Design of Urban Emergency Response System Based on WebRTC and IMS

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Abstract. By integrating WebRTC technology in HTML5 with IP Multimedia Subsystem (IMS) technology in a mobile network, we have developed a new Urban Integrated Emergency Response System (UIERS), which is more suitable for a modernized city. This system can be widely accessed by a mobile intelligent terminal, such as a smartphone, tablet, PDA, or traditional SIP terminal. Users can log in via a web browser and call up users who are in the IMS/PSTN network directly. Without installing applications or loading a plug-in, this system is much more applicable in different platforms and is more easily upgraded and maintained in Browser/Server (B/S) structure mode.

Keyword: urban Integrated Emergency Response System, IMS-WebRTC gateway, WebRTC, IMS

1 Introduction

An Urban Integrated Emergency Response System (UIERS) is a set of information systems integrating with communication, command, dispatch, and position. Without changing the administrative establishment of the emergency department of each section, the UIERS can achieve joint enforcement among city emergency sectors and increase the quality of emergency services.

Public emergency events and all kinds of disasters happen daily in the world, such as crises, exigencies, fires, floods, earthquakes, natural disasters, etc. Some of the most impacting events for people included the "9/11" terrorist attack in NYC in 2001, the worldwide H1N1 flu in 2009, and the violent earthquake in Wenchuan, China in 2008, which remain fresh in our memories. In terms of urban emergency response, the proportion of the urban population in China is over 50%, and cities centralize the majority of industries, services, education, and research. These emergencies are not only threats to people's safety and property, but they also damage the economy and society. To deal with these events, emergency management needs every sector to cooperate with each other. In the meantime, the efficiency of emergency responses decides the degree of damage and influence. Every city administrator has to face the problem of how to build a quick response, multi-sectoral cooperation, and a highly efficient emergency system. Nowadays, the major cities in China have built a mature emergency system, but it has not spread to the middle- and small-sized cities [1].

Since the 1960s, many countries in the world have established UIERS, integrating the management of police, fire, medical care, and other sectors, and they have guaranteed the crews' rescue and security, so other emergency sectors can cooperate closely and dispatch the field personnel quickly. In 1967, the U.S. recommended the creation of a single telephone number that could be used nationwide for reporting emergencies and then established 9-1-1 as the emergency number. In 2000, the European Union stipulated a unified emergency number 1-1-2 in each member country [2].

From the domestic development of the emergency system, Hong Kong has implemented a unified alarm number 9-9-9. The city of Nanning has built a UIERS commanded by the government directly, and this system, which combines police, fire, medical care, and other emergency sectors, has been developed in line with international standards. Nowadays, many major cities in China have built their own UIERS.

The processing of public emergency events in a city is a complicated project, as it requires response team crews to have command ability, cooperative ability, and professionalism in their own field, as well as the participation of various departments in the city. During processing, the team needs to 'respond quickly, process effectively, and communicate fully.' Today, many administrative functions have been assigned to urban departments in great detail, which leads to the need for departmental cooperation to solve emergency events. For instance, in the face of illegal assembly, people need the police to maintain order, the government to negotiate, medical care to rescue casualties, the civil administration to guarantee the normal regulation of life, and even the army to protect people's lives and property. Moreover, the leaders of every relative department will constitute a response team to communicate with each department and to deploy relative staffs more conveniently [3].

The team needs to obtain real-time feedback of a situation from the field personnel's equipment in the process of deployment. However, in reality, the old analog interphone they use cannot satisfy the requirement of processing an emergency, while the new type of digital interphone has yet to be equipped in all departments on a large scale. Emergencies are usually solved by a response team, which is made up of the elites from various departments, and orders can only be given through voice by analog interphone or cellphone. In this situation, orders cannot be delivered to every filed personnel quickly and clearly. When using an analog interphone, the quality of voice is not very good and it is prone to be taped. Besides, the communication range of an analog interphone is very limited and cannot cover the whole city or even a district of a city, which cannot satisfy the needs of the management of the whole city. Thus, the new type of digital interphone seems to be much more suitable for the management of modern cities. However, the high price limits the large-scale deployment of this digital interphone, and the current domestic and foreign digital interphone can only support audio rather than audio-visual business.

Generally, custom digital terminals cannot be deployed to a large number of field personnel. However, with the popularization of mobile communication technology, everyone has a smartphone or tablet now. Therefore, in utilizing this equipment, everyone can offer live multimedia information to the emergency response team and respond to orders and unexpected events throughout the system. A mobile intelligent terminal, which allows access to the system via a web browser with personal account, can make itself into a digital interphone.

Utilizing the IP Multimedia Systems (IMS) and multimedia communication technology based on WebRTC, our system allows field personnel to log in by using a mobile phone, tablet, or any other intelligent terminal and provides multimedia services, such as audio-visual, subscription, and message services. Consequently, our system will be highly flexible and adjustable to expand and implement new types of media and services by using the IMS framework.

The rest of this paper is structured as follows: section 2 introduces the technology used in this new UIERS, section III covers the detailed information of the framework of this new UIERS, and section 4 specifies the significant element: the IMS-WebRTC Convert Gateway. The realization of this system is presented in section 5 and finally, we summarize in section 6.

2 Related Works

2.1 IMS

IMS is an architectural framework for delivering IP multimedia services and it was originally designed by the wireless standards body 3GPP to support a mobile and fixed access network.

To ease the integration with the Internet, IMS uses IETF protocols and Internet technology wherever possible. The framework based on SIP can provide QoS, which IP cannot do, and it has the ability to merge various kinds of integrated services. The horizontal control layer of IMS separates the access network layer from the service layer [4]. From a logical architecture perspective, services do not need to have their own control functions, as the control layer is a common horizontal layer [5].

Using IMS, telecom operators can enter the field of mobile Internet cost-effectively, while mobile operators can introduce new and rich multimedia businesses easily, without interfering with the original voice and SMS services. In this case, IMS solves the interoperability problem between IP telephony (e.g. VoIP) and a mobile core network [6].

2.2 WebRTC

In May 2011, Google released an open-source project for browser-based real-time communication known as WebRTC. WebRTC is a real-time multimedia communication technology that supports browser-to-browser applications without depending on either internal or external plug-ins. By using the HTML label and JavaScript API, WebRTC allows developers to develop applications in a short development cycle. WebRTC aims to build a web browser into a common platform that exchanges real-time multimedia data among all user equipment (e.g., mobile intelligent phone, tablet, and PC) [7]–[9].

Major components of WebRTC [10]:

- getUserMedia - allows a web browser to access the camera and microphone and to capture media.
- RTCPeerConnection - sets up audio/video calls.
- RTCDataChannel - allows browsers to share data via peer-to-peer [11].
- getStats - allows the web application to retrieve a set of statistics about WebRTC sessions.

2.3 Development of Domestic and International VoIP Technology

Nowadays, European and North American mobile operators have almost completed the updates to the IMS network. In consequence, VoIP applications have been used widely, such as Skype, Rebtel, and Localphone. Domestic mobile operators are going through a smooth transition period of IMS upgrades. Many VoIP applications have emerged, such as Alicall, KC, and Weixin. The methods of implementing VoIP business and a comparison of each are shown in Table 1.

Table 1. The implementation and comparison of VoIP

Implementation Methods	*Convenience*	*Scalability*	*Need to install apps or plug-ins?*	*Cross-platform compatibility*	
Application running in mobile phone and computer	★★★☆☆	★★☆☆☆	Yes	Need customization for each platform	★★★☆☆
Flash Application running in web browser	★★★☆☆	★★☆☆☆	Yes	Need Flash support	★★★☆☆
Plug-in application running in web browser	★★★☆☆	★★☆☆☆	Yes	Need web browser installed plug-in	★★★☆☆
Web page callback	★★★★★	★★☆☆☆	No	-	★★★★★
Caller dials system access number	★★★★★	★☆☆☆☆	No	-	★★★★★
Caller dials a special number assigned to a callee by system	★★★★★	★☆☆☆☆	No	-	★★★★★
WebRTC applications running in web browser	★★★★★	★★★★★	No	Need web browser to support WebRTC	★★★★☆

As the above table shows, WebRTC is more prominent than other implementation methods in convenience, business scalability, and cross-platform capability. In terms of convenience, users only need a browser that supports WebRTC function without installing any other application or plug-in. Regarding business scalability, WebRTC can shorten the development cycle of new business. Regarding cross-platform capability, WebRTC runs in a web browser, so there is no need to develop various versions for various platforms.

For these reasons, opting for WebRTC for multimedia services on a website is a convenient and multi-platform option.

3 Framework of Urban Integrated Emergency Response System

3.1 System Framework

As shown in Fig. 1, the system is divided into the following modules:

1. Application and Dispatch Platform. One or more centralized dispatch agents (PC-DC) can configure the basic information of a system according to the leaders of emergency dispatch. According to present business, there are two kinds of application servers.

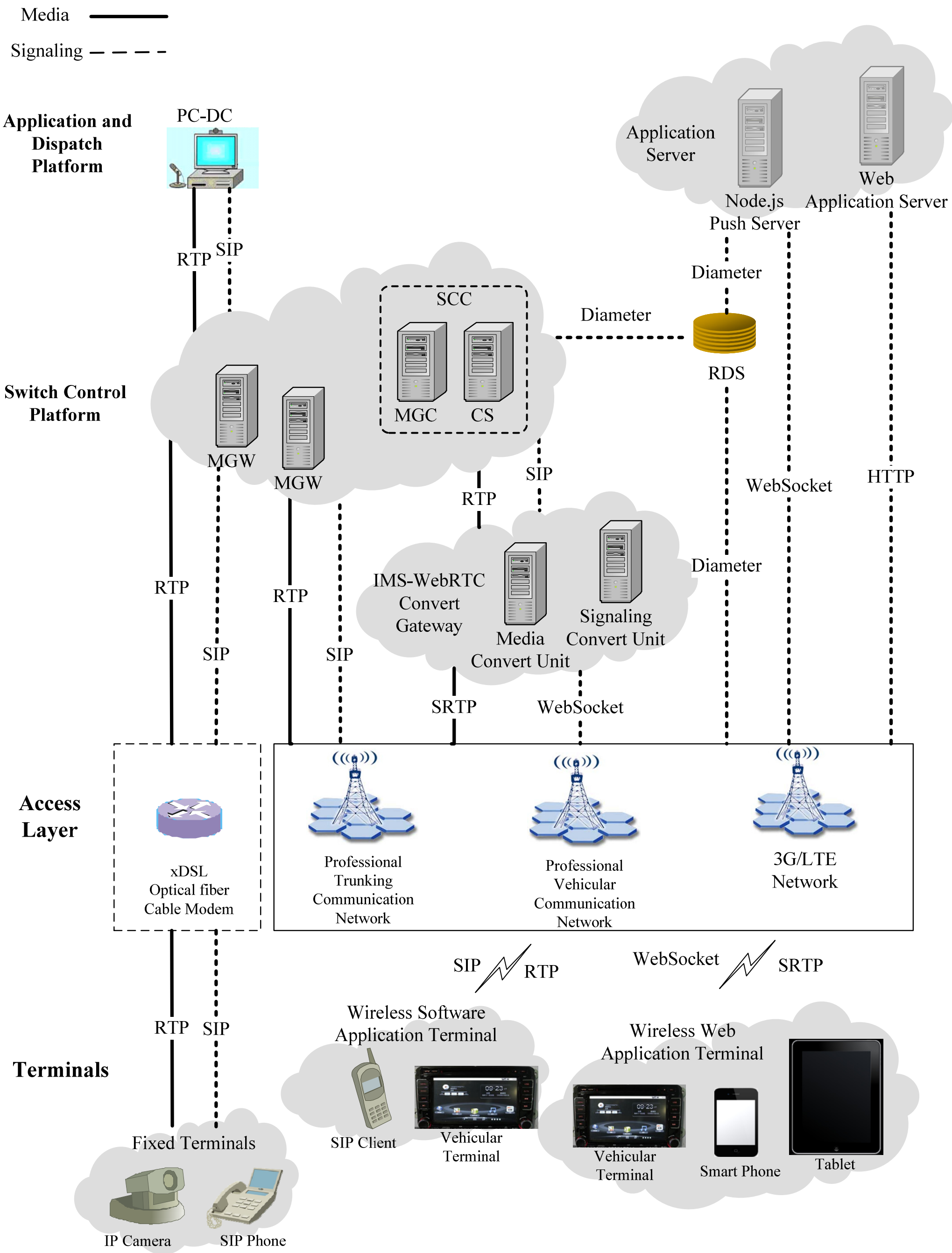

Fig. 1. The system framework

(1) Node.js Push Server. The primary function is to present some information about an organization, as listed for a Web-DC client. Group membership, police records, and field personnel's location can be inquired by restAPI.

Node.js is an open-source, cross-platform runtime environment for server-side and networking applications that are written in JavaScript. Node.js provides an event-driven architecture and a non-blocking I/O API that optimizes an application's throughput and scalability. These technologies are commonly used for real-time web applications [12]. In this system, the Node.js Push Server asks what the clients need from the RDS Server and sends it to the client.

(2) Web Application Server. The primary function of the server is to push web pages to the Web-DC client and to receive requests from clients. By using a web application based on WebRTC, users need not install any software, which is more efficient and convenient for the upgrade of business and services in the future.

2. Switch Control Platform.

(1) Media Gateway (MGW). The primary function is to replay the media stream of each call to a given destinations and convert the format of the media stream for each client user.

(2) Switch Control Center (SCC) is a control center of calls and media in a system, and a SCC includes a Control Server (CS) and Media Gateway Controller (MGC).

a) The CS is the center of call session control in a system. Its primary function is to process call session control, handle signaling related to call session control, and conduct some extensional business. In this system, CS develops in the base of the reSIProcate protocol stack [13].

b) MGC. Its primary function is to control many MGWs to handle media streams, such as the creation, modification, and deletion of a stream. MGC communicates with MGWs in the Media Gateway Control Protocol (MGCP) [14]; [15].

(3) The Relational Database Service (RDS) Server achieves the goal of receiving, storing, and processing the user data by means of the operation of the MySQL database. Diameter protocol is used for communication between CS and the RDS server.

(4) IMS-WebRTC converting gateway. This is the most important element for achieving communication between Web-DC and a traditional SIP client. It consists of the Media Convert Unit and Signaling Convert Unit.

a) Media Convert Unit. Separate from the MGW in SCC, its main function is to decrypt the media stream encrypted in SRTP by WebRTC and then convert VP8, the standard media codec in WebRTC, to H.264 or MPEG-4, which are generally used in a SIP client.

b) Signaling Convert Unit. Its primary function is to encapsulate SIP signaling into WebSocket protocol and de-encapsulate SIP signaling from WebSocket protocol.

3. Access layer platform. This system has the ability to connect to wired or wireless terminals, and subsystems, such as a trunking subsystem, vehicle-mounted subsystem, 3G/4G, and other subsystems accessed by a wired line or wirelessly.
4. Terminals. In the system, users can use traditional wired fixed terminals, such as IP camera and SIP phone, wireless SIP software terminals, and Web-DC clients based on WebRTC. It is suitable for wired fixed terminals to use for duty and road monitoring, while wireless SIP terminals are applied to mobile dispatch, as well as equipped to field personnel and vehicles. Web-DC is used in mobile dispatch; without installing any application or plug-in, a user can log in to a system via a smartphone, tablet, or any common intelligent equipment, which could increase the number of users.

3.2 System Function and Service

The system provides the following services:
1. Audio-visual single call, which is used for a client-to-client communication between both ends.
2. Group call among emergency response team with audio. Leaders can establish a temporary or default talking group. The crew applies for the voice via sending a SIP INFO message when in need of in-group communication. According to the priority of the crew or the permissions of leaders, the crew can take over the voice capabilities so that the others can hear their sound.
3. Group call to emergency response team with audio and video. In the base of last service, leaders can see the live video image of the crew. Besides, leaders can share the video image with someone specific or those in the group. It is more intuitive and convenient to share real-time information with everyone.
4. A query of crew information. Every group member can obtain the crew's information through inquiry (such as real-time location, personal information, etc.).
5. Video monitoring. Leaders can request DC to display several IP cameras' images.
6. Multimedia communication information encryption. WebRTC uses SRTP to encrypt the media stream, so it can protect media information from man-made sabotage, eavesdropping, and modification. The security of Web-DC has a great guarantee.

In this paper, we focus on the IMS-WebRTC convert gateway, allowing IMS framework and WebRTC technology to correspond with urban emergency application scenarios. What's more, on the client side, we propose a structure mixing B/S with C/S, which aids in system flexibility.

4 IMS-WebRTC Convert Gateway

In this system, the IMS-WebRTC convert gateway is an important element and its function is to merge Web-DC based on WebRTC into the traditional IMS network framework, which will allow the types of ends tp be more diverse, the use of ends to be more convenient, and the development of new business to be faster.

4.1 The Design and Implementation of the Signaling Convert Unit

To achieve intercommunication between Web-DC based on WebRTC and the traditional SIP client in an IMS network, the mapping of signaling conversion is the first issue to face. WebRTC uses JavaScript Session Establishment Protocol (JSEP) as the API of signaling [16]; thus, the developer can choose one of the protocols (SIP, ROAP, XMPP, etc.). In this paper, we choose SIP for session control signaling. Web-DC based on WebRTC uses WebSocket to carry SIP, while a traditional SIP client in an IMS network uses TCP or UDP to carry SIP. Therefore, the Signaling Convert Unit must change the carrier of the SIP message, modify the body of the SDP, and control the Media Convert Unit [17]; [18].

In the implementation of a Signaling Convert Unit, we use the libwebsocket open-source library to pack and unpack the WebSocket packet. As shown in Fig. 2, according to the type of received SIP message, it will send a corresponding command to the Media Convert Unit and modify the body of SDP.

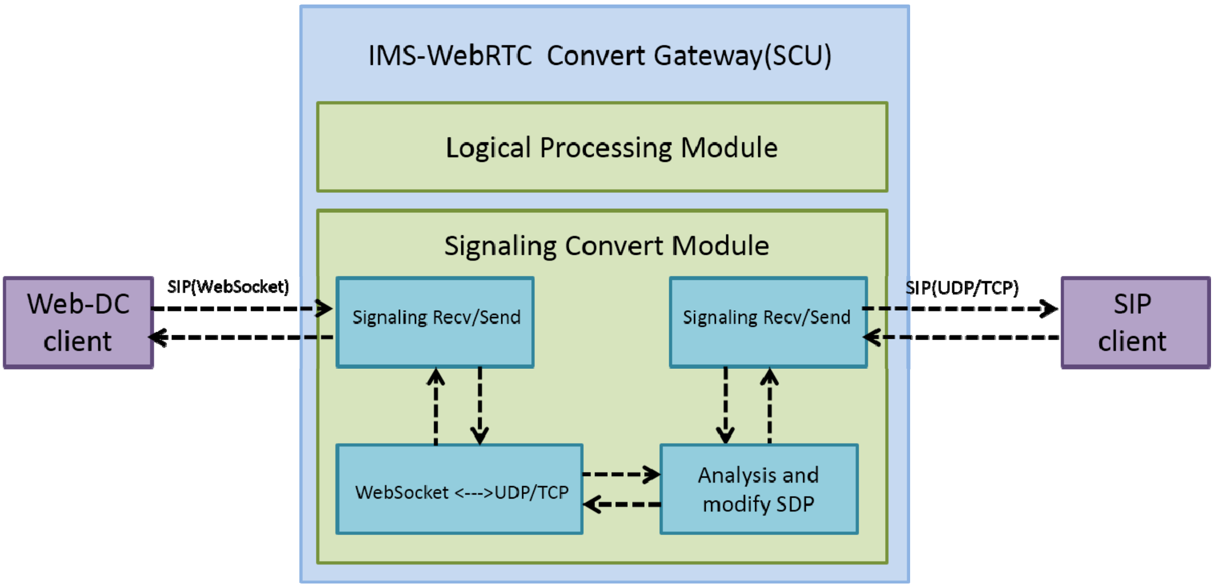

Fig. 2. The SCU framework

4.2 The Design and Implementation of a Media Convert Unit

In a WebRTC client, SRTP [19] instead of RTP becomes the first choice of media transport protocol for maintaining the security of a multimedia session, making use of the VP8 video codec and the iLBC/G.711/iSAC audio codec [20]. In an IMS client, however, RTP is the only choice with the H.264/MPEG-4 video codec and the G.711/G.729/AMR audio codec. It would appear from the above that the only commonality is G.711 audio codec, so the major mission of the Media Convert Unit is to convert media transport protocol and video codec.

As shown in Fig. 3, the Media Convert Unit framework is made up of three modules, which are logical processing, session management, and media processing.

The logical processing module is in charge of interaction with the Signaling Convert Unit, handling commands from the SCU.

The session management module manages each multimedia session. The properties of each session, like status, IP address and ports of destination, the secret key of SRTP, and others, are stored in an object named MediaSession, and all the objects named MediaSession are saved in a list.

The media processing module manages the transport of streams belonging to a session. According to the different types of sessions, there will be one or more streams.

In the implementation of the Media Convert Unit, the functions of media transport are based on the Live555 open-source library, achieving the transmission, pack, and unpack of RTP packets that carry VP8/H.264/MPEG-4 video data. The function of conversion between SRTP and RTP is based on the libsrtp open-source library, achieving the encryption of RTP and de-encryption of SRTP. The function of encoding and decoding the media stream is based on the FFmpeg open-source library, achieving conversion among different video codec. The function of the ICE [21] is based on the libnice open-source library, achieving the network address translation between a private network address and public network address.

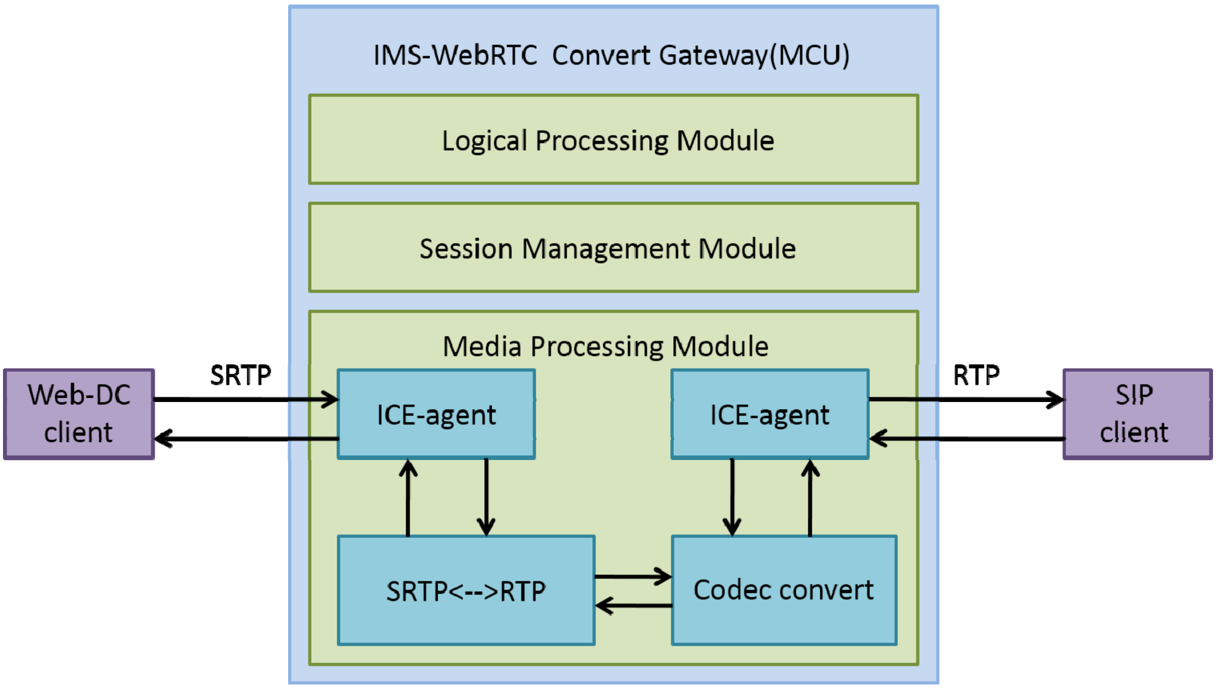

Fig. 3. The framework of an MCU

4.3 The Process of IMS-WebRTC Convert Gateway

Fig. 4 is an example of the Web-DC client launching an audio-visual single call to the SIP client to explain how these two units work.

1. The Signaling Process

(1) First, an INVITE request, encapsulated into the WebSocket packet from caller Web-DC, will be sent to the Signaling Convert Unit of the IMS-WebRTC convert gateway. The request message will carry SDP, which indicates the format that Web-DC supports, the secret key of SRTP, and the parameters used for NAT.

(2) SCU will take the SIP body out from the WebSocket packet when receiving the packet. After modifying the parameters of the SDP body, the SIP message loaded in TCP or UDP will be sent to CS in SCC. Meanwhile, it will announce to the Media Convert Unit to establish a multimedia session and save the connect information of Web-DC.

(3) When CS receives the INVITE request message, it announces to MGW to establish an audio-visual single call session and save information about Web-DC. Then, it will send the INVITE message to the SIP client.

(4) When the SIP client replies to the 200OK message to CS, CS will announce to MGW to save the information of the SIP client and send the message to SCU.

(5) SCU receives the 200OK and modifies some parameters of SDP, then encapsulates this SIP message into WebSocket, sending to Web-DC. After that, SCU announces to the MCU to save the connect information of MGW in SCC.

(6) After Web-DC receives the 200OK message, it will reply with an ACK message to SCU. Then, the SCU can announce to the MCU to start this session between MGW and Web-DC and send this message to CS in SCC.

(7) CS receives the ACK, transmits it to the SIP client, and then announces to MGW to start this session between the SIP client and the MCU of the IMS-WebRTC convert gateway.

2. The Media Process

(1) The MCU will de-encrypt SRTP to RTP when receiving the stream from Web-DC and then convert the VP8 video codec to video codec that the SIP client supports and send it to MGW in SCC.

(2) In the other direction, the MCU will convert the video codec that the SIP client supports to the VP8 video codec when receiving the RTP stream from MGW in SCC. It will then encrypt the RTP stream to SRTP and send it to Web-DC.

In Fig. 5, the string behind field ‘a=crypto:’ is the secret key of SRTP.

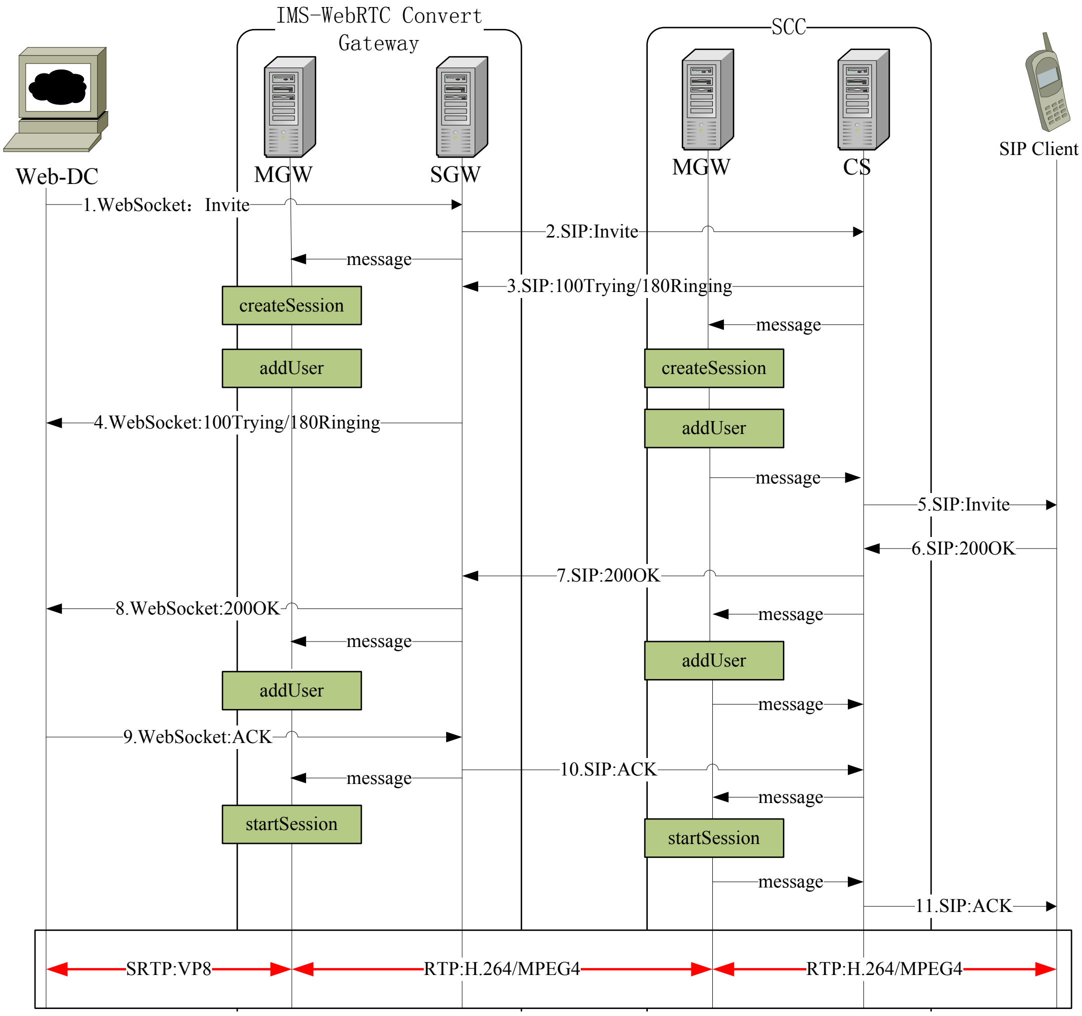

Fig. 4. The session establishment process

```
v=0
o=- 8232683135580877000 2 IN IP4 127.0.0.1
s=Doubango Telecom - chrome
a=group:BUNDLE audio video
a=msid-semantic: WMS qjsCpm0Eew6PNziEjJJzSnr0S8AkjBEhuAou
m=audio 2170 RTP/SAVPF 111 103 104 0 8 106 105 13 126
c=IN IP4 192.168.0.122
a=ice-ufrag:lkIJTJiEdTdheycw
a=ice-pwd:UNhxTpzXvtsLqSvY0ShIVu7T
a=fingerprint:sha-256
ED:CD:77:4D:18:ED:64:20:CD:C9:7C:53:1C:23:FB:87:8F:D9:89:1C:A1:FD:D3:44:30:2E:4C:BC:B1:64:66:93
a=crypto:1
AES_CM_128_HMAC_SHA1_80 inline:d0RmdmcmVCspeEc3QGZiNWpVLFJhQX1cfHAwJSoj|2^20
a=rtpmap:111 opus/48000/2
a=fmtp:111 minptime=10
a=rtpmap:103 ISAC/16000
a=rtpmap:104 ISAC/32000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
  …….
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Fig. 5. The secret key of SRTP in SDP

5 The Realization of the System

5.1 The business of audio-visual single call and video monitoring

After entering a personal user's name and password, we can log in to the system. The user's name we want to call or monitor is inputted, and then the image of each user will be displayed (Figs. 6, 7).

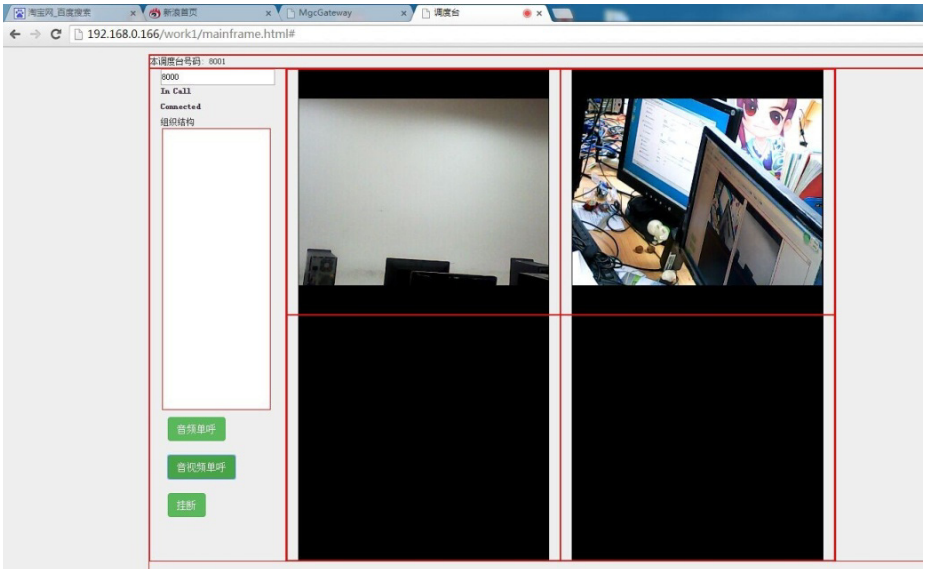

Fig. 6. Audio-visual single call

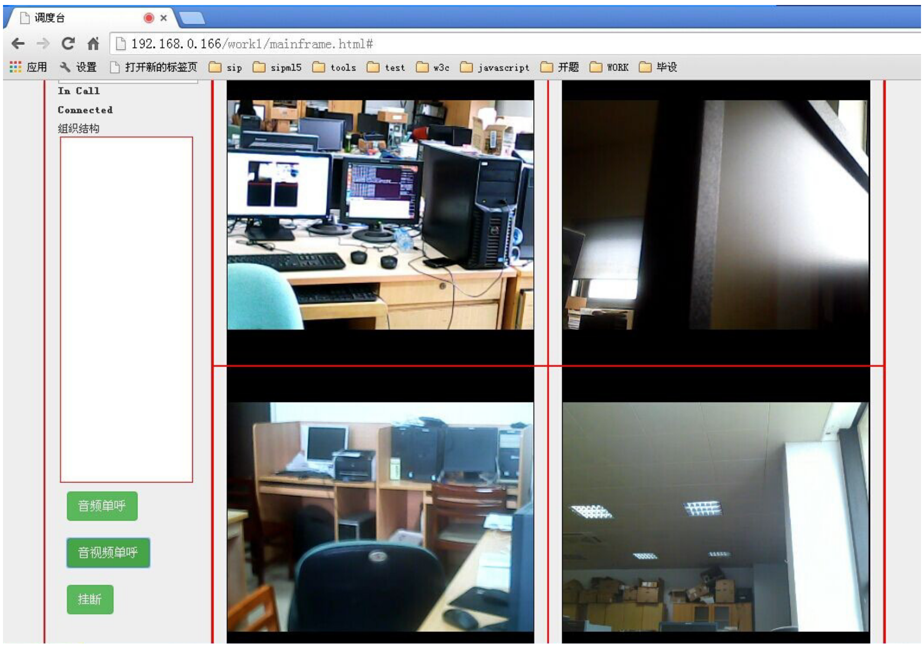

Fig. 7. Video monitoring

5.2 The query of crew's information from the Node.js Push Server

When logging into Web-DC successfully, the crew's information will be displayed (Fig. 8), then the other information needed can be shown elsewhere later.

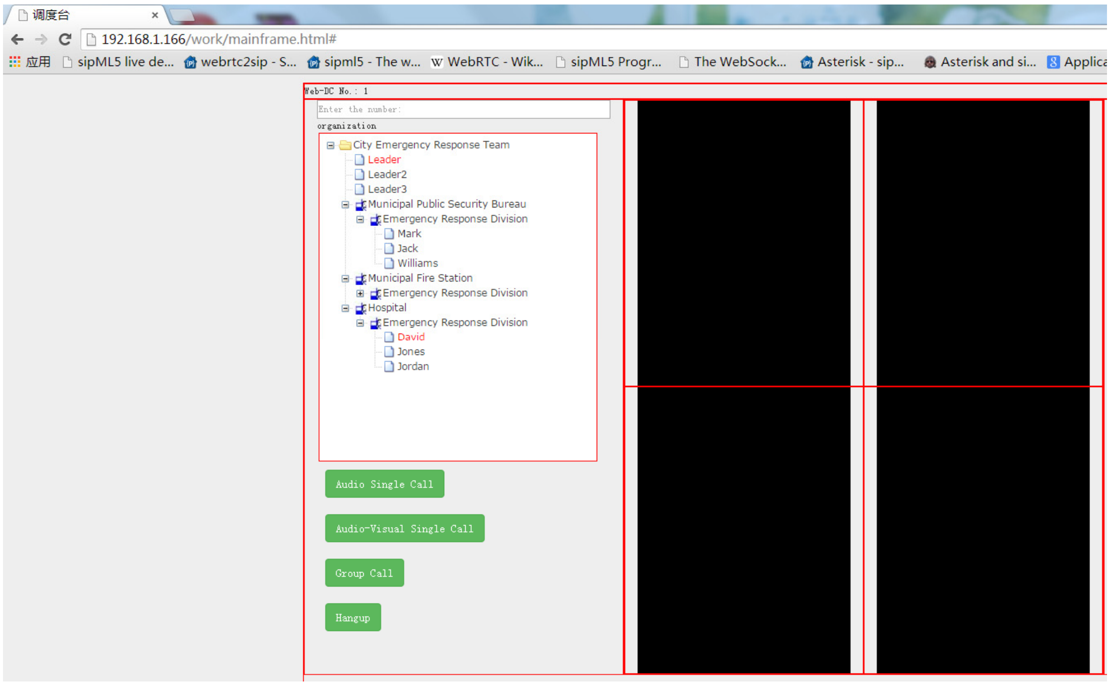

Fig. 8. The presence of the crew's information

6 Summary

By combining WebRTC technology with an IMS network, we could integrate telecom operators' IP networks with mobile operators' IMS networks and enlarge the group of users. The business of IMS will expand to other fields and have an important significance for the future integration of multi-networks, making use of the advantage of WebRTC, including the low cost of development and maintenance.

In this paper, we designed a UIERS, taking full advantage of the IMS network that supports multi-network access, provides a high guarantee of QoS, and has cross-platform flexibility. After some tests, it was shown that the system can meet the needs to support almost every intelligent terminal and access network. In the future, the system will be a strong supplement for dealing with public emergency events.

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