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Oscar Figuerola
Hari Kalva
Antonio Escudero
Ankur Agarwal

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Oscar Figuerola, Hari Kalva*, Antonio Escudero, Ankur Agarwal
Dept. of Computer Science and Engineering, Florida Atlantic University, Boca Raton, FL 33431

ABSTRACT

A large number of health-related applications are being developed using web infrastructure. Video is increasingly used in healthcare applications to enable communications between patients and care providers. We present a video conferencing system designed for healthcare applications. In face of network congestion, the system uses role-based adaptation to ensure seamless service. A new web technology, WebRTC, is used to enable seamless conferencing applications. We present the video conferencing application and demonstrate the usefulness of role based adaptation.

Keywords: video conferencing, adaptation, user roles, WebRTC

1. INTRODUCTION

Since the early days of digital video, there have been many trials and projects evaluating the use of video conferencing for delivering healthcare. One of the first trials using video conferencing for healthcare showed that use of video conferencing does not affect the quality of care [1]. There have been many trials since. Use of video conferencing for psychological health has had more success in adoption due to the nature of care provided [2]–[5]. Use of video conferencing for assisted living was reported in [6], [7]. We focus on a video conferencing application for assisted living facilities where it is desirable to have a guardian participate in the video conference between a patient at an assisted living facility and a doctor.

In spite of positive trials and documented benefits of using video conferencing in health care applications, wide deployment of video conferencing in healthcare has lagged primarily due to lack of communication infrastructure and complexity of implementing video conferencing solutions. Video conferencing requires high bandwidth availability and good quality video. The cost and complexity of developing realtime video conferencing systems is also high. New video compression technologies such as H.264/AVC and VP8 have made possible high quality video at relatively low bitrates. Broadband internet services have become the norm and these developments make wide use of video conferencing in healthcare possible. Reducing the cost of healthcare is another strong motivation for using video conferencing. Responding to industry needs and to take advantage of advances in Internet infrastructure, IETF and W3C are working on a standard for realtime video services using web browsers. The goal is to make video service development as simple as developing any web applications using JavaScript. This standard, known as WebRTC, has the potential to make realtime video services ubiquitous and healthcare applications can benefit greatly from these developments.

1.1 WebRTC

WebRTC is a web technology that enables web browsers with real-time communication (RTC) capabilities [8]. One of the key advantages over other technologies that provide the same functionality is its integration within the browser, and its ease of use through HTML5 and JavaScript APIs, which make specialized plugins unnecessary. WebRTC is aiming to become a standard, and is being developed by a joint effort between the IETF RTCWeb and W3C WebRTC working groups, which are defining the APIs and underlying protocols.

The protocols involved in the establishment of the peer to peer connection and to send the media data are shown in Figure 1. The media data is carried using the secure Real-time Transfer Protocol (SRTP), and the RTP Control Protocol (RTCP) is used to monitor transmission statistics related to the media stream. The architecture of WebRTC allows the media data to flow directly between browsers, but before any media data can be sent, the communication needs to be coordinated. The application exchanges media configuration information using the Session Description Protocol (SDP). This process is called signaling, and it is important to note that WebRTC does not specify the signaling mechanism, leaving it to the application layer. As opposed to the media path, the signaling path goes through an external server that can modify, translate, and manage the exchanged messages. The STUN (Session Traversal Utilities for NAT) protocol

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and its extension TURN (Traversal Using Relay NAT) are typically used to find out the public address and port, and traverse NAT boxes and firewalls so that clients can receive UDP-based media streams.

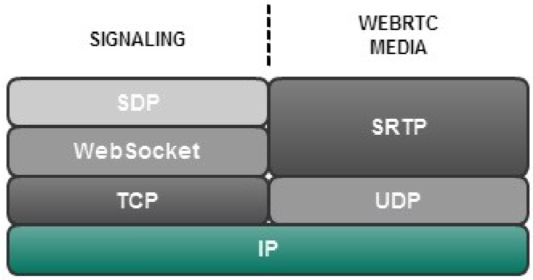


Figure 1. WebRTC's protocol stack

At its current implementation status, WebRTC allows to use the MediaStreams API to get audio and video from a webcam and a microphone, the PeerConnection API to establish a peer-to-peer link for voice and video, and the DataChannels API to share data through a PeerConnection. WebRTC support has been growing constantly since its inception, with Google Chrome, Mozilla Firefox, and Opera, both Desktop and Android versions, currently being the three main implementations, which can interoperate between them. On the other hand, WebSocket [9] is a signaling web technology that provides bi-directional communication channels over a single TCP connection. WebSocket is composed of an API being developed by W3C, and a protocol that has been standardized by IETF. This technology complements WebRTC, allowing to exchange the signaling messages needed to establish a peer-to-peer connection, and it can also be used to exchange messages between a browser client and a server, and vice versa.

2. HEALTHCARE VIDEOCONFERENCING SYSTEM

The system that has been developed uses the real-time communication capabilities enabled in the web browser by WebRTC [8]. This technology is based on HTML5 and Javascript, and allows establishing peer-to-peer communication links both directly between browsers, and between browsers and an intermediary media server.

As an abstraction layer, Licode [10] is being used to set up a Multipoint Control Unit (MCU). Licode is an Open Source platform that relies on WebRTC, and provides virtual room functionalities following a publisher/subscriber model. One of the main advantages of using Licode is that its APIs allow programming the application independently of WebRTC's implementation, which might vary over time, since it is still a standard under development. The basic architecture of Licode is shown in Figure 2. The code of the MCU is run in an intermediary server, which receives the video and audio streams of all the participants of the session.

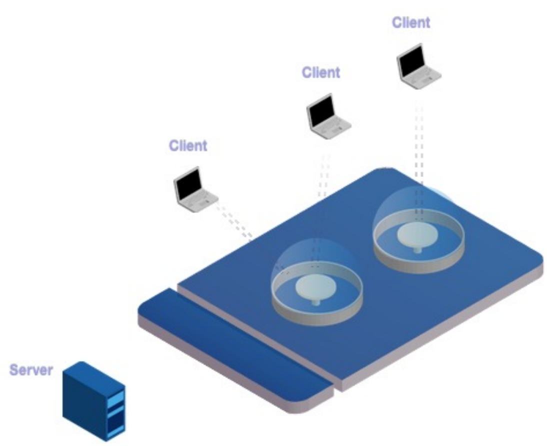
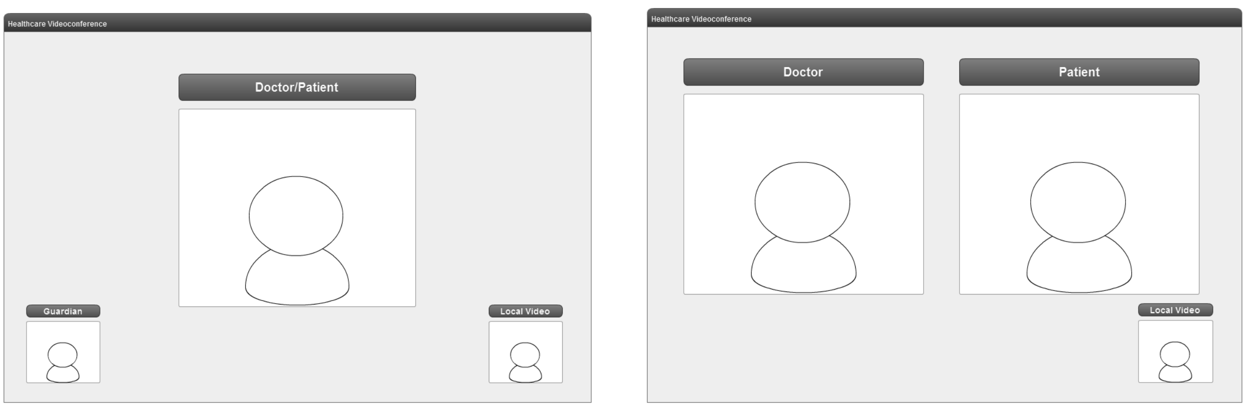


Figure 2. Architecture of Licode MCU

The concept of rooms works well for the use case of a doctor/patient/guardian session, since they need a private space that needs to be independent from other sessions. Licode maintains a list of streams for each room. Browser clients need to explicitly publish their media streams in order to be added to the session, and they need to subscribe to the streams published by the rest of the participants as well.

In the current prototype, the user can select from three different roles, adapting the layout accordingly. One room can host up to three people: a doctor, a patient, and a guardian. Examples of the doctor/patient and guardian layouts can be seen in Figures 3.a and 3.b. In this video conferencing model, the doctor and the patient are the primary participants and the guardian is present to monitor the conversations and provide any necessary information. Given this designed significance of the participant's roles, the guardian video is displayed at a smaller resolution on doctor and patient screens. Before joining a room, the user must give consent to the browser, allowing it to access the camera and the microphone. Since WebRTC is still under development, only the latest versions of Chrome browser for desktop and Android are supported. Firefox has preliminary support and is expected to interoperate with other implementations as the developments continue.



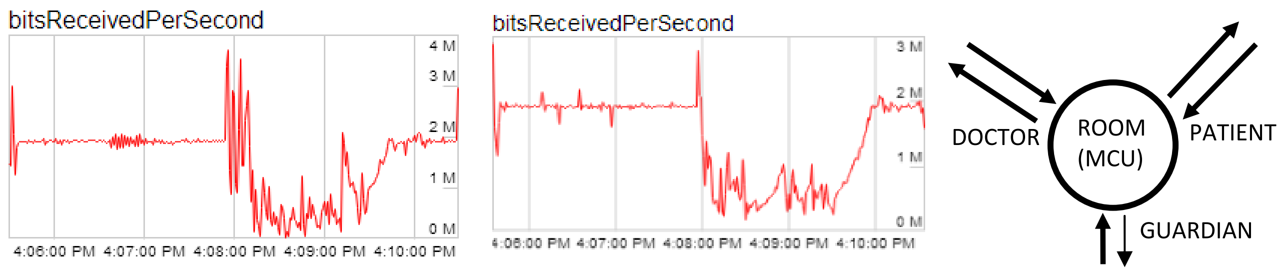
(a) Doctor and patient view (b) Guardian view

Figure 3. Screen layout of the conferencing system

3. ROLE BASED ADAPTATION

WebRTC adapts to changing network conditions by reducing the bitrate of the video in the session. The adaptation mechanism uses RTP statistics and packet loss to make adaptation decisions. In the absence of proper priorities, any of the participant videos can be degraded to alleviate network congestion. This could result in the participating guardian transmitting video at high quality while the doctor's video gets degraded. Such unmanaged adaptation mechanisms can lead to reduced quality of experience and lowered quality of care.

Role based adaptation takes the participant roles into account in making adaptation decisions. If doctor and patient have higher priority roles, the guardian video is degraded first to meet the network adaptation needs. This adaptation mechanism had been implemented using pre-defined participant priorities. The priorities can also be dynamically changed during a session. Context based bandwidth adaptation can be implemented easily by using the methods created to unsubscribe/subscribe to streams based on priority. As an example, a doctor could select an "Examining patient" mode, which would drop doctor's and guardian's video streams, keeping only the voice streams, in order to make sure the best possible video quality is received from the patient. Although right now streams can be dropped/subscribed based on priorities interactively by calling the implemented functions, there is ongoing work on trying to automate bandwidth adaptation based on the statistics provided by WebRTC for each stream.



(a) Patient video received by doctor (b) Doctor video received by patient (c) Room configuration

Figure 4. Role based adaptation

Each of the participants send a single copy of video/audio to the room and the video/audio is distributed to all the receivers that have subscribed to that particular video/audio. If any of the receivers experience congestion, this is conveyed to the sender, and the sender reduced the quality of the video alleviate the congestion and adapt to the changing network conditions. In the proposed three-party conferencing, without any priorities, if the guardian has poor network conditions and experiences packet loss in the downstream direction (receiving video), the doctor and patient both reduce the send rate there by decreasing the quality of video they transmit. The quality of video received by the doctor and patient drops even if the connection between them is not experiencing any congestion. With role based priorities, since guardian has a low priority for both sending and receiving video, the guardian can unsubscribe to doctor and patient video there by restoring the quality of the session between the doctor and the patient.

Figure 4 shows the results of such role based adaptation. The plots are from the actual system implemented with simulated network congestion. Each participant transmits video at 2 Mbps under normal conditions. The experiments were conducted in a lab with PCs connected to an Ethernet. The plots show the bitrate of the patient video received by the doctor and vice versa. In the evaluated scenario, the download bandwidth of the guardian was reduced to 0.5 Mbps using the NetLimiter bandwidth shaper. When the network congestion is encountered at the guardian, those loss statistics result in the patient and doctor videos being coded and transmitted at a lower bitrate/quality. This can be seen at time 04:08 PM in the plot. A minute later at about 04:09 Pm, the guardian unsubscribes to the patient and doctor videos. With the congested receiver (guardian) no longer receiving videos, the receiver statistics do not show any loss and the quality of the video is slowly restored to normal by 4:10 PM. This two minute period can be reduced by adapting programmatically. The results shown were generated interactively and adaptation periods extended to easily demonstrate this mechanism using plotted data.

The supported features of the proposed system are:

- Doctor/patient/guardian session with independent audio and video streams.
- Ability to drop/subscribe to streams manually based on priority.
- Encrypted media streams provided by WebRTC.
- Video/audio bitrate adaptation based on bandwidth provided by WebRTC.

4. CONCLUSION

We presented a video conferencing system with role-based priorities for use in healthcare applications. The system used Web technologies and emerging WebRTC standard to enable seamless healthcare conferencing applications. Use of web technologies ensures the availability of the application on commonly used platforms. We propose and demonstrate the utility of role-based adaptation for such applications. Adapting video conferencing sessions based on participant roles will lead to higher quality of experience for all participants in the session. Adaptation can be automatic or interactive based on application needs.

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