Assignment on Location-based Services

Using WEBRTC and based on node.js you are kindly requested to develop the following chatting web application hosting location-based features for two distant chatters (Figure 1). Your implementation should respect the following rules:

- Each user should send audio and video streams from his microphone/camera to the other player and receive audio and video from him.
- Audio and video will pass via the webrtc peer connection api.
- All graphics and texts will pass via the webrtc data channel.
- Location coordinates from both users will be exchanged via the webrtc data channel.
- You can use the graphics you like.

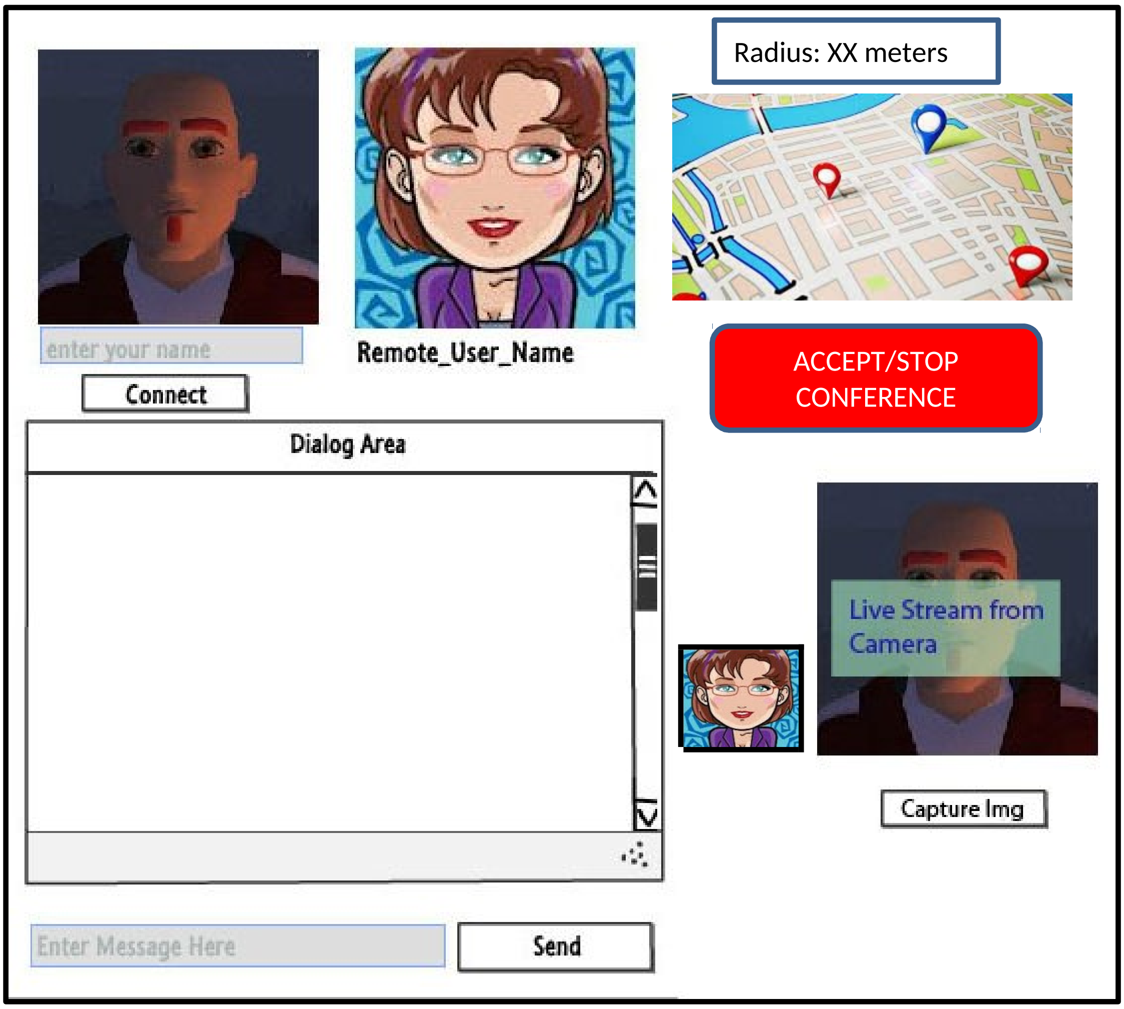

Figure 1: web page of your chatting web application

The scenario of this web app is as follows:

Initially, when a user connects to the web app, he sees at the right frame the stream from his camera. Also, the ***map area*** illustrates his position over a google map. Using the ***Capture Img*** button he can take an image of himself, which will be presented in the left-top frame. The user can update his picture as many times as he wishes, during a session. In the box ***ENTER YOUR NAME*** the user inserts his Nickname used for the chat. Pressing ***Connect*** the user connects to the other user. Hence, he can see the picture and the Nickname of the other user at the appropriate frames on top of the page, next to his image and start chatting with him using only text messages. Text messages are written within the "**Enter Message Here**" box and are also recorded within the Dialogue Area as "<u>Name A says: *message*</u> ". Within the Dialogue Area will be also recorded all the events that concern the two users. In addition, in the map area he can see the position of the other user, along with his position. Each position's mark over the map should illustrate the Nickname of the corresponding user. Within the ***radius box***, the user starting the chat can define the distance within which he wishes to start an audio/video conference with the other user. When both users come close enough to each other (geographically) then the button **ACCEPT/STOP CONFERENCE** appears to both sides. When a user presses this button, a sound is play-backed at the other side and an ACCEPT CONFERENCE event is recorded in the dialogue area. When both sides have accepted the conference, they can talk at real time using webrtc. Both sides can stop the conference at any time they wish pressing the appropriate button.

<u>FOR DEMONSTRATING THE APPLICATION, YOU SHOULD USE FAKE LOCATIONS.</u>

<u>YOU CAN WORK AS TEAMS OF TWO PERSONS, IF YOU LIKE.</u>

<u>IF YOU ALSO LIKE YOU CAN TRY IMPLEMENTING THE SAME WEB APP FOR MORE THAN TWO USERS.</u>