WebRTC based social video broadcasting

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Abstract- WebRTC (Web Real-Time Communication) is an API definition drafted by the World Wide Web Consortium (W3C) that supports browser-to-browser applications for voice calling, video chat, and P2P file sharing without plugins[1]. Historically, RTC has been corporate and complex, requiring expensive audio and video technologies to be licensed or developed in house. Integrating RTC technology with existing content, data and services has been difficult and time consuming, particularly on the web.[2]

With WebRTC being developed as a web standard by relevant bodies such as IETF and W3C to ensure industry consensus, real time communication (RTC) on web applications should be as easy as entering text in a web page. It can be done without the need to download/update any plugins or extra piece of software. It would come with native support from the browser.

This project is about building a video broadcasting application using WebRTC on a social broadcasting web platform which will be used to launch the application. Our main task in this project would be the survey of this new technology WebRTC. Next, we need to think about how to do signaling to get the public IP address of the destination node so that a p2p connection could be setup. Finally, create a HTML5 responsive web application for this WebRTC app and use phonegap to convert this to mobile apps as well.

**Keywords - WebRTC**

INTRODUCTION

- **Problems to be addressed-** As WebRTC is just client side JavaScript API, it becomes a problem when the users are behind the NAT server. Then, it is not straightforward to get the public IP address of the destination node.
- **Why are these problems important?** - The above mentioned problem is important as most professional working environments would be behind a NAT server and not giving public IP. As WebRTC is just client side JavaScript API to be run on the browser, it needs public IP of the destination machine to establish a peer to peer connection.
- **Applied technologies & Solution--** WebRTC API, HTML5/CSS3, JS/jQuery, PHP, Node.js, ASU MobiCloud Environment, phonegap and MySQL. Our solution to the problem would be using STUN server to establish the connection. But the STUN server would work for about 80%-90% of the time. If the destination is behind the NAT server

than we would need to employ TURN server to get the public IP address. The downside with the TURN server is its cost, but there is a free and open-source TURN server available from google, which we plan to use.

- **Expected Outcome -** A social broadcasting web application with user profile management similar to Google™ Hangout, but the upside would be it would work without the need of any plugins. Android users would be able to download an app.
- **Project Management Plans -** We'll be driving the project into multiple phases, depending on the work. First phase would be the setup of user management in the site. The next would be setting up video conferencing using webRTC. Secondly, we would take care of signalling using Node.js. Thirdly, testing the project for bugs and focus on load testing as to figure out how many people are able to join the conference. All the work would be divided amongst all the group members such that each would have the chance to work on every technology used in the project and the load is balanced as well.

SYSTEM MODELS

A. System Model
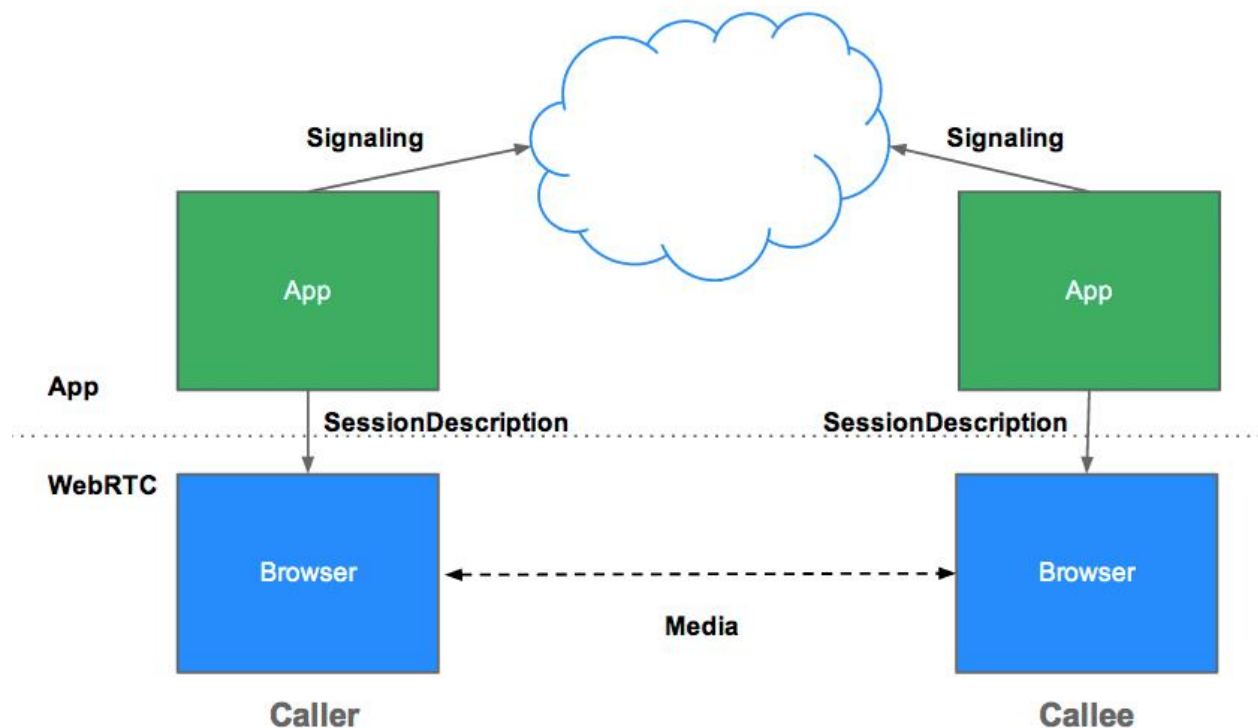
The architecture described above is called JSEP, JavaScript Session Establishment Protocol. Once the signaling process has completed successfully, data can be streamed directly peer to

peer, between the caller and callee—or if that fails, via an intermediary relay server. Streaming is the job of RTCPeerConnection[2].

B. Software and Languages

- webRTC API
- HTML5/CSS3
- Twitter Bootstrap
- JavaScript
- PHP
- Node.js / node packages express.js. socket.io
- MySQL database
- Chrome with ripple emulator extension

PROJECT DESCRIPTION

A. Project overview

The objective of this project is to create a social broadcasting application which allow users to broadcast & record the event which can be viewed by any user on the internet. The social platform will let the user choose the categories they are interested in and they will get notified about all the upcoming events of the selected categories. We can also provide an option for the user to sign in through facebook or google plus and have them send out invites to their friends.

The project is divided into six main tasks which includes Survey, Environment setup, Requirement Analysis, System design, Implementation, Testing.

Survey
In this phase a detailed research on WebRTC, Signaling server, STUN server, TURN server is done. What are all the API's provided by WebRTC, what all we can use for signaling (signaling methods and protocols are not specified by WebRTC standards[2]), what all if any free STUN and TURN servers we can use for our project. Also find out about which network topology to use star vs mesh. Each has its merits and cons.
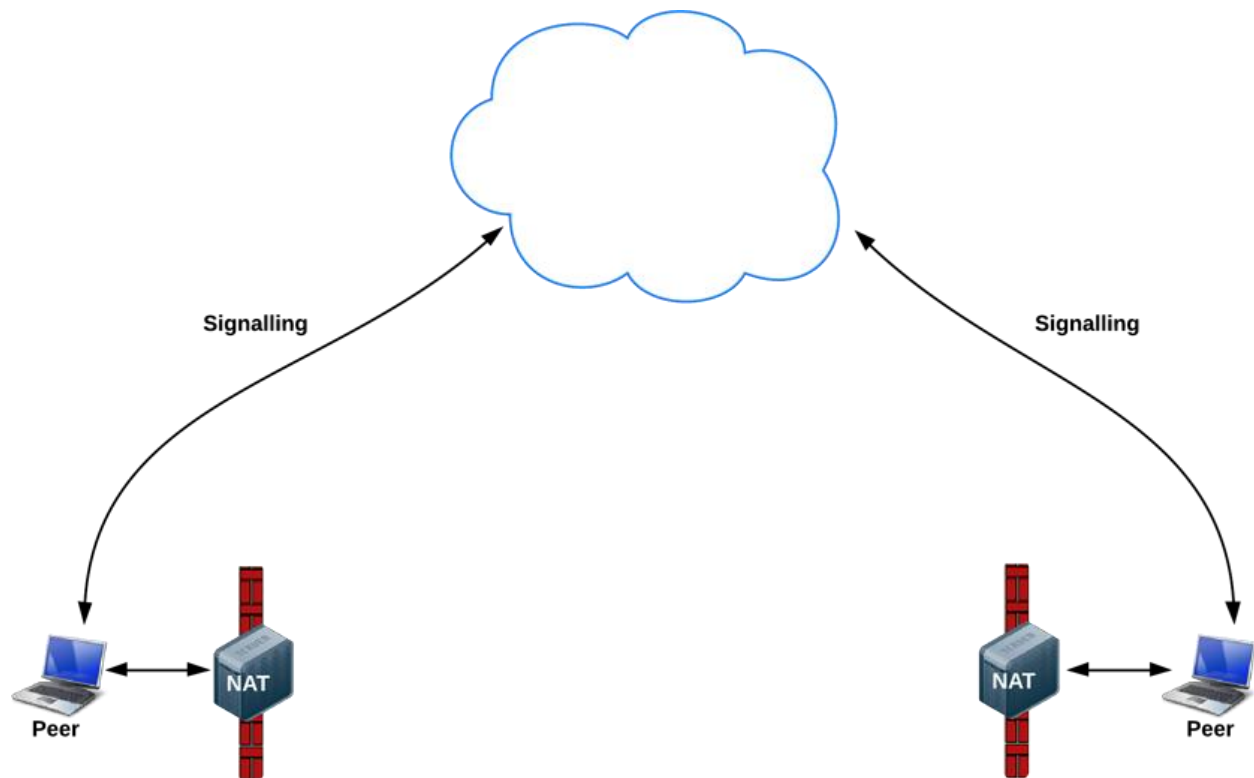
The real world

As we can see in the diagram above, In reality most devices live behind one or more layers of NAT, some have anti-virus software that blocks certain ports and protocols, and many are behind proxies and corporate firewalls. WebRTC apps can use the ICE framework to overcome the complexities of real-world networking. ICE tries to find the best path to connect peers. It tries all possibilities in parallel and chooses the most efficient option that works. ICE first tries to make a connection using the host address obtained from a device's operating system and network card; if that fails (which it will for devices behind NATs) ICE obtains an external address using a STUN server, and if that fails, traffic is routed via a TURN relay server.[2]
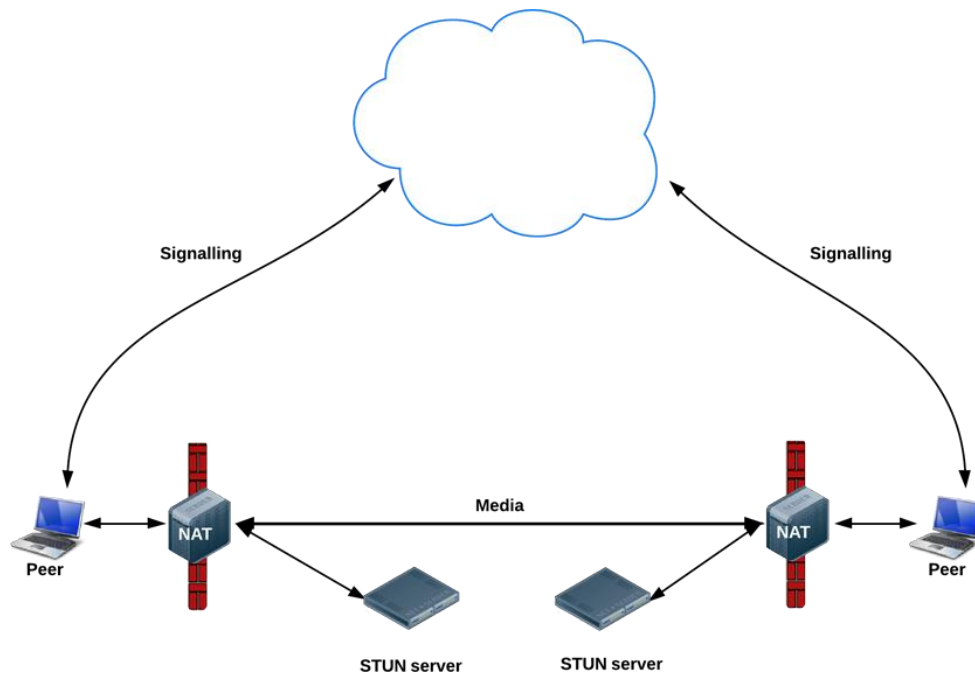
We can implement using either star or mesh topology. Using star we do not have to worry about overloading network bandwidth but everybody can't each other like in mesh.
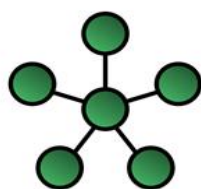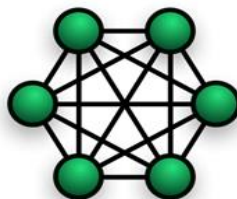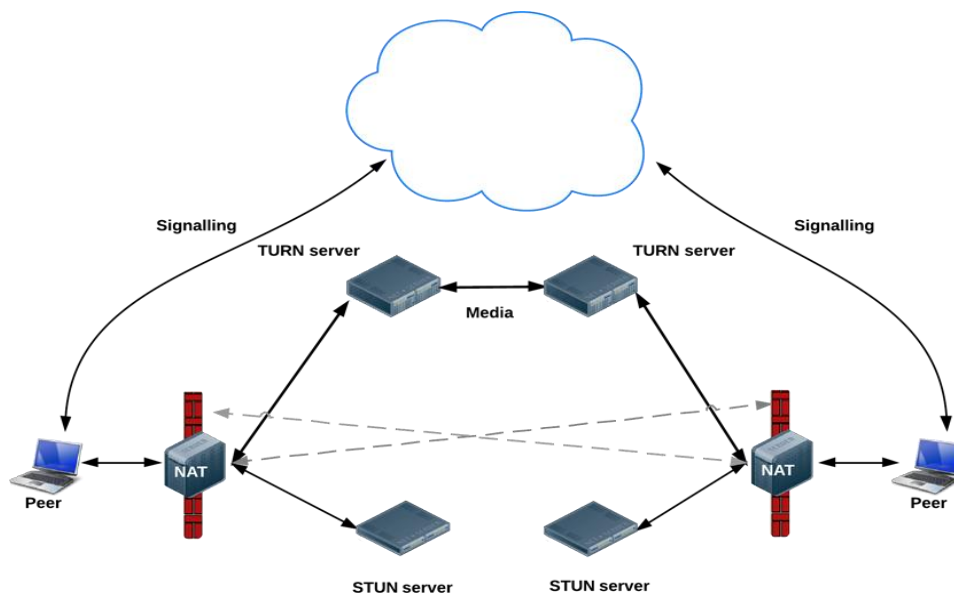

Star Toplogy   Mesh Toplogy

Using STUN servers to get public IP:port addresses[4]

The full Monty: STUN, TURN and signaling[4]

We built our signalling server using node.js with socket.io.js or else we can use ready made signalling servers like webRTC.io, easyRTC, Singalmaster.

Environment Setup

We request access to servers in the cloud environment. We would be in need of a couple of virtual machines. These would serve to be the basic necessities to start the project. Once this is completed, we move on to downloading and installing softwares on the virtual machines and the client side nodes. We would be making use of Node js, a browser that supports WebRTC preferably Google Chrome. Also this version of the browser should be compatible for HTML5 and CSS3.

Requirement Analysis

In this phase we would analyse the user needs and requirements for the project. Create a detailed functional requirements document.

System Design

In this phase a system is designed based on the proposed model which includes creating UML diagram which will help in identifying how the system will run and how entities interact with each other.

Implementation

● Task 1: HTML5 responsive web application

Make a responsive website using HTML5 and twitter bootstrap. The website should have login page and should show dashboard to the logged in users. {Completed the Home screen, Login screen, Currently working on user dashboard and styling the chat rooms.}{Completed}

● Task 2: Setting up user management on the website

We come up with database design for the website. Write backend code in a scripting language like PHP to connect the website front end to the database. {Completed.}

● Task 3: Setting up the signaling server

Using node.js set up the signaling server or use readymade signaling servers which would handle the establishment of the p2p connection. {Completed.}

● Task 4: Setting up the STUN/TURN server

We plan on using the Google's free and open source STUN/TURN server and configuring that to our project setup. {The connection is being established using STUN server. Need to configure a fallback TURN server}{Same status.}

Testing and Re-engineering

In this phase a set of test cases will be created to measure the efficiency and performance of the system. After running the test cases re-engineering will be done on the processes which needs improvement. This is an important phase in the project and will be repeated until all the project requirements are met.

B.   *Project Task Allocation*

This is a 3 member group project. We are not planning on having any team lead sort of thing and everybody would be having equal responsibility. As, we are using GIT on the mobicloud for the SCM, that would help us collaborate on the project as well. Each of us would pick up tasks and complete their part and push to GIT.

C.   *Deliverables*
   a. A customized web site for users to sign in (can sign in through Facebook or Google plus) {It requires approval from Facebook for which we need to submit the app first. Now, we'll submit the app and embed Facebook login.} {Completed.}
   b. The users after signing in should be able to watch any video being broadcasted as well as broadcast the videos. {Completed.}
   c. The users before signing in has the privilege of watching prerecorded videos.{Completed}
   d. The chatroom can be interactive where the user can chat or raise a question and the broadcaster can respond to the user. {Chat is implemented but no button provided "Raise Hand"} {Username is coming along with the chat now. }
   e. The signed in user has the option of sending out invites to his/her friends from either Facebook or Google plus. {Not Done} {It requires approval from Facebook for which we need to submit the app first. Only Email, basic profile information could be accessed without app review. }

Software Packages, Tools and languages used for this project
   ● HTML5/CSS3
   ● Twitter Bootstrap
   ● JavaScript
   ● PHP
   ● Node.js / node packages express.js. socket.io
   ● MySQL database

Algorithms

   ● For signaling server using node.js and socket.io.js
   ● For recording the video and storing it.
   ● For breaking the storage of the video according to time ex 10min or 15 min video.

D. Project Timeline

Phases	Start Date	End Date	Duration (in days)
Prework	9/2	9/5	3
Development	9/5	10/1	25
Testing	10/1	10/6	5

E. Mid-Term Progress

a) Tasks Completed:-

Database Structure

```
mysql> desc user;
+------------+-------------+------+-----+---------+----------------+
| Field      | Type        | Null | Key | Default | Extra          |
+------------+-------------+------+-----+---------+----------------+
| user_id    | int(11)     | NO   | PRI | NULL    | auto_increment |
| user_name  | varchar(20) | YES  |     | NULL    |                |
| first_name | varchar(20) | YES  |     | NULL    |                |
| last_name  | varchar(20) | YES  |     | NULL    |                |
| email_id   | varchar(30) | YES  |     | NULL    |                |
| age        | int(3)      | YES  |     | NULL    |                |
| password   | varchar(50) | YES  |     | NULL    |                |
+------------+-------------+------+-----+---------+----------------+
7 rows in set (0.00 sec)

mysql> desc video;
+----------+--------------+------+-----+---------+----------------+
| Field    | Type         | Null | Key | Default | Extra          |
+----------+--------------+------+-----+---------+----------------+
| name     | varchar(256) | NO   |     | NULL    |                |
| source   | varchar(256) | NO   |     | NULL    |                |
| category | varchar(256) | NO   |     | NULL    |                |
| duration | int(11)      | NO   |     | NULL    |                |
| start    | timestamp    | YES  |     | NULL    |                |
| end      | timestamp    | YES  |     | NULL    |                |
| video_id | int(10)      | NO   | PRI | NULL    | auto_increment |
| user_id  | int(10)      | YES  | MUL | NULL    |                |
+----------+--------------+------+-----+---------+----------------+
8 rows in set (0.00 sec)

mysql>
```
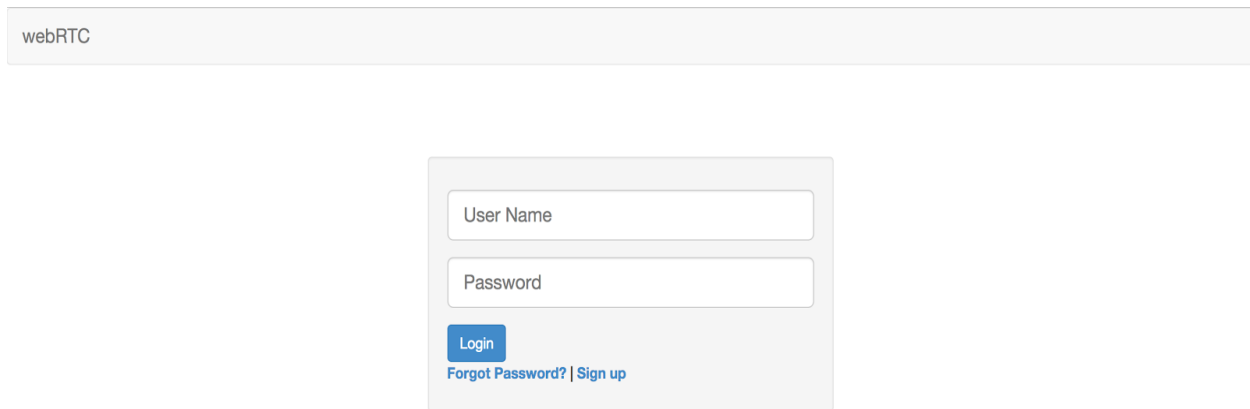
These are the two tables user table and video table. We would be adding few more tables in addition to this such as category table, user table when he logs in through facebook etc as we enhance the functionalities.

Home Screen:- This screen will serve as the front page of the website where we'll have pre-recorded videos categorized in two categories i.e. General and Music. There are only two categories for now but we plan to add more categories like Education, Spirituality etc which might have its sub categories like Math, Physics, Buddhism, Zen etc. When the user will click on the video the video will be played in full-screen mode. The navigation bar will have a "Login" button for the user to register/ login into the System.
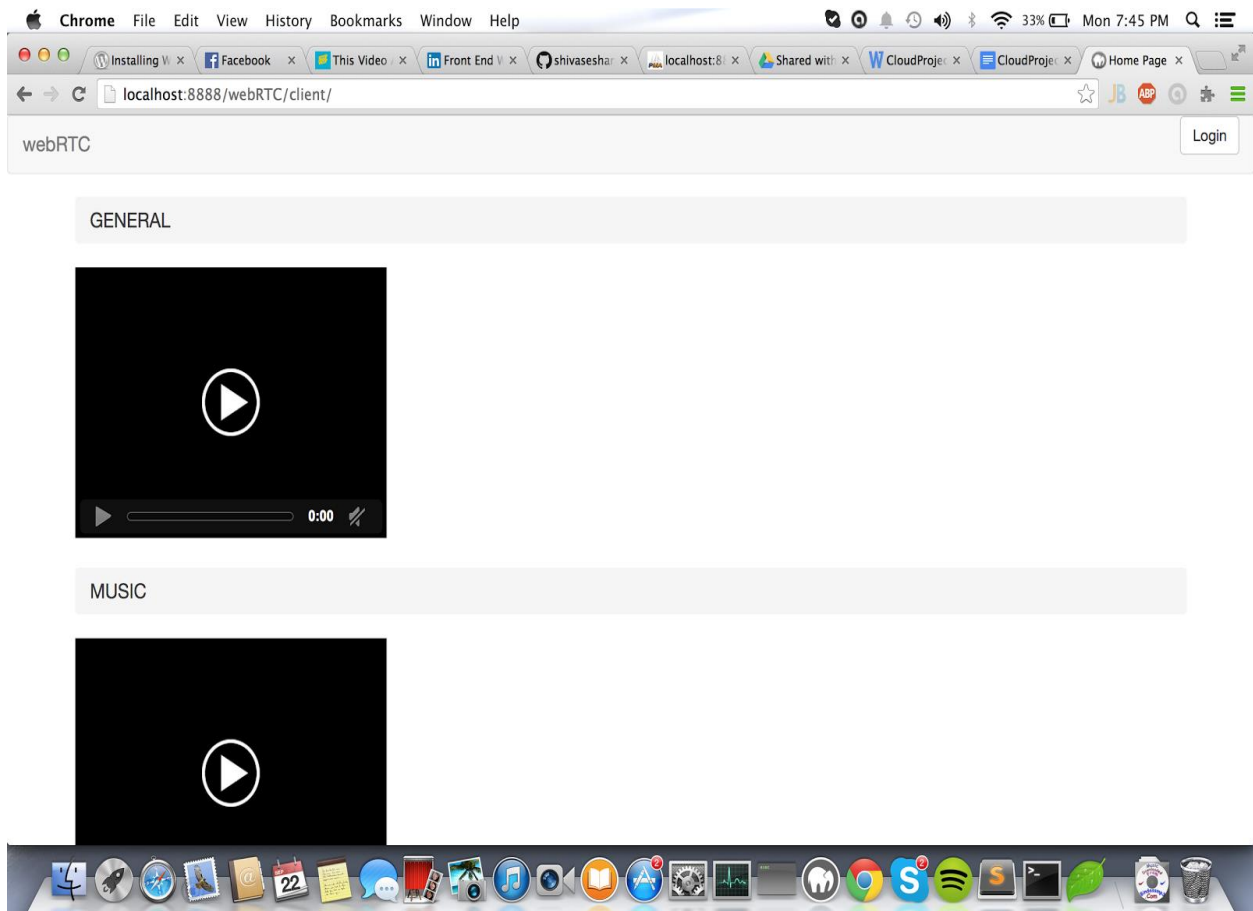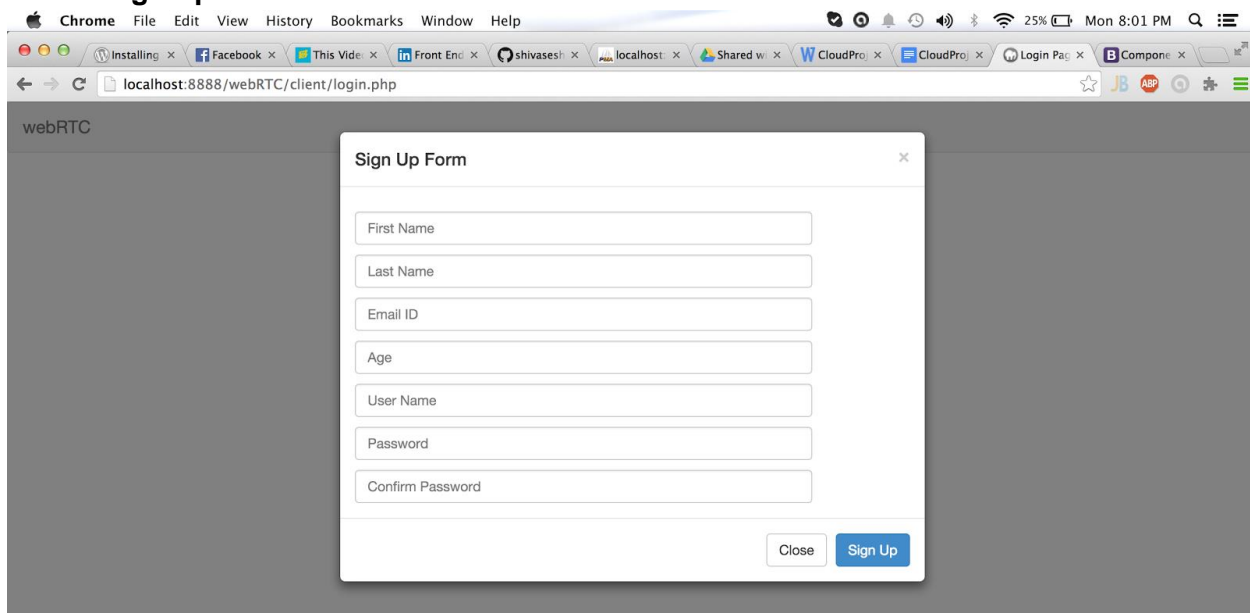
Login Screen:- This screen will prompt the already registered user for a valid Username and password and give an option to new user to register into the system.

webRTC

User Name

Password

Login

Forgot Password? | Sign up

Sign-up Form:-

webRTC video conferencing:-
The initiator is able to see all the participants, where the participant is just able to see himself and the initiator. This is following the star topology where the initiator is center. Also chat window is provided people can chat in real time.
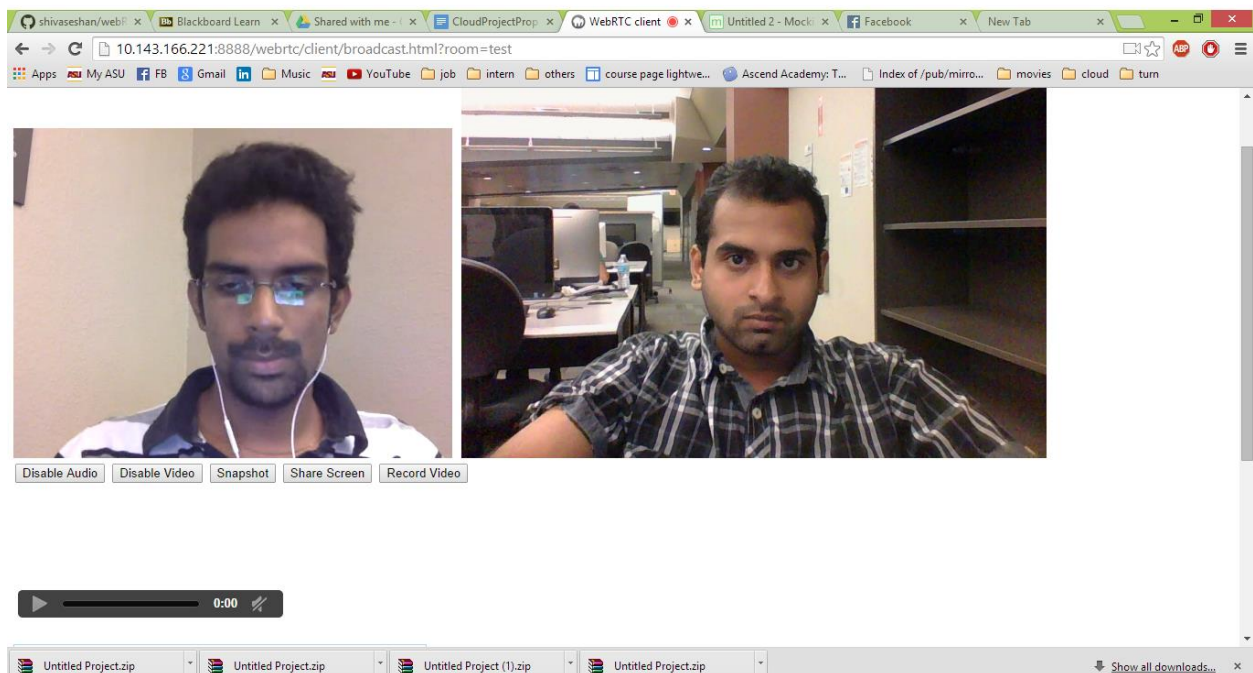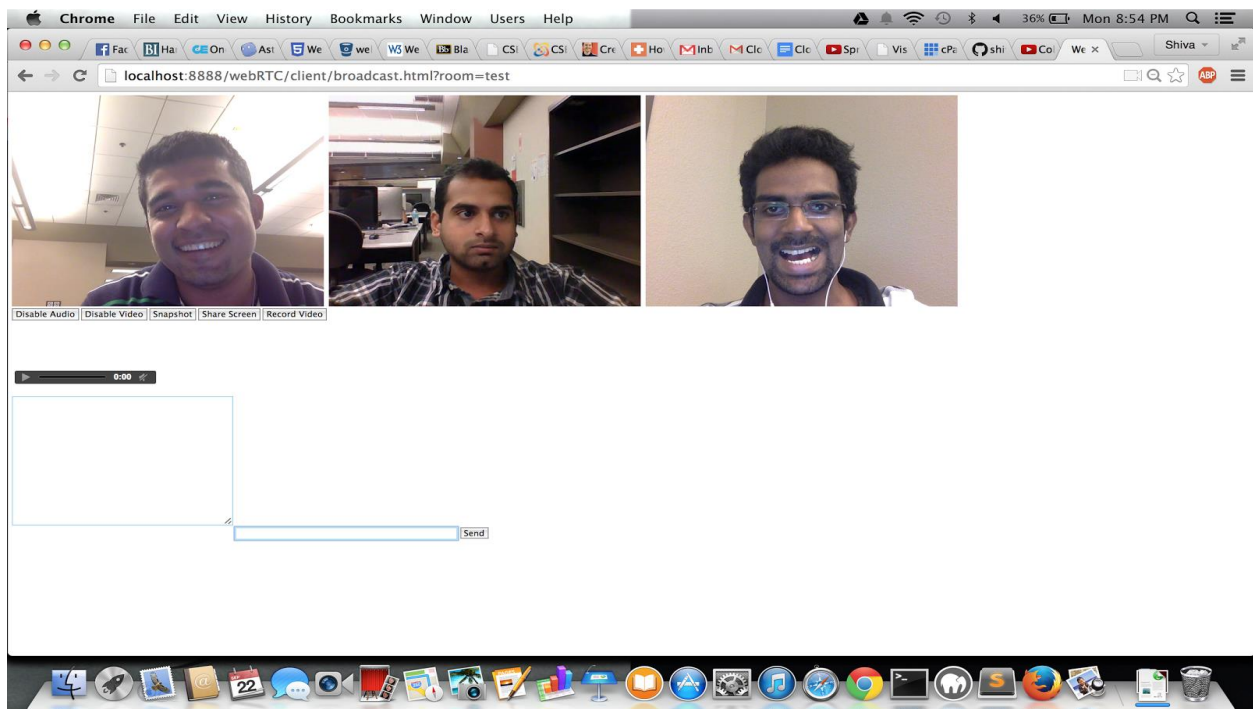
- Disable Audio/Disable Video button works by setting the audio stream/video stream to false. Disable video won't switch off the webcam.
- Snapshot will capture the current frame in the person's video tag and give the user an option to download the image.
- Screen sharing is not working presently, as we are not able to find the experimental flag in chrome v37 and beta 38, which unables to do screen cast.

- Record video button provided records audio and video simultaneously but separately. As MediaStreamRecorder API has not been implemented in chrome natively we are using RecordRTC.js which is free open source implementation based on w3c MediaRecorder API (which is still in draft).

Deployment on vlab
- Installed xampp server
- Installed node server
- Deployed the web application

a) **Tasks Completed:-**

Home Screen: - This screen will serve as the front page of the website where we'll have pre-recorded videos categorized in two categories i.e. General and Music. There is a list on the top of page which will show the list of upcoming events by different users of the website. There are only two categories for now but we plan to add more categories like Education, Spirituality etc. which might have its sub categories like Math, Physics, Buddhism, Zen etc. When the user will click on the video the video will be played in full-screen mode. The navigation bar will have a "Login" button for the user to register/ login into the System.
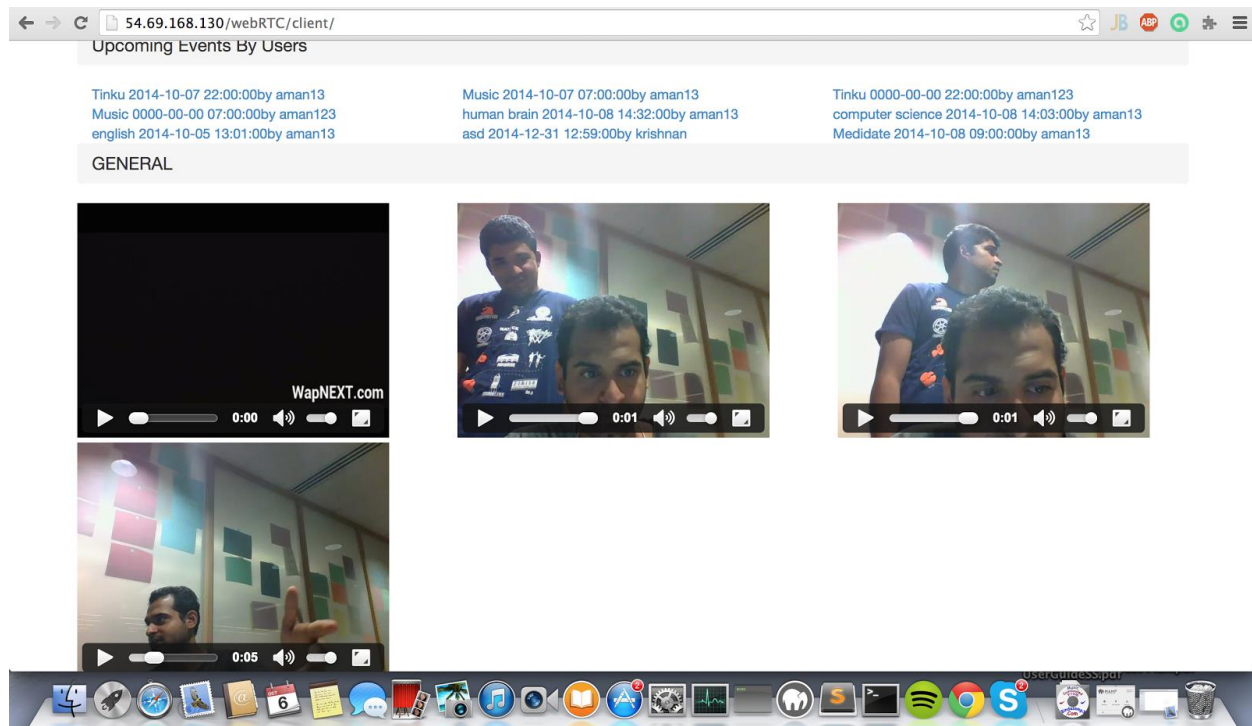
User Dashboard:-

In this page, the user has the option of using four tabs. First is "Dashboard" where the user is able to see his upcoming events. The second is "pre-recorded videos" where user is able to view few of the pre-recorded videos of other users. The third tab is "your videos" where user will be able to see the videos recorded by the user. The last tab is "User Profile" where the user will be able to see and edit his profile basic information.
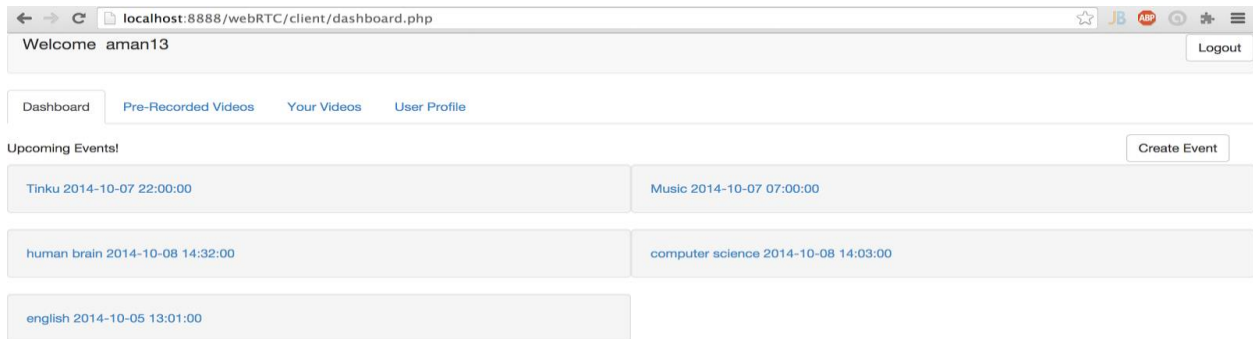

Event Creation Form:-

The event creation will be used by the logged-in users to create events which they want to broadcast. The form will ask for details about the event which includes Event name , event description , Event date , Event start and end time
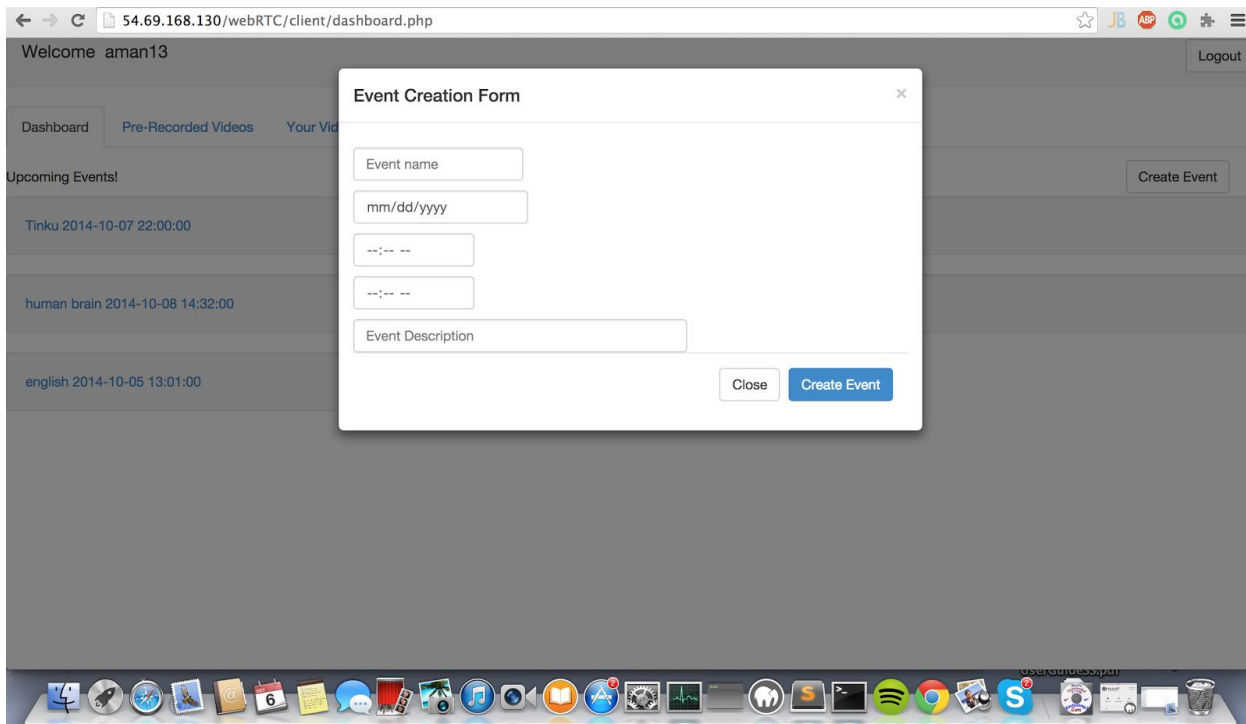
Event Created:-

Once the user will fill the details of the event in the event creation form. The event will be created and it will get added to the list of upcoming events shown on user's dashboard
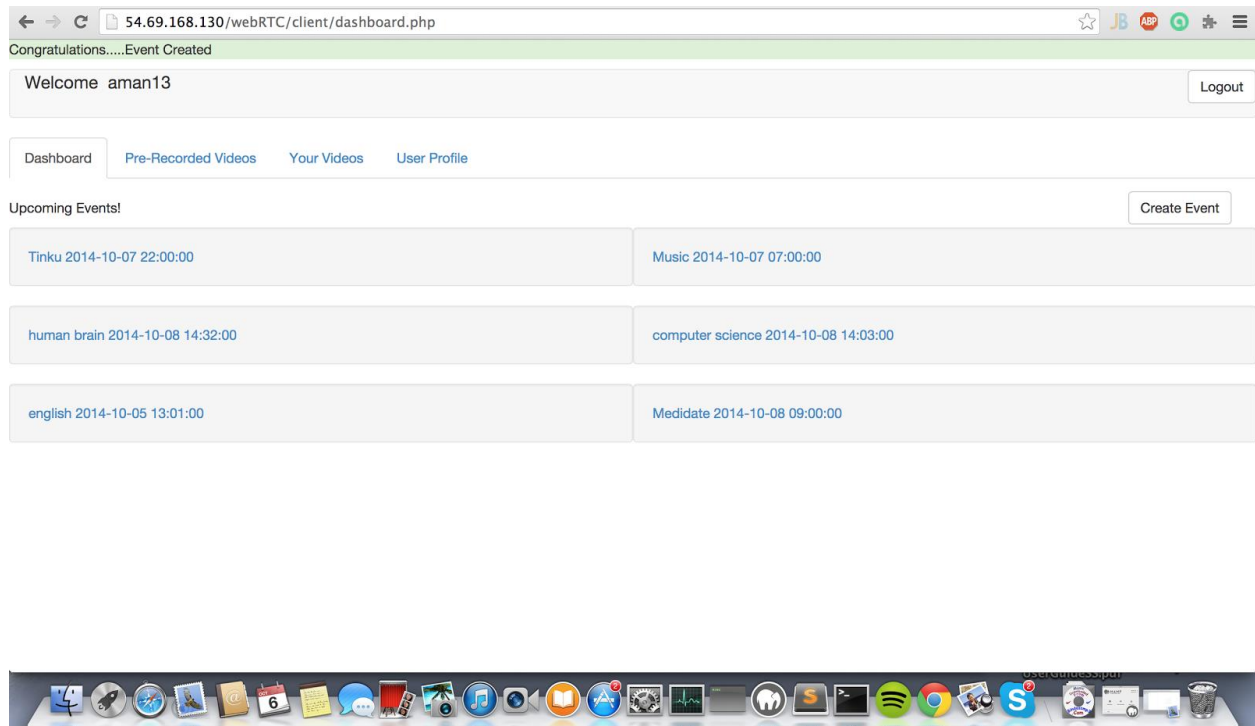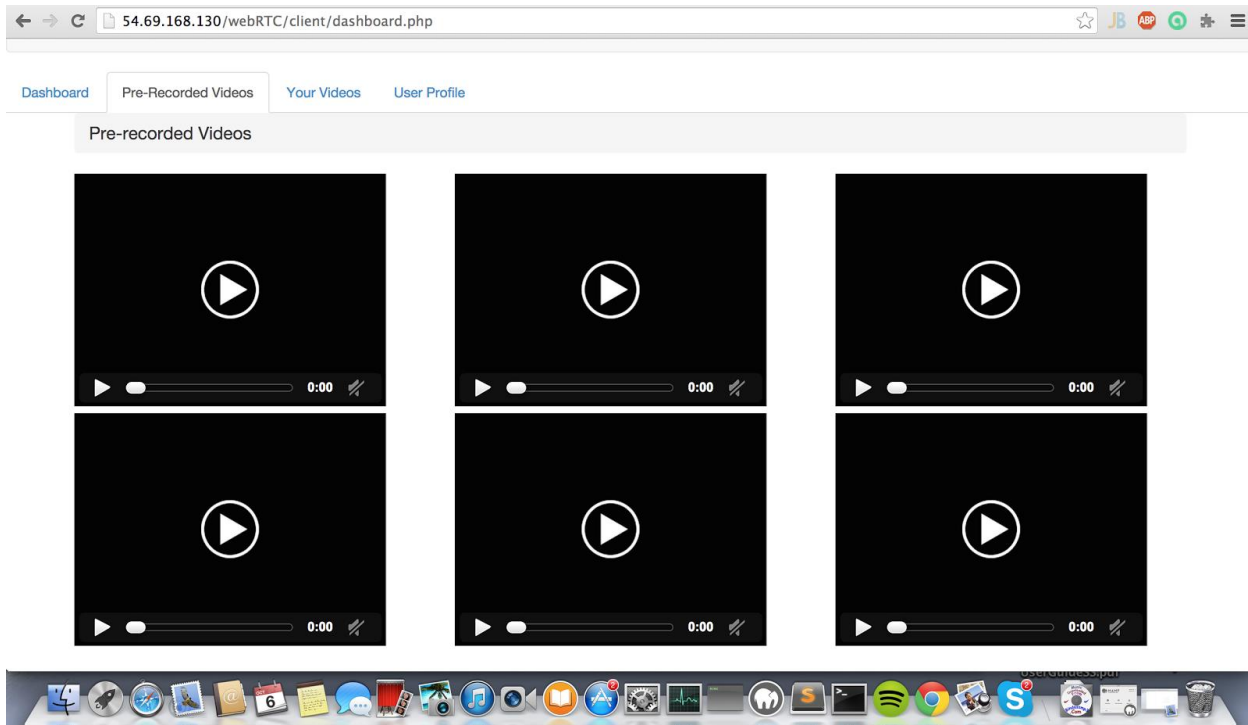
User Profile :-

Here the user would see the information he provided while at the signup page. If he wants he can edit any information.
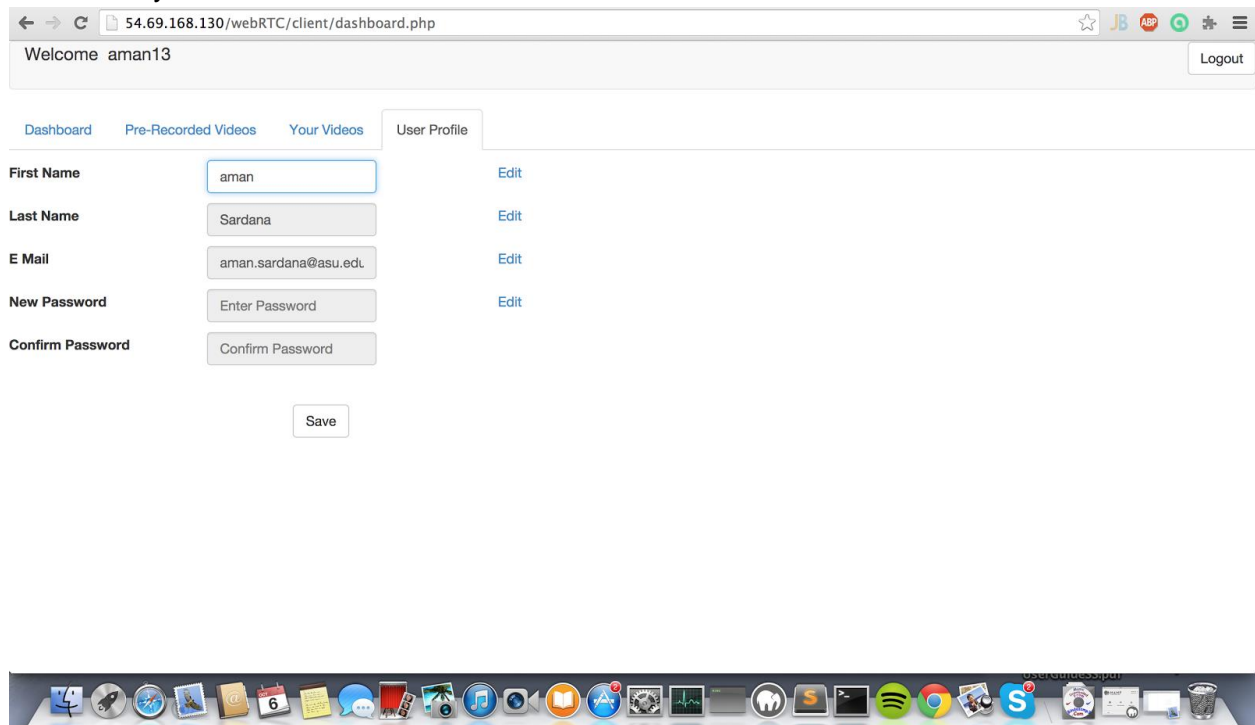
Pre-recorded Videos:-

This tab shows all the prerecorded videos uploaded by any user.

Broadcasting Room:-

We tried to scale the remote video box to full width, but that resulted in the height also going down to maintain the aspect ratio. That results in scroll bars for the video box. As we can see from the below screenshot that the chat window now shows the username along with the chat. Now, there is also an 'End' button which can be clicked to end the call. It redirects the user back his dashboard.
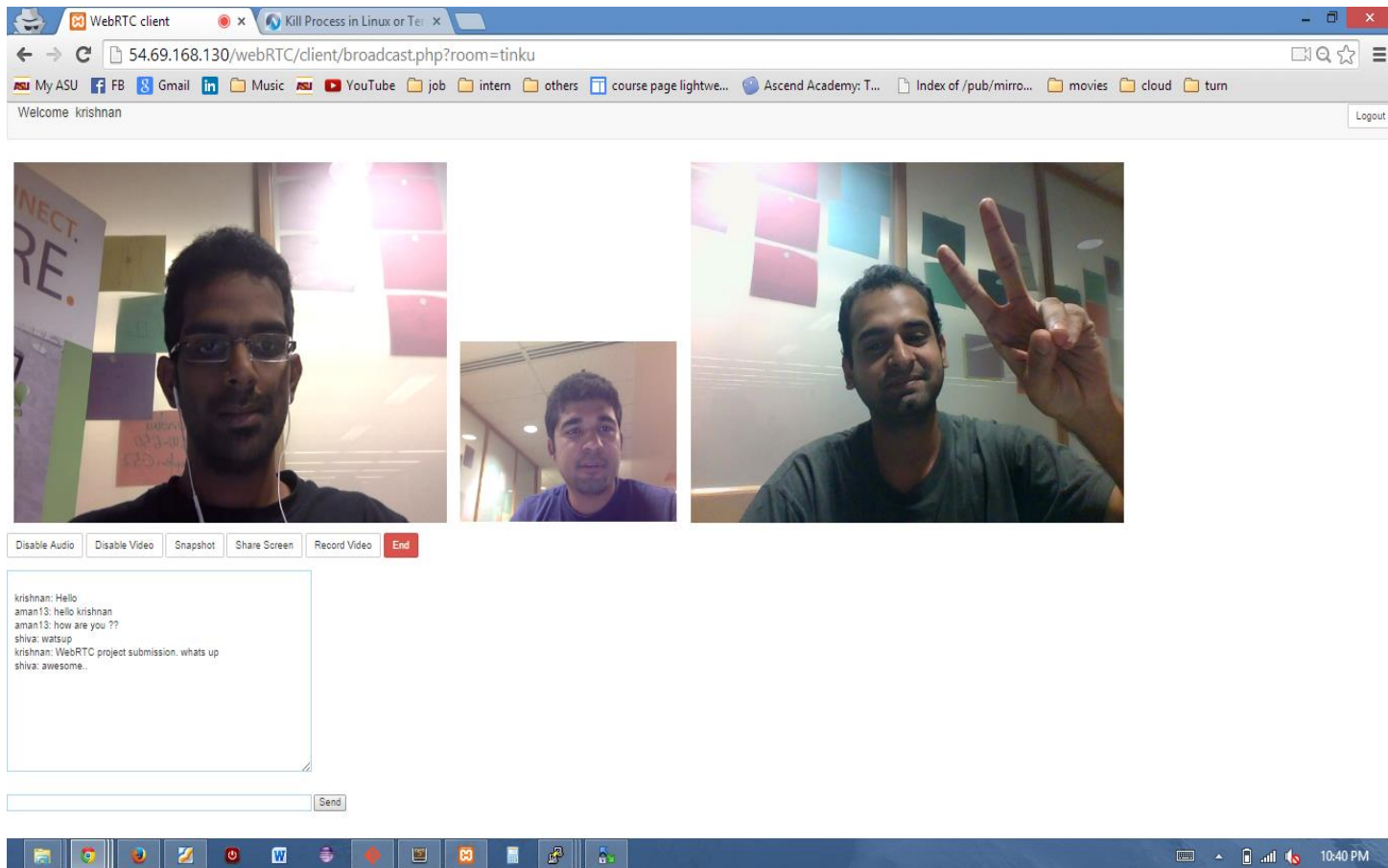
Facebook Login :-

When a user clicks the facebook login page, he is redirected to the facebook sign in popup window. Once he logs in, he is asked to provide permission to the app. After the user gives the app the permissions, he is directly logged in and greeted with his facebook name.
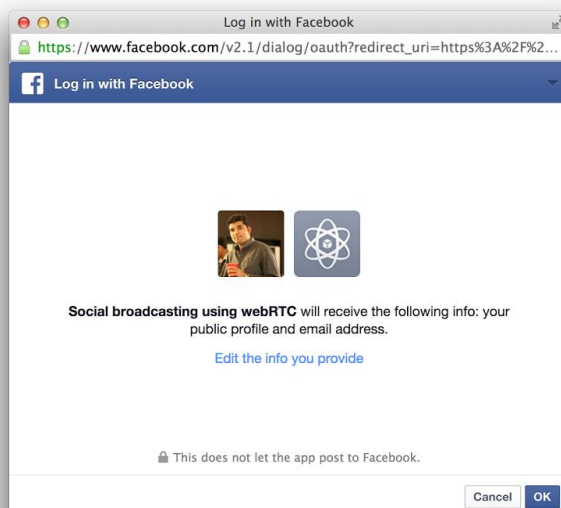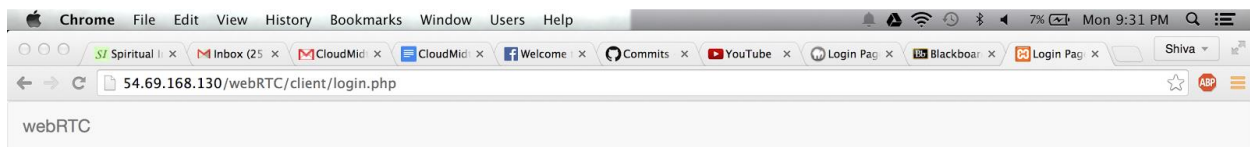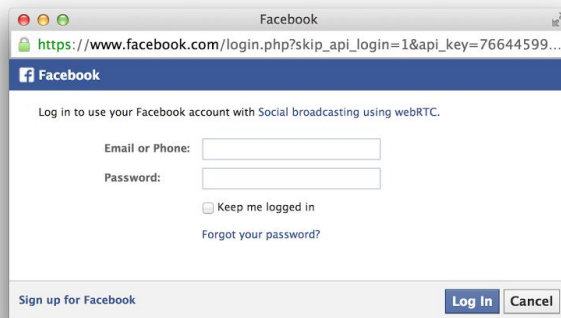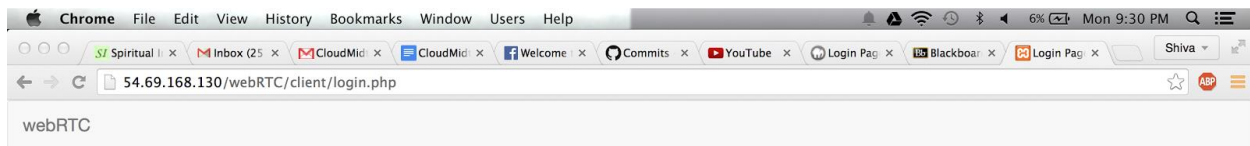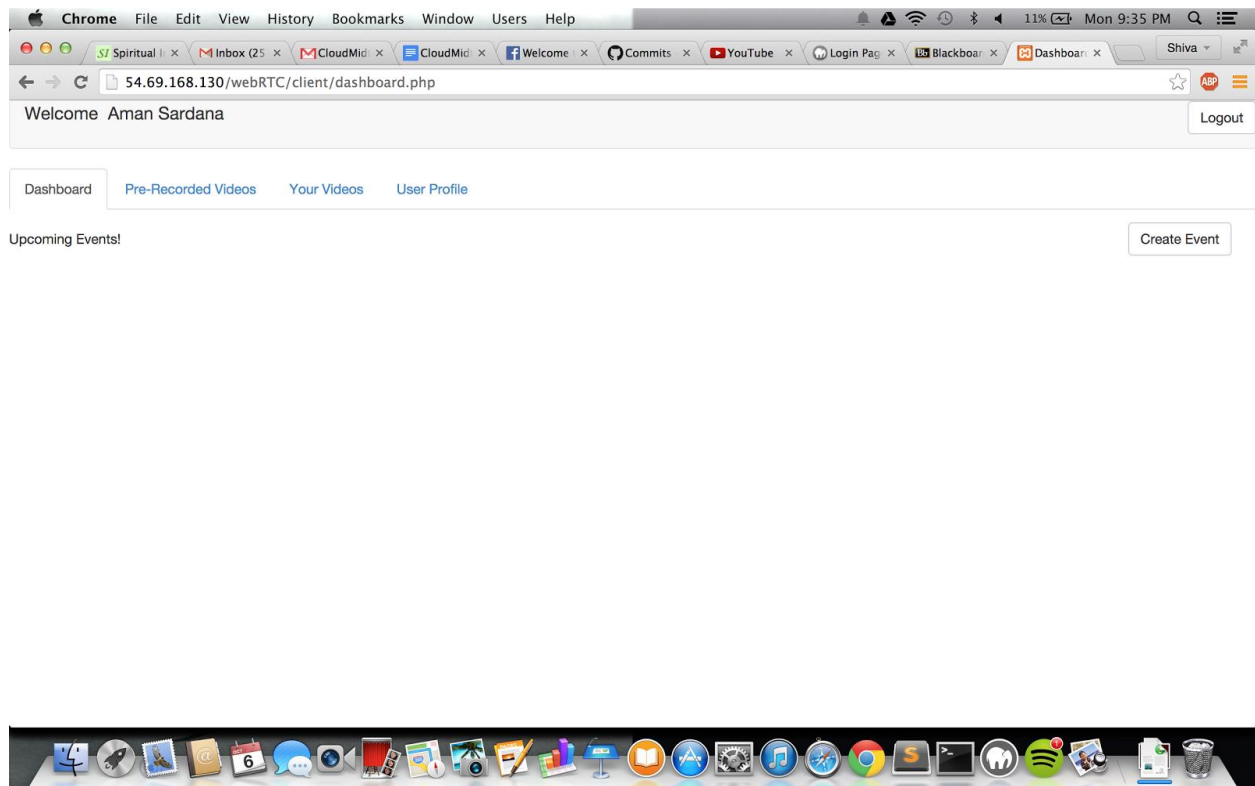
Record Video :-
Chrome doesn't yet support the functionality to record audio track and video track in a single file. So, we were recording audio track as .wav format and video track as .webm format separately. Now we have combined the audio track and video track using ffmpeg and making a single webm format file. Webm is a video format is intended primarily for royalty free use in HTML5 video tag. It takes some time for the video to get combined on the backend. After it gets combined into a single file it starts showing up on the pre-recorded videos.

Hosted on AWS:-
The virtual machines that we got was not being reached by our local machines. We tried to enable all traffic through all ports using ip tables in the vm and yet we couldn't ping the vm from our local. Hence, we decided to host our application on amazon's EC2. We were able to successfully deploy our application on ec2 and now it can be accessed from any local machine. The url to our application is http://54.69.168.130/webRTC/client/.

Database Design:

We created video table which stores the video information that the user uploads. The fb table stores the facebook users information such as email, full name and facebook ID. The event table takes into account the event that the user posts. It includes fields such as event start time, end time, event description etc.

b) **Potential Future Enhancements**

More login options:-
We can have more login options amongst the social networks sites. Like facebook login, we can have a Google+ login button as well for the users to login using their Google Plus account.

App posting to user's feed:-
Provisions for the app to post content on the user's feed. In case of Facebook, we need to go through the process of app review to get the permissions to post content on behalf of user through oAuth. After getting the permission we can provide features like creating Facebook events for him, post about a new broadcasting event he has created and likes on the webRTC app so that his friends circle gets to know about the app.

Phone Gap :-
We have created the app with HTML5 and responsive design in mind using bootstrap. Hence, we can use phonegap to convert our HTML5 app to mobile OS platforms.

Risk Management of the Project

- Nodes which are behind a NAT server will find it difficult to establish the connection. To avoid this, we would configure TURN server. If lot of traffic starts going to the TURN server then that might crash (Load Balancing is out of scope of the project). It would be a third party server that we would be using, so not having much control over that.
- WebRTC doesn't enforce a maximum limit on the number of connections which we can establish. So, as a result we cannot accurately predict the number of connections before which the system might start to hang. It totally depends on the bandwidth of the users in that session.
- WebRTC is a new technology and as it is with any new technology it is not that widely supported as of August 2014. It is supported on Chrome 23, Firefox 22 and Opera 18 browsers for desktop but not Safari and IE.
- As of August 2014, it is not yet fully supported across all mobile devices. The mobile browsers Chrome 28, Firefox 24 and Opera Mobile 12 supports webRTC for all android. There isn't much support for other mobile platforms.
- As of August 2014, it is not yet a complete nor stable, and as such is not yet suitable for commercial implementation.[3]
- Signaling mechanisms aren't defined by WebRTC standards, so it is up to us to make the signaling secure. If the attacker manages to hijack signaling, they can stop sessions, redirect connections and record, alter or inject content[2].

CONCLUSION

Summary - To conclude, this project is about user being able to do interactive video conferencing and being able to broadcast it to the public. There would be a recording available of the session so that the anybody can view it again later. The video conferencing would be achieved using webRTC and the web app hosted on ASU MobiCloud environment. We would have to use Node.js to do the signalling. Presently, it would be a star topology where the broadcaster sends video signal to everybody, so that everybody could see him, but rest not everybody can see everybody else. So this would be simpler in complexity than Google Hangouts but the main advantage of this project is that it would be plugin free. WebRTC is built right into the browser natively. Although till now Chrome, Firefox and opera are the browsers to support this.

Future Extension - Each user (video broadcaster) can start having subscribers. The system would work much like the YouTube subscription where the subscriber gets notified about the

new broadcast from the user. Users would be able to sign in with Facebook and Google+ accounts. When they do that they would be asked permission to send their friend's subscription request. Implementation of mesh topology where everybody can see and hear each other The one who is currently speaking would come in the center of the screen and all others would be lined up at the bottom, much like Google Hangouts.

ACKNOWLEDGMENT

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