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Live Classroom System based on FFMPEG+RTMP Technology

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Abstract

This article first analyzes the problems of the current live broadcast platforms. After that, according to the problems and the needs of colleges and universities, the live broadcast classroom system has been designed, functional, and technical. Focusing on the level of use technology, delay optimization, message concurrency optimization, etc., and control the delay to the scope of 137ms-210MS; in the case of comprehensive consideration, the live broadcast class system is deployed to the three servers, and and the three servers, and and of. Establish a microservice cluster to govern multiple services through NACOS. After a semester trial, the live broadcast class interface is stable, the picture quality is clear, the delay is low, and the practical effect is good.

Keywords

Live Class; FFmpeg; RTMP; Delay; Concurrency.

1. Instruction

With the development of network technology, webcasting is becoming more and more popular, especially the epidemic in the past three years, which has entered the teaching of network live broadcast into the teaching of all levels. Compared with traditional offline teaching, live broadcast teaching is real -time, and is not affected by time and place. Students can learn open live courses at any time and anywhere. New education mode [1]. However, when using the Internet live broadcast platform, the number of people in the live broadcast will be unlimited, the classroom teaching interaction is poor, and the screen is unstable. [2-3] Therefore, how to build a live broadcast classroom unification with good live broadcast effects, controlled access, and good interaction is currently an urgent need for research.

Due to the difference in use technology, most of the live broadcast systems have many problems such as unclear video quality, poor live live broadcast, and non -good support for high -definition streaming media. In addition, most of the live broadcast platforms have adopted the encoding data according to the avoidance of concurrency problems and system stability, which greatly improves the delay of the live broadcast system, resulting in the difference between the live broadcast screen and the actual time is far from the actual time. Essence.

This article starts with problems such as low -time real -time and low codec efficiency of the existing live broadcast system. Based on the current mainstream FFMPEG audio and video processing framework, it optimizes audio and video collection, decoding, and encoding processes. Real -time HD live broadcast system based on the RTMP protocol. This framework has the characteristics of high transplantation, flexibility and efficiency of data processing, and provides a strong and reliable platform for audio and video processing of the live broadcast system.

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2. Live Classroom System Design

2.1 System Architecture Design

The designs of live classroom system mainly include four levels: access layer (mobile phone or PC), performance layer, unified connection layer (network protocol, gateway service, agency mechanism, etc.), and microservice management. The microservice management also includes business clusters, databases, and basic layers. The specific design is shown in Figure 1.

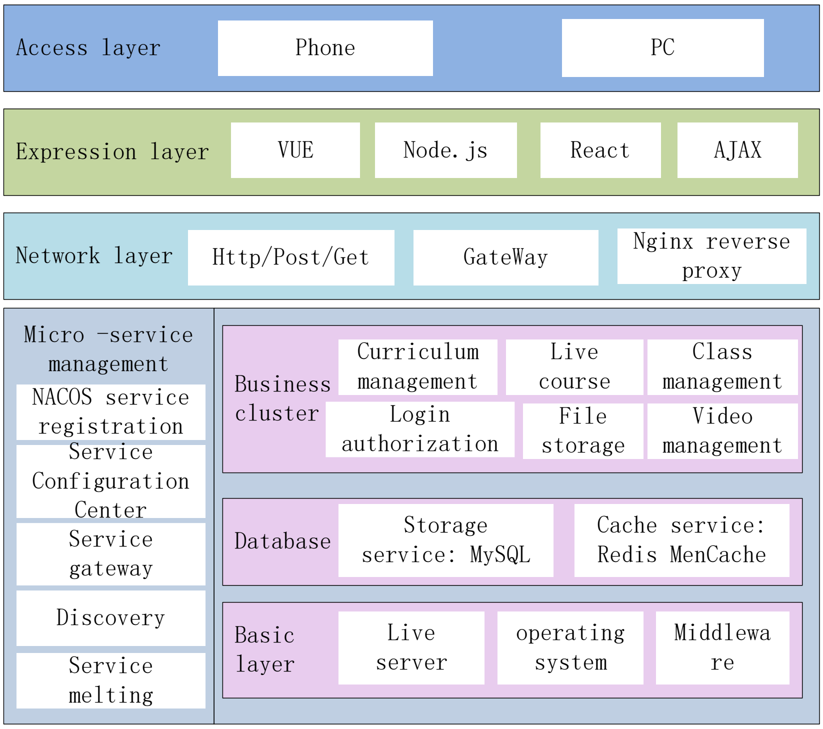

Figure 1. Live classroom system architecture

2.2 System Functional Process Design

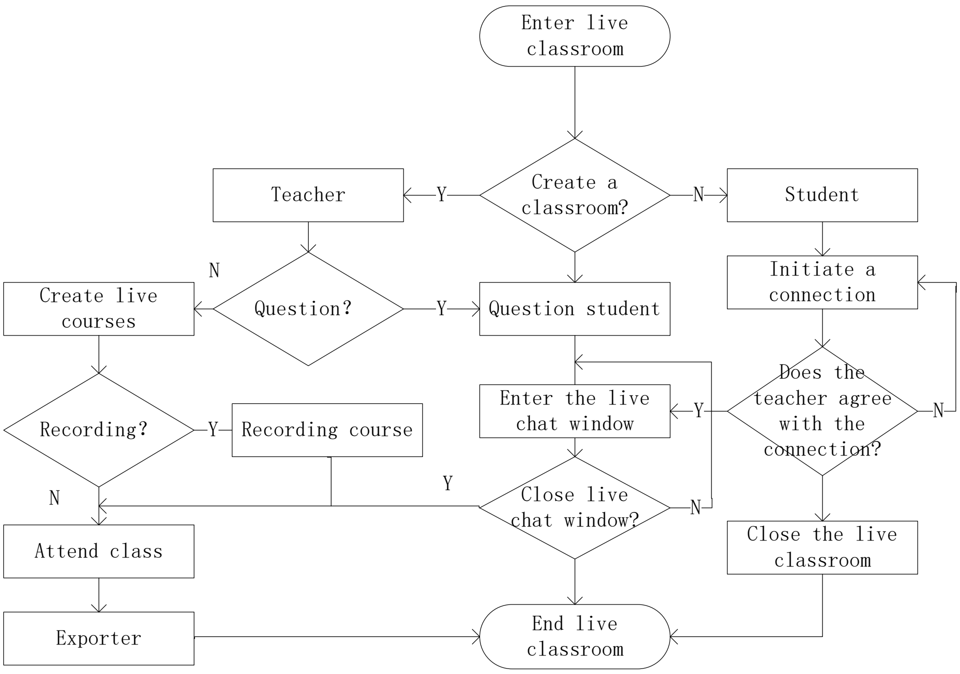

Figure 2. Live classroom function flow chart

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The live broadcast class adopts the microservices architecture, which is based on interactive models and the network as the main media. The system uses the B2C mode to separate the user from the management user. It is the front desk user system and back -end management system. The front desk system includes homepage data display, teacher information display and details Viewing, curriculum information display and details viewing, curriculum video online viewing, online broadcast online viewing, login registration, etc. The background management system mainly includes the user's authority management, the homepage broadcast map management, the front desk data statistics, the content of the curriculum, the video, classification management, teacher management, etc. The functional flow chart of the live broadcast classroom is shown in Figure 2.

3. Live Classroom Technical Framework

The technical framework of the live broadcast class is generally divided into three parts: anchor push end, RTMP server, and play streaming end. The server mainly forwards the data. The user's stretch end includes the protocol analysis layer, decoding layer, and rendering layer, as shown in Figure 3.

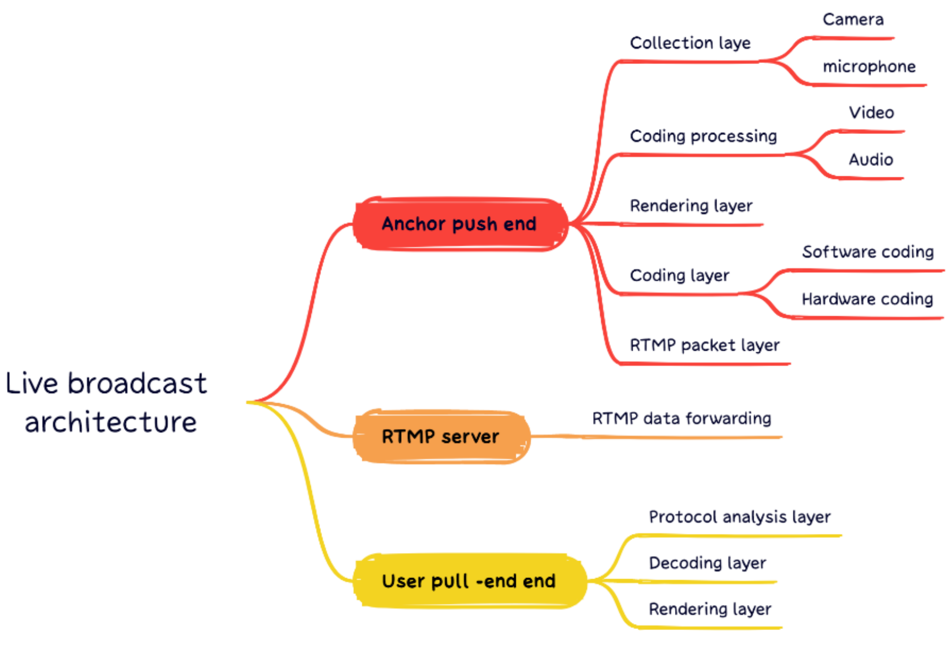

Figure 3. Live classroom technical framework

3.1 Live Technology Analysis

1) FFMPEG audio and video processing technology

FFMPEG [4-5] is an open source project containing a variety of audio and video processing methods. It has the leading code library LibavCodec, which is currently leading in the field of audio and video processing. plan. FFMPEG has high efficiency not only in audio and video processing, but also has very good scalability, which can be compiled and transplanted under multiple platforms. This system mainly uses Libavcodec, Libavformat, Libswscale, Libavdevice, Libavutil, which are used for audio and video coding, reused packaging, audio and video separation, format -sized stretching, data collection, data collection, data collection, data collection, and data collection of audio and video. , Reducing verification, etc.

2) RTMP's streaming media transmission protocol

RTMP [6-7] provides a two-way message multi-way reuse service on the basis of reliable streaming (TCP), which transmits parallel data related to time related to time, such as audio, video and data messages. The basic unit of the data transmitted by RTMP is Message, but the smallest unit transmitted in fact is Chunk (message block), because the RTMP protocol is Blocks, these blocks are

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chunk. RTMP is a TCP -based application layer protocol. Reliable communication is achieved through three handshake mechanisms. Its control flow and data flow are not separated and are organized as Message structure.

The RTMP starts with the network connection from the handshake. It contains three fixed size blocks. The three blocks sent by the client are named C0, C1, C2; the three blocks sent by the service terminal are named S0, S1, S2. After the connection is successful, the data information can be effectively transmitted. The handshake connection is shown in Figure 4.

The client first sends C0 and C1.

The server sends S0 and S1 after receiving C0 or C1.

When the client is accepted S0 and S1, C2 starts.

When the server collects C0 and C1, S2 is sent.

After the client and servers receive S2 and C2 respectively, shake hands to complete.

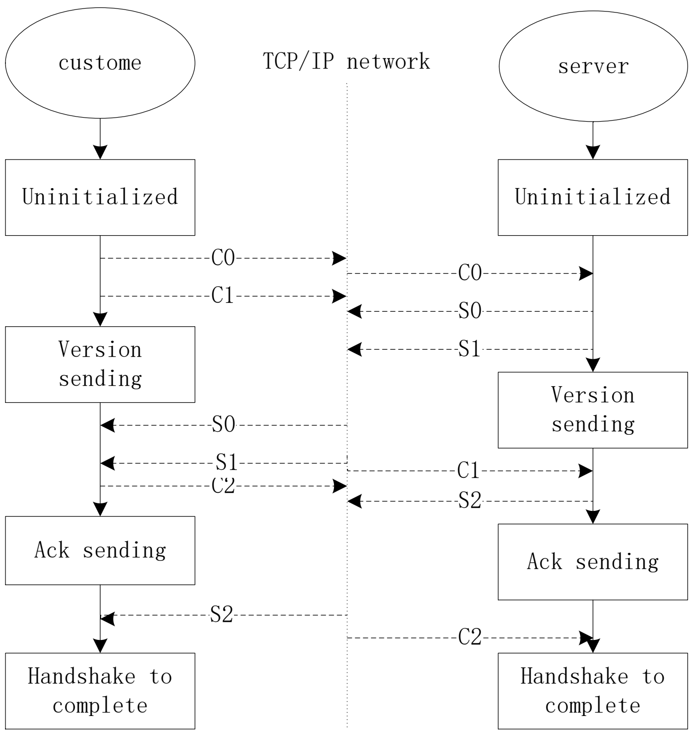

Figure 4. RTMP handshake connection

3.2 Delay Analysis

The delay of the live broadcast is mainly due to the common role of push -flow, server side, and broadcast stream end. This article is mainly analyzed the problem of live broadcast delay on the server side and both ends of the path. The terminal cavity size and slicing size. By converting the complex application model of the server end into a minimalist technology model, the minimalist technology model first performs a delayed analysis. The average delay of the plan. The test conditions of this article are as follows:

Data stream model information: single road 720p@25 1200kbps video stream +44100 dual -channel 16 bit rate AAC audio stream;

Delay time path: Push from the anchor to the rendering display layer → streaming media server → user pull -end rendering layer delay calculation.

This system is designed with a multi -threaded scheme. After the analysis of the various nodes of the overall architecture of the live broadcast, we believe that the thread with each thread below 40ms processing capacity is considered to be delayed. Therefore, the delay point is calculated:

1) Delay of hardware equipment: Simple definition 5 to 10ms takes the maximum value of 10ms;

2) Code swallowing 1 frame caused delay: 40ms;

3) Packing package delay: 0 ~ 23ms take the intermediate value 12ms;

4) Network transmission delay: RTT20MS;

5) TCP Cumulative delay: 20ms ~ 80ms takes the intermediate value 50ms 50ms;

6) Streaming media RTMP server buffer delay: Generally, the delay is 0ms. When special circumstances, the first frame can be severely stuck, resulting in the delay time reaching seconds.

The overall addition, the scope of the delay is: 97ms ~ 130ms, and the normalization delay is about 162ms. Because of the actual application we have ignored the impact of various thread scheduling, the Ringbuffer at each stage may have 1 to 2 frames in cushioning, so the scope of live broadcast delay fluctuations should be increased by 40ms-80ms-210ms on the basis of the above range.

3.3 Delay Optimization Strategy

1) Reduce data encoding conversion. In the case of neglecting the output delay of the equipment hardware, the output supports 720p@25 yuv420p data through the input data processing of the camera, thereby reducing the performance loss required for the thread processing before the camera data reaches the encoding. Directly output 720p@25YUV420P to the encoding layer and rendering layer, and the code can be delayed at 0ms in theory.

2) Coding processing adopts multi -threaded asynchronous processing method, as shown in Figure 5. Before the data frame is reached by the next frame of the collection layer, the preset pre -processing thread is used to process the first frame data. After that, directly copy it to the pre -processing thread to deal with the time -consuming operation. The processing time is far less than 40ms. After the processing of the processing thread is finished.

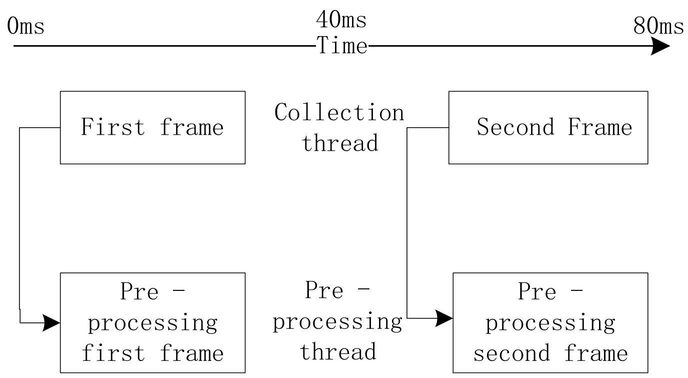

Figure 5. Multi -threaded asynchronous treatment

Optimize the server keyframe cache strategy: In the server's single video stream GOP range, the streaming end may make a pull request at any time node. When the pull request is just after the first I frame, the streaming end needs to wait for a GOP Time can be parsed to the next I frame, causing the first frame to parse the stuck. By configuring the cache in the RTMP server, this system allows the server to cache the current GOP keyframe, thereby ensuring that when the newly pull request of the service, get the B/P frame of the current time, improve the analysis time, solve the possibility of solving the possibility The first frame analysis of stuck problems appeared to avoid the phenomenon of video stream second -level delay.

3.4 Message Concurrent Analysis

When the number of live broadcast rooms reaches 500 and more than people, the database is likely to produce a large number of slow query and lock locks, which will cause some functions of the live

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broadcast room to have a brief failure, and at the same time, it will be stuck on the student side. This live broadcast system mainly uses the following ways to solve the problem of message concurrency.

1) Establish HTTP requests, combine CSS, JS, pictures, etc., optimize the live broadcast webpage; set up a anti -theft chain mechanism to prevent excessive malicious requests other websites.

2) Use CDN to accelerate, cache some static resources such as pictures, videos, etc. to the closest network node to the user, solve the problem of overload problems caused by the large number of users, shorten the user to watch the delay, thereby improving the server's response speed.

3) Take into account the load balancing. Use the Nginx agent and the Gateway gateway technology to distribute the request to multiple servers for execution, and then cooperate with the CDN acceleration to maintain the overall performance of the server cluster.

4) Database optimization, use Redis and MySQL cache to achieve fast access to data; use some tools, such as MYCAT to do some splitting work or reading and writing separation of the system, and put the database of query and written databases in the live broadcast system Separate to share the pressure of the database, while ensuring that the performance of MySQL is not damaged, and the speed of access is improved.

3.5 Video Data Collection

In the SCR dynamic link library, screen information collection and filtration methods have been encapsulated. During the user use, the system API interface will automatically call the packaged DLL to obtain the anchor screen information. Enter the flow of the video to read the video frame. Until the video frame is successful, the decoder will automatically decoding; if the read fails, it will be read again, as shown in Figure 6.

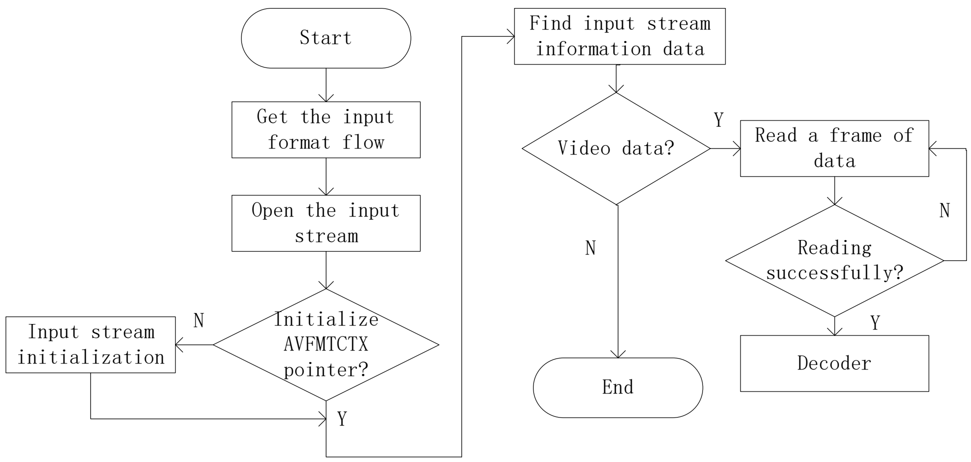

Figure 6. Video collection process

4. Live Classroom Deployment and Implementation

4.1 Live Classroom Deployment

In order to ensure the security of the system, the live broadcast class background service and the database are decentralized and deployed in the inner network of the server. The service deployment of the live broadcast class is scattered in three servers with different performances. It is divided into a course live broadcast sub -service, curriculum business sub -service, user business child Serve. By deploying the course live broadcast sub-service in the server that is better in memory and storage, it is used to build an RTMP-based live server. When requesting the live service subsystem to play HTTP-FLV or RTMP stream, the server will trigger FFMPEG direction to the direction RTMP source pulls and pushes to the server local, temporarily slowing live streaming video and audio slicing in the server, and the push -streaming operation of the live server is completed. Live video flow can be

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optimized by adjusting the cache size in the live server, thereby improving the performance of the live broadcast service.

Curriculum business sub -services and user business sub -services are deployed in servers with large and small specifications. Its course business sub -service is mainly responsible for a series of logical processes in classroom teaching. User business sub -services are responsible for services and additional expansion services for teaching -related user behaviors.

4.2 Live Classroom Implementation

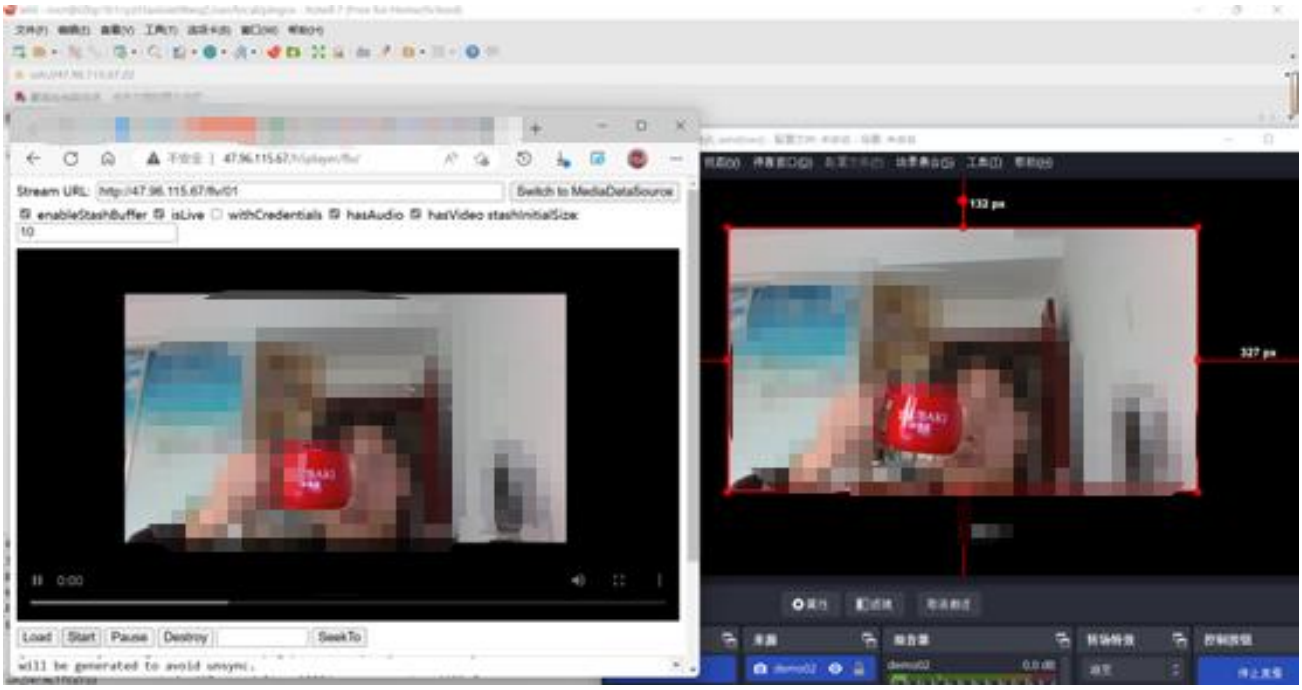

Figure 7. RTMP live server test (stretching end on the left, push end on the right)

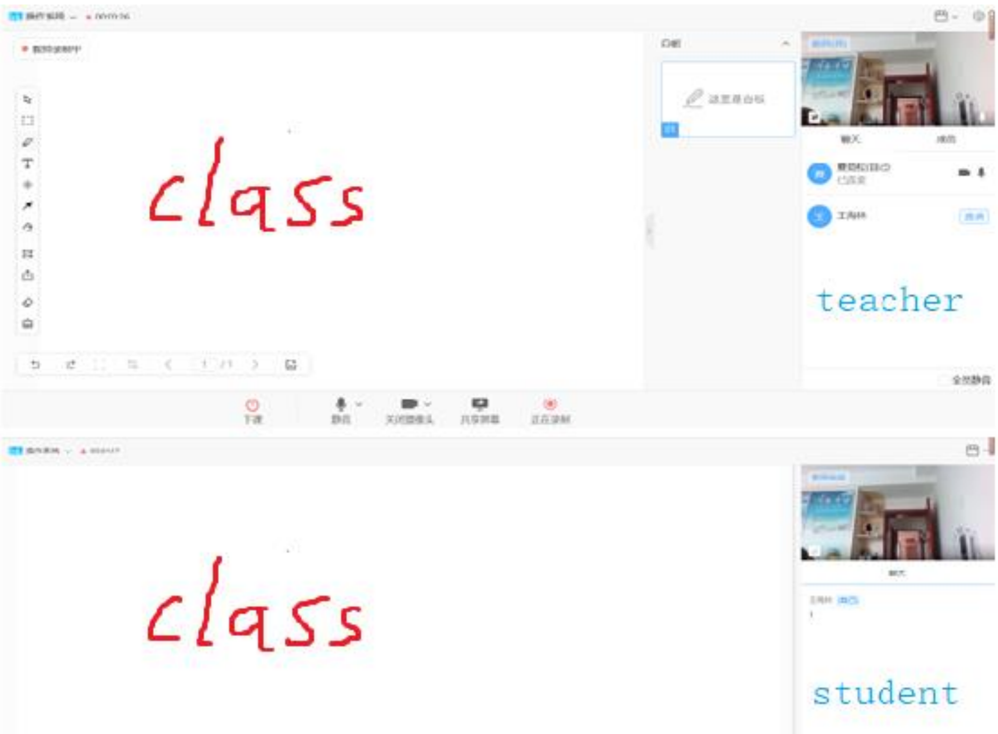

Figure 8. Live Classroom Interface

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This system architecture will split physical split on the machine on the core service, business services, and extension services. Each service system is registered to a unified registration center by setting up microservices cluster, and multiple services are uniformly governance through NACOS. Essence Use the nginx reverse agent and the configuration of the GateWay gateway to uniformly enter the background service. Data messaging interacts between the student side and the teacher side via the live server, and the video flows to the server locally through the live server, and then directly transmitted to the CDN (Content Delivery Network) to the student side and the teacher end. ; This system can run on the PC and mobile browser. The system adapts to different platforms, where the web side does not need to install any program. Implementation effect is shown in Figure 7-8.

5. Conclusion

The live broadcast classroom system is mainly analyzed from several aspects of system architecture, system design function, technical framework, deployment and implementation. First, a detailed research on the problems of the current live broadcast platform and the problems faced by the live broadcast classroom With the system function, select the current relatively stable technology and optimize the technology. Especially for detailed analysis and optimization of delay and news concurrency, to ensure that the live broadcast classroom screen has clear, stable, and small delay. After that, the live broadcast class was deployed, and various factors were comprehensively considered, and the service deployment of the live broadcast class was scattered in three servers with different performances to ensure that the stability and transmission of the live broadcast was smooth. After a semester trial, the live broadcast class system is stable, the picture quality is clear, and the delay is small. The overall adaptation of the current live broadcast of the classrooms of colleges and universities.

Acknowledgments

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