Development and Application of Remote Online Court Trial Mediation System Based on Internet Plus Mediation

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Abstract: Under the new judicial service mode of "internet plus Mediation", this paper adopts Web technology as the core and combines WebRTC real-time communication technology to complete the construction of remote online trial mediation system. The system will rely on Node.js to complete the design and configuration of signaling server, to support getUserMedia () algorithm and MediaStream () model object to complete audio and video collection, and adopt PeerConnection () method to complete the connection of different end users, so as to realize online mediation based on audio and video calls. At the same time, the system will also provide system login, instant messaging, file storage management and other functions with the help of Apache Web server. Through the remote online trial mediation system, the parties can participate in the trial through various terminal devices, which breaks through the limitation of time and space in the traditional trial mediation mode and greatly shortens the circulation cycle of the case. This not only improves the efficiency of court trials, but also enhances the richness of judicial services, thus promoting the construction and development of online trial mediation mechanism of "smart court".

1 INTRODUCTION

Court mediation, also known as litigation mediation, refers to a way of closing a civil case under the auspices of the trial organization of the people's court, in which the litigants negotiate on an equal footing, reach an agreement, and are recognized by the people's court to end the litigation activities. (Yu, 2021) As an important component of civil litigation, the development of trial mediation system marks the progress of China's civil trial system, and also confirms the significant achievements made in the construction of the rule of law in China. With the continuous improvement of the rule of law system, the people's court trial mediation system has gone through many stages and been continuously strengthened, and it has become the best way to solve civil rights disputes. However, in practice, the traditional court trial mediation mode is facing many challenges. First, China's civil litigation cases continue to grow at an average annual rate of 10%. The huge number of cases requires a lot of time and energy, which seriously restricts the deepening of mediation work. Secondly, the judge's handling of the mediation procedure is arbitrary, the mediation behavior is not standardized, and the necessary methods and skills are also lacking in the process, resulting in the low efficiency of the whole work, resulting in a long cycle of case circulation and fruitless adjustment for a long time. Third, the mobility of the parties in civil cases is relatively large, and regional differences and time conflicts have become important reasons that hinder the normal development of trial mediation. In the new era of social development, it is urgent for people's courts to update the concept of judicial service, improve the modernization level of judicial service and innovate the construction of civil dispute resolution mechanism.

At present, with the rapid development of digital information, the new model represented by "internet plus Mediation" is constantly promoting the transformation of mediation activities to networking and digitalization. The trial mediation of people's courts has also been deeply empowered by the new generation of information technology, and new dispute resolution mechanisms such as remote trial and online video mediation have been gradually integrated and innovated, providing new possibilities and implementation methods for solving civil disputes. In view of this, this paper holds that, based

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on the new judicial service mode of "internet plus Mediation", using Web technology as the core, and combining WebRTC real-time communication technology to complete the construction of the remote online trial mediation system, it provides a set of comprehensive application solutions for the shortcomings of the court trial mediation mode. The system will take the people's court as the leading factor, fully consider the use needs of the parties to the case, complete the networking and digital construction of the court trial mediation process, set up a standardized mediation implementation standard, and take into account the requirements of rationality and legality, thus forming a new mode of court trial mediation in civil cases. In the process of using the system, the judge is connected with the original and the defendant by remote video, and the parties' identity check, online investigation, court mediation and signing of mediation agreement are carried out according to law, which effectively breaks the time-space barrier of the traditional court mediation system and crosses the barriers of different mediation paths. It has become an important measure for the people's courts to strengthen their own construction, improve their own functions and enhance their judicial service capabilities, and it is also a new attempt to further realize the construction of "smart courts". (Ding, 2019)

2 INTRODUCTION OF KEY TECHNOLOGIES

2.1 Web Technology

Web is a distributed network information service system for publishing, browsing and querying information. The whole structure involves multiple hardware devices or software systems, aiming at solving the storage, analysis and application problems of data information that can't be completed by a single device. (Wu, 2020) The essence of Web architecture is an interactive process of data information, that is, the processing of request content and the presentation of response content are completed under certain preconditions. Figure 1 shows the flow chart of the standard dynamic resource Web request and response mode. Among them, the Web architecture can be divided into two parts according to the functional category: Web client and Web server, and the Hyper Text Transfer Protocol (HTTP) will be adopted between them to ensure the safe transmission of data information. Files is a static resource, and Web Application and DataBase together form a dynamic resource, both of which will obey the call of Web Server. Static resources are mostly designed HTML pages, which are quick to call and present, but poor in interactivity. Dynamic resources can respond freely according to the application program, which has strong interactivity. Based on the Web architecture, all technologies applied in the construction process are collectively called Web technologies, which contain many contents. According to the application design of Web client and the programming and development of Web server, Web technologies can be divided into two categories: client technology and server technology. Common client technologies include HTML, CSS, JavaScript, while server technology will be subdivided into three branches: scripting language, database and server. At present, the mainstream scripting languages are ASP.NET, Java, PHP, Python and so on.

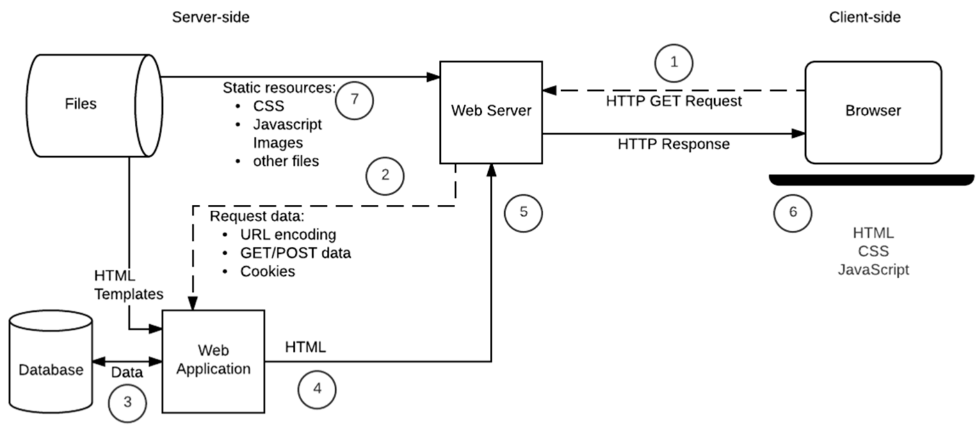

Figure 1: The request / response process for the Web

2.2 WebRTC

WebRTC, namely Web Real-Time Communication, is essentially a real-time communication technology on the web side, which aims at realizing end-to-end real-time audio-video interaction between browsers quickly and conveniently. (Zhang, 2020) WebRTC is not a single type of technology, but a real-time audio and video technology system that integrates audio and video acquisition, coding and decoding, network transmission, multi-dimensional display and other functions. It is functional, open-source and universal, and it is a Web application development framework conforming to HTML5 standard.

WebRTC realizes the real-time audio and video communication on the Web, mainly relying on three functional modules: audio and video acquisition, signaling server and connection management between the end and the end. Under the function of audio and video collection, WebRTC completes the whole process of audio and video content from collection to display through a series of processing frameworks of audio engine (VoiceEngie) and video engine (VideoEngie). Among them, getUserMedia () algorithm, as an internal preset and packaged API, can quickly obtain audio and video streams of devices, and give a fixed label. When the end-to-end connection is completed, the MediaStream object MediaStream can be sent to other clients according to the label. In addition, different codecs can be selected to control the content quality of audio and video streams under WebRTC. ISAC encoder can set audio sampling frequency, rate range, algorithm delay and other parameters, and realize echo cancellation, noise suppression, trembling prevention and packet loss compensation with NetEQ. VP8 video codec can expand the application scene of video streaming communication, and has both image stabilizer and image enhancement functions to meet the needs of high-quality video communication.

The connection management between the end and the end is the key to realize the real-time audio and video communication on the Web end. WebRTC will establish P2P connection by means of the point-to-point connection communication object RTCPeerConnection. RTCPeerConnection, as a unified interface for network connection, media management and data management, will focus on the configuration of session description information (SDP) and interactive connectivity establishment mode (ICE). SDP contains two parts: the description of session level and the description of media level, which must be transmitted during the establishment of P2P connection. The key description is shown in Figure 2. (Zhou, 2020) However, the setting of ICE framework will focus on solving the problem of the best path selection for establishing network connection between the two ends, breaking the restriction of IP address conflict when the two ends are not in the same public network environment.

The SDP and ICE information under WebRTC will be exchanged in a series after the P2P connection is established. All the processes involved in this stage are collectively called Signaling, and the control and deployment of the whole exchange process need the support of signaling server. WebRTC itself does not provide a signaling server, so it completely needs users to design and implement it according to their own needs. In the actual development and application, the socket.io class library under WebSocket is usually used to build the signaling server.

Session description
v= (protocol version)
o= (originator and session identifier)
s= (session name)
c=* (connection information -- not required if included in all media) One or more Time descriptions ("t=" and "r=" lines; see below)
a=* (zero or more session attribute lines) Zero or more Media descriptions
Time description
t= (time the session is active)
Media description, if present
m= (media name and transport address)
c=* (connection information -- optional if included at session level)
a=* (zero or more media attribute lines)

Figure 2: Common SDP description content

2.3 Node.js

Node.js is a JavaScript running platform based on Chrome browser engine. Its core application lies in making JavaScript participate in the design and development of Web server, simplifying the process of building Web applications and improving the efficiency of system development. For the design and development of the signaling server in this paper, Node.js perfectly fits the characteristics that the server is more important than input and output in task scheduling and is not good at calculation. Node.js will cooperate with web socket to provide great technical support for developing real-time interactive applications with long connections.

2.4 Development Process

According to the above application requirements, the author completes the configuration and deployment of the development environment of the remote online trial mediation system. The development content of the system is divided into two parts. One is the design and development of audio and video online communication function module based on WebRTC technology. The second is to complete the integration and encapsulation of the system in Java environment, form a standard Web application, and publish it on the server side.

First of all, WebRTC technology has completed the functional encapsulation of the overall framework, and has been preset in Chrome, Safari and FireFox browsers. In the development process, you only need to call the device camera code in the page through JavaScript code, as shown in Figure 3. Secondly, the key to the realization of WebRTC technology lies in the design and development of signaling server and the realization of end-to-end connection management. The development environment of the signaling server will be based on the Windows10.0 operating system, the integrated development tool will be VisualStudio Code, and the development environment will depend on Node.js After the installation of Node.js is completed, use the require instruction to load the HTTP module, and use the http.createServer () method to create the server. The listen method is bound to port 8080, and the function is used to receive and respond data through request, response parameters, thus realizing the basic construction of signaling server. (Ge, 2020)

```
navigator.getUserMedia = navigator.getUserMedia || navigator.webkitGetUserMedia ||
navigator.mozGetUserMedia || navigator.msGetUserMedia;
    navigator.getUserMedia({video:true,audio:false},function(stream){
    video.srcObject = stream;
    },console.log)
```

Figure 3: GetUserMedia () algorithm calls the camera of the device to obtain video content

Finally, for the end-to-end connection management, it is necessary to continue to complete the corresponding settings under the signaling server. The whole process will involve user registration, call establishment and response. Among them, the key implementation lies in the SDP information exchange under RTCPeerConnection () method and the establishment of ICE interaction mode. SDP information will complete the exchange of SDP description information between two endpoints through "offer" and "answer" signaling, while the ICE framework can transmit the address of the ICE server through RTCPeerConnection when it is founded. Through the construction of the key signaling server, the audio-video real-time two-way communication function of the remote online trial mediation system is realized.

For the development of the whole system, Linux is the operating system, CentOS is the version, Java is the basic development environment, JDK version 1.8.0_91 is the development kit, Apache 2.4 is the Web server, IntelliJ IDEA is the Java integrated development tool, and MySQL is the database. After the above software systems are installed and configured one by one, the construction of Web application development environment is completed. This system uses Maven 3 to manage the project structure, divides the whole project into several engineering modules, and completes the design and development of the whole system based on Spring MVC architecture. Through the introduction of the above key technical theories, the overall environment of the system development, the configuration of related software and tools are determined, and the technical feasibility of the overall project of the remote online trial mediation system is also clarified.

3 DETAILED FUNCTION IMPLEMENTATION

The system supports users in two different roles, namely, court judge and case party, to complete account registration and identity verification by submitting information, and to log in and use the system with unique identification information. In order to improve the security of use, the system will encrypt the user password with hash algorithm, which will be used as authentication method to complete the user login authentication. The key code of implementing RSA encryption algorithm in Java language is shown in Figure 4. (Li, 2020)

```
public class RSAUtils
        KeyPairGenerator kpg;
        try {kpg = KeyPairGenerator.getInstance(RSA_ALGORITHM);
        } catch(NoSuchAlgorithmException e)
        {throw new IllegalArgumentException("No such algorithm-->[" + RSA_ALGORITHM
+ "]");}
        kpg.initialize(keySize);
        KeyPair keyPair = kpg.generateKeyPair();
        Key publicKey = keyPair.getPublic();
        String publicKeyStr = Base64.encodeBase64URLSafeString(publicKey.getEncoded());
        Key privateKey = keyPair.getPrivate();
        String privateKeyStr = Base64.encodeBase64URLSafeString(privateKey.getEncoded());
        Map<String, String> keyPairMap = new HashMap<String, String>();
        keyPairMap.put("publicKey", publicKeyStr);
        keyPairMap.put("privateKey", privateKeyStr);
        return keyPairMap;
```

Figure 4: Key code of RSA encryption algorithm implementation

3.1 The Parties to the Case

3.1.1 Personal Authentication

The parties to a case must undergo identity verification before conducting court mediation. Under this function module, users can upload photos or scanned documents of valid certificates to the system and submit them for approval. After the approval, they can wait for court mediation in the system.

3.1.2 Video Communication

Under this function module, after authentication, the parties (both parties) of the case will enter a specific function interface, and automatically start the camera or microphone under the terminal devices such as computers, tablets and mobile phones for real-time audio and video communication. The content of the video communication will be based on the actual situation of the case and the real demands of the parties, and the court trial investigation and mediation will be conducted according to the prescribed procedures.

3.1.3 Instant Communication

At the same time of video communication, the parties to the case (both parties) can use different forms of information content to express or transmit part of the content in the process of trial investigation or trial mediation. The system supports a variety of information interfaces, and images, texts, video and audio, and computer screens can be uploaded and displayed. At the same time, instant communication also supports the function of viewing and clearing historical news, providing a multi-dimensional communication channel for court investigation and court mediation.

3.1.4 Confirm of Conciliation Statement

Under this function module, the parties to the case (both parties) can obtain the electronic version of the mediation agreement, and support online reading and electronic signature. After the parties (both parties) confirm the case, they will submit it to the system for retention, which will have legal effect. If one party disagrees, the system will withdraw the mediation agreement and return it to the people's court for further trial or mediation.

3.2 Judge Users

After logging in to the system, the legal official users will import the corresponding cases into the system according to the application time of court mediation issued by the parties to the case (both parties), and conduct court mediation according to the scheduled time. The judge will review the identity information of the parties (both parties) in the case, and after the review is correct, the system will establish WebRTC multi-party audio-video conversation. As the core of trial investigation and trial mediation, the judge presides over and organizes the whole mediation process according to law. After the parties to the case (both parties) reach a mediation agreement, synchronize the electronic text with the parties to the case and complete the whole mediation process. In addition, the judge users can export and save all the contents in the video communication and timely communication of the case, so as to facilitate the on-demand trial of the case and the inquiry of the case file in the later stage.

4 CONCLUSIONS

This paper takes the trial mediation of civil cases as the research object, aims at promoting the reform of the new trial mediation mode, and builds a remote online trial mediation system with the help of the new generation of network information technology. The system will be based on Web technology, take WebRTC real-time audio-video communication technology as the core, and realize the diversified development of the trial mediation mechanism of the name court under the "internet plus Trial". It is conducive to breaking the limitation of time and space in the traditional court mediation mode, shortening the circulation cycle of cases, speeding up the efficiency of case handling, improving the level of judicial services, and making a beneficial attempt for the all-round construction of "smart courts" in the new era.

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