Design and Implementation of Protocol Conversion Server Based on GB/T 28181 and WebRTC

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Abstract. Aiming at the problem of interconnection between GB/T 28181-2016 "Technical requirements for information transport, switch and control in video surveillance network system for public security" (GB/T 28181) standard and Web Real-Time Communications (WebRTC) protocol, a protocol conversion server based on GB/T 28181 and WebRTC is designed and implemented. Firstly, based on the improvement of GB/T 28181 protocol stack and WebRTC protocol stack, an audio-video communication protocol stack integrating GB/T 28181 protocol and WebRTC protocol is designed, and the rules of communication between GB/T 28181 protocol and WebRTC protocol are agreed. Secondly, this paper analyzes and summarizes the interoperability of GB/T 28181 protocol and WebRTC protocol in terms of signaling method and interaction process. And on this basis, a proxy conversion server based on the signaling protocol of WebSocket transmission and the session initiation protocol (SIP) protocol of UDP/TCP transmission is designed and completed to realize the conversion function of GB/T 28181 signaling and WebRTC signaling. Finally, the feasibility and correctness of the method are verified through experiments, and the interoperability problem between GB/T 28181 protocol and WebRTC protocol is effectively solved.

Keywords: GB/T 28181 standard, WebRTC protocol, protocol conversion, protocol interoperability

1 Introduction

In recent years, with the release and promotion of the GB/T 28181 protocol standard, the security industry standardization of video surveillance networking has made significant development. GB/T 28181 standard provides a unified signaling interface, video and audio codec interface for the interconnection of video monitoring platform and equipment, simplifying the interconnection between different manufacturers docking work, effectively promoting the extensive sharing of video surveillance image resources [1].

At present, with the rapid development of 5G technology, real-time video and audio communication technology is constantly updated. WebRTC [2], as a real-time video and audio communication technology built on a web browser, provides a series

H. Wu et al. (Eds.): CSIEDE 2022, ACSR 103, pp. 800–808, 2022.
https://doi.org/10.2991/978-94-6463-108-1_89

of audio and video solutions such as audio and video capture, network transmission, audio and video codec, etc., which can realize cross-browser and cross-platform video and audio communication functions [3].

In this paper, we propose a scheme for the interoperability of GB/T 28181 and WebRTC. The first part of the paper mainly introduces the GB/T 28181 protocol and WebRTC protocol, including the protocol stack, transmission protocol and signaling methods; the second part of the paper is the design and implementation of the GB/T 28181 and WebRTC signaling protocol conversion module, respectively from the signaling method correspondence, interaction flow and other aspects; in the third part, the experimental environment is set up and the experimental results are analyzed, and we summarize the whole paper and point out the future work direction.

2 GB/T 28181 Protocol and WebRTC Protocol

The GB/T 28181 protocol is based on SIP [4] and specifies the interconnection structure of public security video surveillance networking system, the basic requirements of transmission, exchange, control and security requirements, as well as the technical requirements of control, transmission process and protocol interface, is the national standard in the field of video surveillance. The WebRTC is a technology that supports web browsers for real-time audio and video calls. In the bottom layer, audio engine, video engine and network transmission engine are provided respectively to realize reliable transmission of video and audio data [5].

2.1 GB/T 28181 Protocol Stack and WebRTC protocol Stack

The control signaling of GB/T 28181 protocol is based on SIP protocol, the media stream is transmitted by Real-time Transport Protocol (RTP) or Real-time Transport Control Protocol (RTCP), the video compression adopts H.264 coding algorithm, and the audio compression adopts G.711 and other common algorithms [6]. In the process of video and audio transmission and control, we need to establish two transmission channels: the session channel and the media stream channel. The session channel is used to establish a session between devices and transmit system control commands [7]; the media stream channel is used to transmit video and audio data. The protocol stack of GB/T 28181 is shown in Fig. 1.

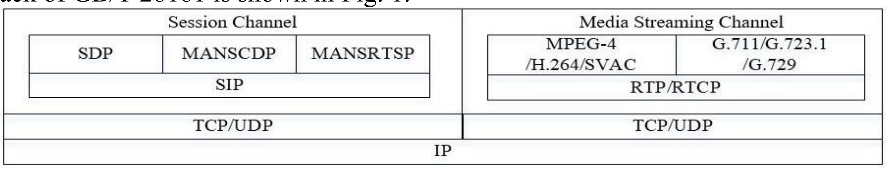

Fig. 1. GB/T 28181 protocol stack

According to WebRTC protocol, Secure Real-time Transport Protocol (SRTP) or Secure Real-time Transport Control Protocol (SRTCP) is adopted for streaming media data transmission, and UDP encrypted by Datagram Transport Layer Security

(DTLS) is used for transmission [8]; WebRTC supports interactive connectivity establishment (ICE) technology, Session Traversal Utilities for NAT (STUN) and Traversal Using Relay NAT (TURN) to establish and maintain UDP end-to-end connections; video compression in WebRTC protocol supports G.711, AAC and other encoding formats [9]. The WebRTC protocol stack is shown in Fig. 2.

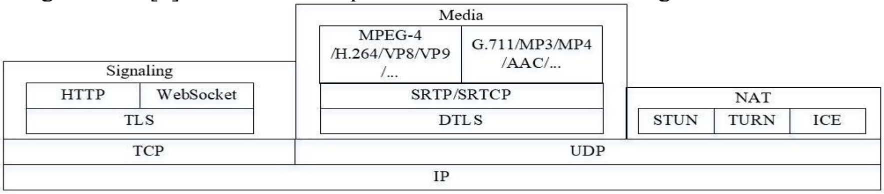

Fig. 2. WebRTC protocol stack

Based on the above two protocol stacks, an audio-video communication protocol stack based on GB/T 28181 standard and WebRTC protocol is improved and designed, as shown in Fig. 3.

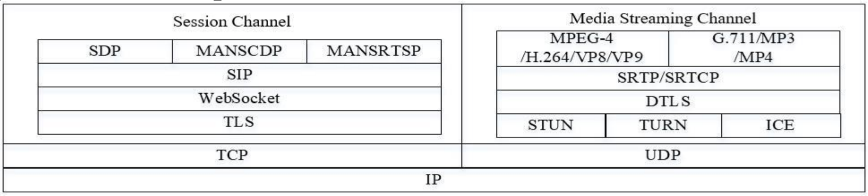

Fig. 3. Audio and video communication protocol stack based on GB/T 28181 and WebRTC

2.2 GB/T 28181 Signaling Methods and WebRTC Signaling Methods

The GB/T 28181 protocol and the WebRTC protocol have a lot in common, they belong to the application layer protocol, and more importantly, they have similar session control methods, and the protocol implementation usually uses the same underlying protocols [10]. The description of SIP signaling methods is shown in Table 1.

Table 1. GB/T 28181 Basic signaling method description.

Signaling method	Method description
REGISTER	It is used to register the UAC on the UAS and complete the address binding.
INVITE	It is used to initiate a session and invite users to join it.
ACK	It is used for replies to received INVITE requests.
UPDATE	It allows UAC to update the parameters of a session without changing the session state.
OPTIONS	It is used to query the server for supported methods, content types, encodings, etc.
CANCEL	It is used to cancel an outstanding conversation request.
BYE	It is used to end the session.

The signaling methods of WebRTC are mainly divided into two categories: the first category is the signaling sent by the client to the server, and the second category is the signaling sent by the server to the client [11]. WebRTC signaling methods and meanings are shown in Table 2.

Table 2. WebRTC basic signaling method description.

Signaling method	Method description	Type
join	The user joins the room.	Client
leave	The user leaves the room.	Client
message	End-to-end exchange of information (offer/answer/candidate).	Client
joined	The user has joined the room.	Server
left	The user has left the room.	Server
other_joined	Another user has joined the room.	Server
bye	Another user has left the room.	Server
full	The room is full.	Server

3 Design and Implementation of SIP Proxy Conversion Server

The system mainly includes two modules, GB/T 28181 signaling module and SIP proxy conversion server module, as shown in Fig. 4. SIP signaling server is the GB/T 28181 signaling module, which mainly completes the interaction processing with GB/T 28181 device terminals and the registration authentication of GB/T 28181 device terminals. The WebRTC signaling server mainly completes the interaction with the WebRTC client and the registration and authentication of the WebRTC client, while the signaling conversion module realizes the conversion function between GB/T 28181 signaling and WebRTC signaling.

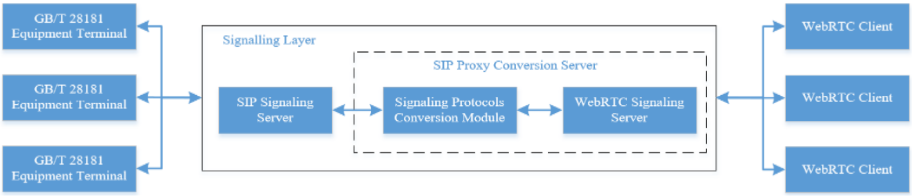

Fig. 4. System structure schematic

3.1 Signaling Protocol Conversion Module

The signaling conversion module should have the conversion function of GB/T 28181 signaling and WebRTC signaling to complete the two-way interaction between the GB/T 28181 equipment terminal and the WebRTC client. According to the conversion direction of signaling, the module can be divided into two parts:

SIP-WebSocket Direction Signaling Conversion. It mainly receives the SIP request message sent by the GB/T 28181 equipment terminal, converts it into the corresponding SIP over WebSocket protocol request message, and sends it to the WebRTC client; it also receives the response message returned by the WebRTC client through the SIP over WebSocket protocol, converts it into the corresponding GB/T 28181 standard SIP response message, and sends it to the GB/T 28181 device terminal.

WebSocket-SIP Direction Signaling Conversion. It mainly receives SIP over WebSocket protocol request message sent by the WebRTC client, converts it into corresponding GB/T 28181 standard SIP request messages, and sends it to the GB/T 28181 device terminal; it also receives the response message returned by the GB/T 28181 device terminal through GB/T 28181 standard SIP protocol, converts it into the corresponding SIP over WebSocket protocol response message, and sends it to the WebRTC client.

3.2 Signaling Interaction Flow

This paper takes real-time video and audio on-demand as an example and designs the interaction flow between GB/T 28181 SIP signaling and WebRTC signaling as shown in Fig. 5. Among them, the SIP proxy conversion server includes the WebRTC signaling server and signaling protocol conversion module.

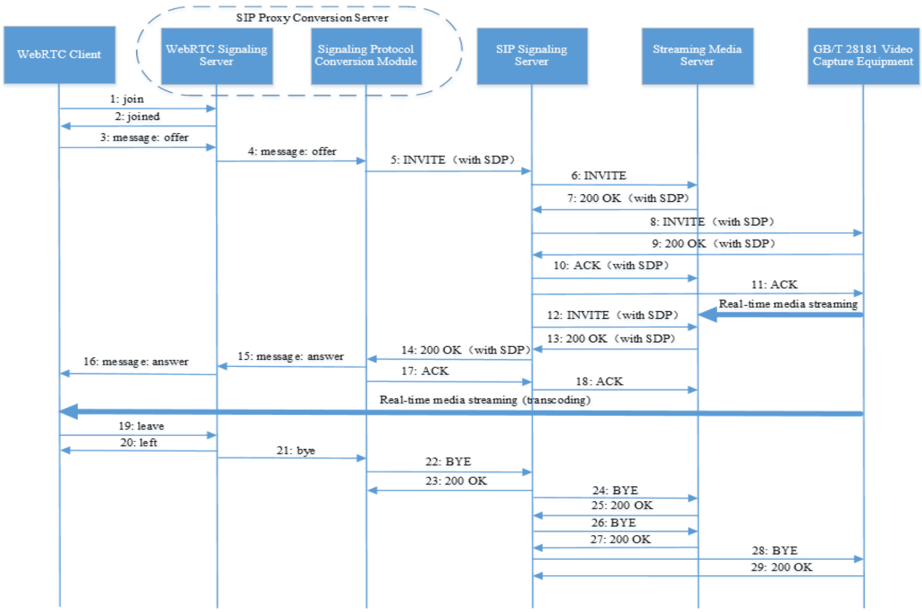

Fig. 5. GB/T 28181 signaling and WebRTC signaling interaction flow

The specific steps of its signaling interaction process are as follows:

- The WebRTC client sends a join message to the WebRTC signaling server in the SIP proxy conversion server, which carries the IP address of the WebRTC client and the port number on which the client receives the video and audio streams, and the WebRTC signaling server in the SIP proxy conversion server receives the request and returns the joined message;
- The WebRTC client sends an offer message to the WebRTC signaling server in the SIP proxy conversion server, and the WebRTC signaling server in the SIP proxy conversion server receives the request and forwards it to the SIP signaling server by converting it into an INVITE message through the signaling protocol conversion module;
- Messages 6 to 13 establish a three-way media session between the SIP signaling server, the streaming server and the GB/T 28181 video capture terminal, and the session negotiation mainly includes the port on which the streaming server receives and forwards media streams and the port on which the GB/T 28181 video capture terminal sends media streams;
- The SIP proxy conversion server receives the 200 OK response message forwarded by the SIP signaling server and forwards it to the WebRTC client through the signaling protocol conversion module into an answer message;
- The streaming server receives the message 18 ACK and then transcodes the received live media stream and forwards it to the WebRTC client;
- If the media stream reception is completed, the WebRTC client sends a leave message to the WebRTC signaling server in the SIP proxy conversion server, and the WebRTC signaling server in the SIP proxy conversion server returns a left message after receiving the message and converts the bye message sent by the WebRTC signaling server into a BYE message through the signaling protocol conversion module. The messages 24 to 29 are the signaling process of session connection disconnection between the SIP signaling server, streaming media server and GB/T 28181 video capture terminal, so that the signaling interaction process is completed.

4 Experiment

In this paper, we use the existing NodeJS to implement the WebRTC signaling server and add the signaling protocol conversion module to get the SIP proxy conversion server by secondary development, which realizes the conversion from the signaling protocol based on WebSocket transmission to the SIP protocol based on UDP/TCP transmission and completes the interoperability of SIP session messages. With the help of the streaming media server ZLMediaKit, a real-time video monitoring test environment is built using GB28181.WinTool test program of the GB28181.Solution project as a SIP signaling server. In the LAN environment, the WebRTC client requests access to the video camera supporting GB/T 28181 standard through SIP over WebSocket protocol, and successfully obtained the live images captured by the camera, as shown in Fig. 6. The real-time images obtained after several observation experiments do not have significant delays and have good real-time performance.

Fig. 6. PC Chrome previews GB/T 28181 video camera surveillance screen

This experiment captures the call signaling and response signaling of SIP proxy conversion server after conversion by WireShark tool, as shown in Fig. 7 and Fig. 8. After several tests and analyses, the feasibility of the SIP proxy conversion server signaling conversion function is verified, which can effectively solve the interoperability problem between GB/T 28181 protocol and WebRTC protocol.

```
Session Initiation Protocol (INVITE)
  Request-Line: INVITE sip:34020000001320000033@192.168.1.101:5060 SIP/2.0
  Message Header
  Message Body
    Session Description Protocol
      Session Description Protocol Version (v): 0
      Owner/Creator, Session Id (o): 34020000001320000033 0 0 IN IP4 192.168.1.100
      Session Name (s): Play
      Connection Information (c): IN IP4 192.168.1.100
      Time Description, active time (t): 0 0
      Media Description, name and address (m): video 30004 RTP/AVP 96 97 98 99
      Media Attribute (a): recvonly
      Media Attribute (a): rtpmap:96 PS/90000
      Media Attribute (a): rtpmap:98 H264/90000
      Media Attribute (a): rtpmap:97 MPEG4/90000
      Media Attribute (a): rtpmap:99 H265/90000
      Unknown: y=0200004418
      [Generated Call-ID: d0225e8c13dc65d3d9876625fbfd8129@0.0.0.0]
```

Fig. 7. Call request signaling transformed by SIP proxy conversion server

```
Session Initiation Protocol (200)
  Status-Line: SIP/2.0 200 OK
  Message Header
  Message Body
    Session Description Protocol
      Session Description Protocol Version (v): 0
      Owner/Creator, Session Id (o): 34020000002000000001 591 591 IN IP4 192.168.1.101
      Session Name (s): Play
      Connection Information (c): IN IP4 192.168.1.101
      Time Description, active time (t): 0 0
      Media Description, name and address (m): video 15060 RTP/AVP 96
      Media Attribute (a): setup:active
      Media Attribute (a): sendonly
      Media Attribute (a): rtpmap:96 PS/90000
      Media Attribute (a): username:34020000002000000001
      Media Attribute (a): password:12345678
      Media Attribute (a): filesize:0
      Unknown: y=0200004418
      Unknown: f=
      [Generated Call-ID: d0225e8c13dc65d3d9876625fbfd8129@0.0.0.0]
```

Fig. 8. Response signaling transformed by SIP proxy conversion server

5 Conclusion

In this paper, we propose a scheme for GB/T 28181 and WebRTC signaling conversion to address the problem of interoperability between GB/T 28181 and WebRTC protocols. Through the process of WebRTC client accessing GB/T 28181 media device, the paper verifies the possibility of interconnection between WebRTC client and GB/T 28181 device terminal from one aspect, realizes the transmission of media stream from GB/T 28181 terminal to WebRTC client direction. It effectively solves the interoperability problem between GB/T 28181 protocol and WebRTC protocol.

Acknowledgements

I would like to thank my supervisor, Mr. Shuming Jiang, for his valuable guidance throughout my studies to bring my work to a higher level. This work was supported by the National Key Research and Development Program of China (2019YFB1404700).

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