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Open Source Digital Solution to Replace Auditoriums

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Abstract: ***This proposed model looks to bridge the gap of communication between the employees and their peers in a company by using Real Time Communication. On an average day, conducting a session in an auditorium is a tedious task; our solution eliminates the need for an auditorium that can be easily incorporated within company premises. To maintain data integrity, security has been ensured at device, link and content layers. With minimum movement of the workforce, they can efficiently communicate amongst themselves reducing logistical issues. Firstly, the device is secured by incorporating firewalls. The content is secured by various encryption protocols and the link is secured by Secure Sockets Layer (SSL) certification. The end system is evaluated by sending multiple requests to the device at the rate of a few concurrent requests at a time, to ensure overall stability of the system.***

Keywords: WebRTC, E-Auditoriums, Real Time Communication, Secured Communication

1. Introduction

A full-scale auditorium always has the problem of being able to accommodate only a fixed amount of people, and does not suffice to most meetings held in offices. Primarily the chief concern of a session in an auditorium is to be able to logistically accommodate all the employees of the firm. In this sense, an auditorium may fail to do so. Hence, the development of an E-Solution to an auditorium solely motivated by the problems faced in the inhouse expenses in conducting a meeting came into existence. Today, auditoriums are popularly used for various reasons in a wide plethora of domains, and there is no concrete solution to address the various problems regarding the same.

Owing to the aforementioned concerns, gave rise to a E-solution to an Auditorium using Real Time Communication protocols, and primarily also not to compromise the integrity and validity of the data being transmitted, 3 different layers of security have been incorporated.

WebRTC, Web Real Time Communication, one of the very evolved collection of communication protocols and application programming interface that enables real time communication between the devices connected in the peer to peer fashion. As of today, WebRTC is being standardized by the W3C (World Wide Web Consortium) and IETF (Internet Engineering Task Force). The proposed model uses OpenWebRTC which is an open source implementation of WebRTC to make our solution feasible to all sets of workforces and domains.

One – to – Many Broadcasting, where one of the peers connected in the network acts as the master of the session, and the remaining peers connected are all receiving the broadcast of the master. To facilitate smooth functioning of the system, a media server and a NodeJS server are used to transfer and serve the data to various peers. Yet, this solution has a considerable amount of limitations owing to the network bandwidth and the availability of the peer nodes in a company is a still a question for contemplation.

The current existing model does not include the security features and is not open source.

Software Architecture

As figure 1 explains, the end system was built, keeping in mind various parameters out of which the utmost importance was given to security and credibility of the system. The Software Architecture diagram above mentions the components that were implemented and used in building the product.

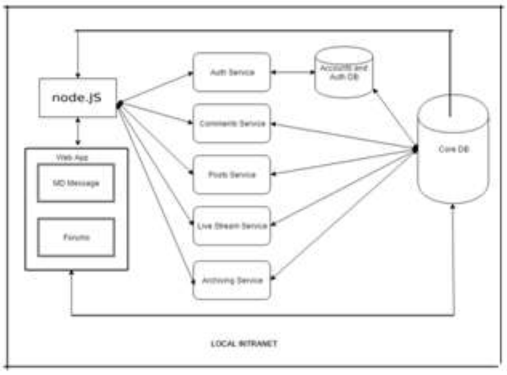

Figure 1: Proposed software architecture for the model

Firstly, to ensure the person logging into the session was credible and reliable an Authentication Form was implemented, compared with a core database to facilitate the authentication service.

Secondly, once a peer is inside the system, he/she may instantiate the session and hence forth will be called the master, by starting the One – to – Many Broadcasting service by running the media and NodeJS servers.

Thirdly, in an ongoing session, there can always be a chance to ask questions at the master by his or her peers, which can be done by Forums Service. A simple yet powerful upvote system has been incorporated to increase the importance of the question and Natural Language Processing Algorithms have been implemented to remove and adjudicate redundant questions.

Fourthly, the master has been allowed the privilege of individually of monitoring the said connected peers in the network by using the Connected Device Services, which is essential to the session.

Fifthly, the master can share his screen in order to entitle his employees to be able to view certain files and folders, for a better understanding and a overview of the session.

The End system is a web application which can run on any device that supports Real Time Communication Protocols, along with required peripherals to start a video broadcast or view one.

Features

The core of WebRTC is implemented using node.js[1]. This WebRTC server runs on a special media server which runs all the authentication, the forums and one to one video services. The authentication is done using a powerful JavaScript middleware named passport.js[2]. The forums are designed to be flexible and easy to use. One to one video services are used to provide unique connectivity between the host and the attendee.

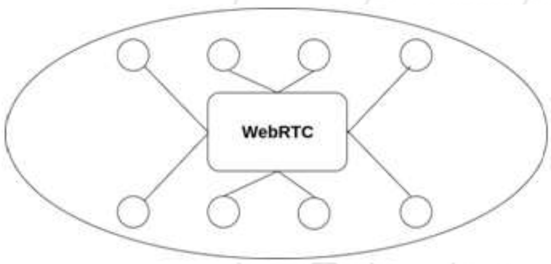

Figure 2: Node Structure of WebRTC

For the user to login to the system Oauth based token authentication and authorization is implemented. This system ensures that a secure delegated access to the system with processes being the features of the system. The processes issue access tokens to navigate around the resources.

Forums are essential part of this system. These are designed to ensure the attendees of the meeting are entitled to ask questions to the host and the host may choose to answer them via comment on the same post or may choose to answer it using the one to one chat service. The redundant questions are removed using the Natural Language Toolkit [3] and a dictionary of English Words. The forums also consist of an upvote system where the attendees can upvote and their voice is heard.

One to One video service is used to make sure the host is able to answer certain questions to the attendees privately. The host also has the ability to check on the behavior of the attendee so as to have a confirmation of his body language and behavior.

Security

As mentioned in figure 3, the three security layers incorporatedare as follows:

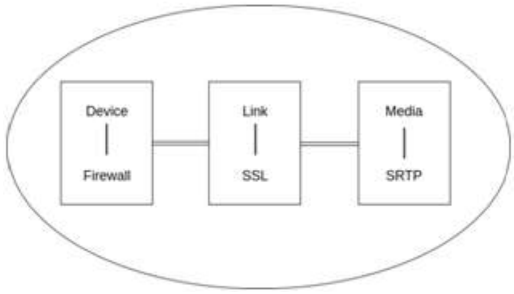

Figure 3: Security level incorporated

- Link: The link is secured with Secure Sockets Layer (SSL) [4] Technology. This is a Transport Layer Protocol which uses symmetric cryptography and unique keys are generated for each connection. This connection also ensures integrity of transmitted data.
- Content: The video content security is ensured by Secure Real Time Protocol (SRTP) [5]. This encrypts the whole media stream. For encryption and decryption of data, SRTP uses Advanced Encryption Standard (AES) as the default cipher.
- Device: The security of the device is ensured by using firewall. This ensures that the only open port is the one through which this system communicates and the intruder gains no access to this system unless granted physical access.

Performance

One of the underlying issues that any model faces, is performance and actual implementation. To validate and improve the performance, the end system was subjected to the following tests, where a 1000 requests were sent at the rate of 10 concurrent requests at a time as mentioned in figure 4. As every request approaches, it undergoes the following series of events; firstly, the Authentication service takes over, and authenticates the user trying to log in to the interface. Secondly, if there is an ongoing session, the session service takes over in intimidating the master, which a new node has joined the session. Finally, the end system is maintained and the master and user can have a hassle-free broadcast, resulting in an establishment of connection between the master and the node that has just joined. In order to test the model under actual working conditions, the above tests were applied to prove its authenticity.

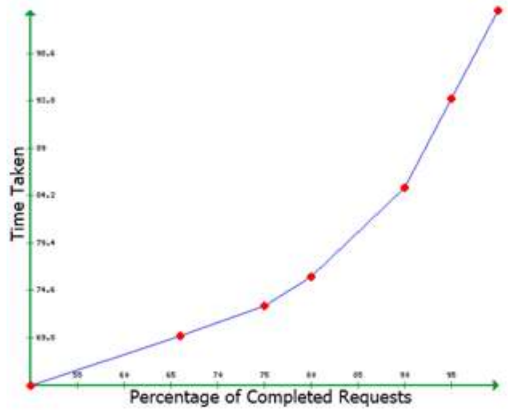

Figure 4: Percentage of Completed Requests vs. Time Taken

Conclusions

The Open source digital Solution for an Auditorium, was developed keeping in mind the logistical space constraints that are faced in MNCs and big corporations in conducting meetings. To be able to facilitate communication without movement of the company workforce was the primary objective of this project. Extensive research and literature survey was done on Real Time Communication Protocols and One – to – Many Broadcasting. Since, there is no extensive solution to address the problems faced in Auditoriums, there is a need for creating this tool kit to extensively facilitate or enhance the office workflow.

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