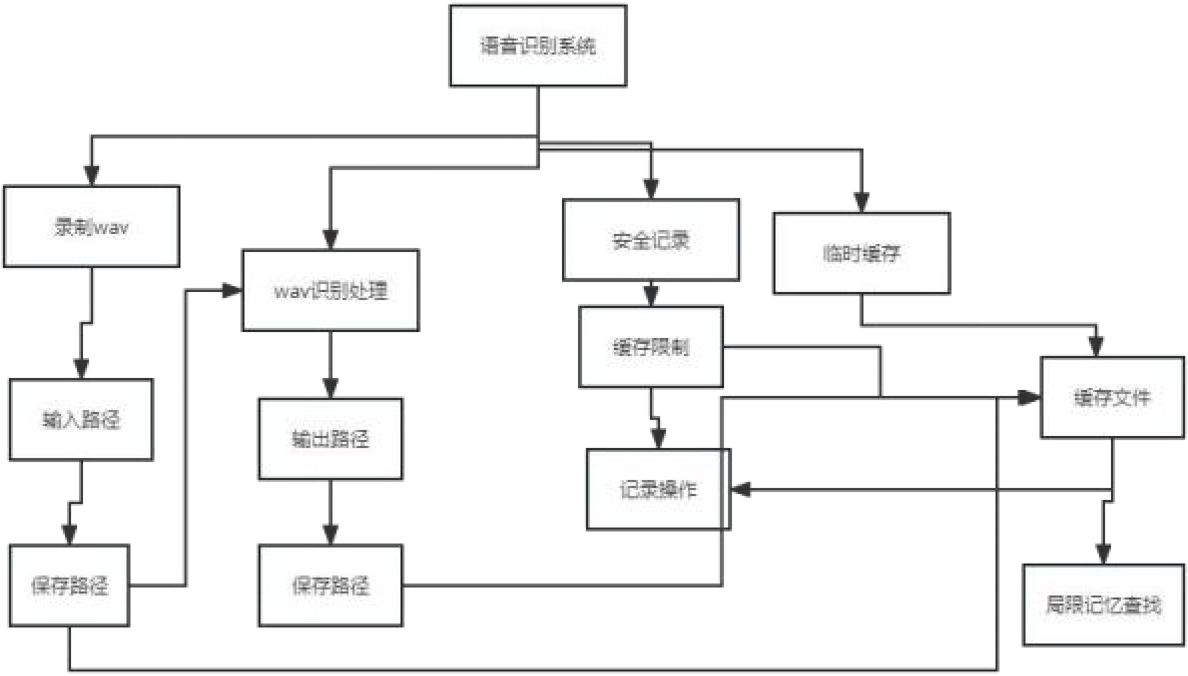
fundamental algorithm



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Memory cache

The project here assumes that the local computer is a cloud server, which has set up four different documents

spin	2023/7/21 8:36	文件夹
spin_1	2023/7/21 8:36	文件夹
word	2023/7/21 8:03	文件夹
word_1	2023/7/21 8:03	文件夹

Here we use the method of _ + 1 to increase the cache folder to make it memory storage.

(1) Spin >>>spin _1

Spin is the document for recording audio, and spin _1 is the server cache storage

(2) word>> word _1

iabove

* The purpose of establishing a memory buffer folder is to enable it to retrieve the memory and conduct secondary training when the content is required.