# EavesDrop-Live Podcasting

1
2
3
4
5
6
7
8
9
10
11
12
13
14
15
16
17
18
19
20
21
22
23
24
25
26
27
28
29
30
31
32
33
34
35
36
37
38
39
40
41
42
43
44
45
46
47
48
49
50
51
52
53
54
55
56
57
58
59
60

SCHOLARONE™
Manuscripts

# EavesDrop

**Granthali Jadhav**

Information Technology

V.C.E.T

jadhavgranthali@gmail.com

**Akshay More**

Information Technology

V.C.E.T

moreakshay725@gmail.com

**Hardik Yewale**

Information Technology

V.C.E.T

yewalehardik@gmail.com

**Prof. Sainath Patil**

Information Technology

V.C.E.T

sainath.patil@vcet.edu.in

*Abstract*— **Interactive Live streaming is the process of transferring real-time data over the internet. The data can be of type audio, video or any common data type. Interactive live streaming is an important feature of applications and platforms in which the actions of the audience affect the content of the communication.[1]Simultaneously, Podcasting is one of the types of contents that people are attracted towards, the idea of podcasting includes recording the audio or video file and uploading it to a platform where people can access it. Now, with the thought of mixing the above two, the idea of an interactive podcast was born. We analyse that low delay is the restrictions for Web Interactive Streaming.[1] The interactive communication over the internet with the restrictions of real-time boundaries requires a framework that will allow the development of the proposed project. Real Time Communication which means has a wide-industry areas. RTC is the standard and it also increases the browsing model which allows the access to live streaming systems which consists of Social Media, Television, Chatting Applications as well as the Communication Media. Users are allowed to read the comment, write the comment, record the sessions/audio/video, edit audio/video, edit audio/video which is done within a Cloud Infrastructure and also has Services of Quality**. **As things are in the perspective, the development of the application is the most important part where, using different technologies a system can be capable enough to stream audio and video data over the internet which can handle a said number of users which can make audio rooms and discuss a topic among themselves which can uniform where several people can hear them and they also have a chance to add value to the conversation by their inputs.**

**Keywords**— **Interactive, podcasting, Live-streaming, Real-Time Communication, time-critical, cloud.**

## I. INTRODUCTION

Podcasting, the conception of audio entertainment, where multiple People can hear the discussion, the host is having, thus, creating a window for a special type of content where the spectator becomes a listener because they do not need to look at the screen to enjoy the content. A new occasion is created for individual as the person doesn't need to check his phone, just play and hear to podcast whenever they want like doing work, going for a jog, etc. The rearmost social trends as well as the technology that advances this have resulted us to the emergence of varied famous web- grounded live streaming platforms, similar as Facebook Live, YouTube Live,

Instagram Live. These platforms are aimed to increase the scalability and are therefore Real-Time, this will also show high delay [1]. Now, one needs to understand that, the operation of settling a real- time medium isn't easy and is dependent on real- time architecture which allowed the inventors to produce a real- time terrain where multiple requests can be transferred from multiple sides. These requests can include questions from one end, acknowledgement from multiple lines, audio crimes, and numerous other effects that count during the live streaming of the data. To put effects into perspective, the operation of a real- time terrain can be done by using the RTC Framework, which is the real-time communication frame that allows the user to do the below effects. There live multiple fabrics that give these features and one of the most popular bones is Google's on Web RTC frame. The WEB-RTC which is open-source allows the users to view video content as well as comment on it while streaming it on real-time for communication. WEB-RTC allows us the real time communication including the API that helps to transfer voice and audio through WEB using JavaScript Code. Lately, New platforms were introduced for Real-Time that is it dispatches services browser bedded operation or web operation. Within this Operations, WEBRTC has stepped into similar interest as many new performances which supports API is common for all Browsers like Google Chrome, Mozilla Firefox,etc.HTML5 is supported by WEBRTC that establishes and holds the Connection of Peer and Media Stream as well as it can be for Chat Rooms using API to have P2P communication within peers [2]. Though the Web RTC frame works well, it comes with a handy fiscal cost which after exploration was found a probability to avoid. With the operation of the AGORA RTC Framework, which is less expensive and provides a healthy amount of original talk time, it was smarter. Now as we've talked about how we can apply the RTC frame along with live-streaming capabilities, the most important part of the operation resides in the development of the actual operation. The development of the operation was believed to be effective in FLUTTER as it supports both Android and IOS. Also, using the Firebase for Database and Server purposes was begin to be veritably effective as it handed the functionality which was necessary for the operation.

In 2010, Server Socket.io was created. Socket.io works in Two-ways that is it allows Bi-Directional Communication which is done between Server and Clients. The Server

Socket.io Server also has a huge community, which means chancing help is fairly easy. This allows the developer to open the connection and helps in real time communication which is a phenomenon of the time. This communication can be done only when Client holds Socket.io within the browser also Server should also have Socket.io package. Data can be transferred in the form of JSON. Now-a-days for Popular Web operations which is real-time uses PHP which is tradition method and is hard. It includes pooling the changes into server, having track for timestamps, etc. Sockets plays an important role in Real-Time Systems Architectures as well as Bi-Directional Communication within Clients through the Servers. It also says that server pushes the guests in the room. When a event is generated, the users are pushed to particular Guests. The Socket.io Server is very popular and is used by Zendesk, Yammer, Microsoft Office as well as Trello as well as many real-time applications. GitHub is the most important JavaScript which is depended on Node Package Manager Module i.e. NPM .

## II. PROBLEM STATEMENT

Transfer of audio data packets from the source to the destination but this follows the path of all the number of participators and those participators can hear the audio packets which eventually means they hear the discussion that the hosts are having and thus serves the idea of Podcasting. Participation of the followership in the discussion can make the participants feel like they're a part of the discussion and thus put a positive impact causing the hype of the operation through the audience themselves. The Live-Streaming System Solves the problem of few problems in the existing Interactive Live Streaming. One of them has the significance that the systems are Web based. Hence, at the last, there has been an important trend to shift towards the operations of the Web. Know few features are similar to multimedia, that had limited support traditionally. The operations that were totally dependent had to find other way and had to calculate the non-plugin Standards.[1]
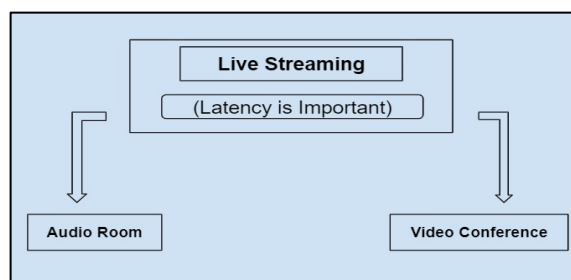


Fig. 1. Importance of Latency in Realtime Systems

## III. RELATED WORKS

In the Internet- based period, the works of similar systems are found in different aspects with the different purposes of development, as everything has its faults; the systems were constructed to make the user use the system more efficiently and accessibly. Thousands of content is posted on social media in the form of written content, videotape, but there's no scope of increase in audio content. There are hardly any platforms available to explore someone's audio skills. Colorful inquiries include system features like chat boxes, audio and videotape relations in live sessions. The exploration included the armature of WebRTC, Chat Box algorithm, different approaches for interactive communication and much

further. The Interactive relation through audio approaches is encouraged with the existing systems. Applications of The Live Interactive Streaming's are given mentioned below:

- Skype, Hangouts from Google, FaceTime from Apple helps in Video Conferencing
- Users can understand what is passed in the real world through Remote Rendering Systems.
- Surveillance systems: allows the client to send the video through server.

### A. ARCHITECTURE OF WEBRTC

The most important illustration of a videotape conferencing system is skype held by Microsoft before the actuality of WebRTC. WEBRTC provides Peer-to- Peer Browsing [4]. Peer Connection enables the stoner communication that is direct between all the users we can also say it is peer-to-peer communication or it can be browser-to-browser communication [7]. The framework allows P2P communication which is end-to-end translated for both audio as well as for video content and the data is transmitted. The framework of WEBRTC includes 3 parts:

1) layer for Web Developer

2) layer for the browser developer

3) Service layer:

This tier contains Video Engine, Voice Engine , Tools that helps in Transport Communication.

Voice Engine helps in Audio Processing where it includes Audio processing and decoding. Videotape Machine deals with image processing and codec. The transport point includes ciphering the collection of audio and videotape and transmitting it over SRTP protocol.

It's important to understand that the WebRTC isn't just a single API, rather it is APIs Collection which also includes protocols for working groups that is similar to W3C which is World Wide Web Consortium and IETF which means Internet Engineering Task Force. [2]

*Additional Features of WEBRTC:*

i) Streaming of Audio, Video or any data

ii) Report Errors to Coordinate Communication

iii) Understand the information about the Network.
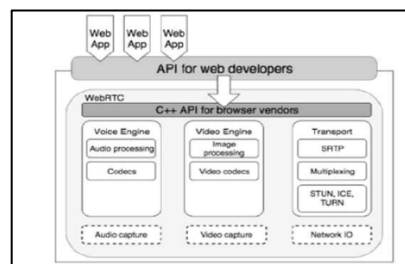
iv) communicate streaming audio, video, or data



Fig. 2. Architecture of Web RTC

### B. INTERACTIVE LIVE STREAMING APPROACH

We shall describe and dissect numerous of the colorful approaches available for web- grounded interactive live

streaming. One of them that is Refreshing Image and M-JPEG approaches and is frequently employed with infrastructures as well as execution on live-streaming process. Also, we include some new approaches which are more infrequently used, several of which have only lately come available thanks to their reliance on new under-development Web- Norms. The figure Explains the simplified working of Live Streaming Interaction from the time it's captured to which it's displayed. The inflow starts when the IP webcam captures a frame. Different IP webcam models live, and different models support different affair formats. Maximum used are JPG (separate images). The affair from the cameras is encouraged to, the streaming platform counting on the source and target formats through those formats. It may not be necessary to transcode it into a special format. Transcoding can take a big quantum of processing power and adds some quiescence. The server will store images and stream the for the people who are surfing over the Internet. Various Channels of Client Server Protocol are available .If the data is delivered the surfer will get the data that is handed and then calculates the native element it will help to reuse, render and crack through JavaScript.[1]
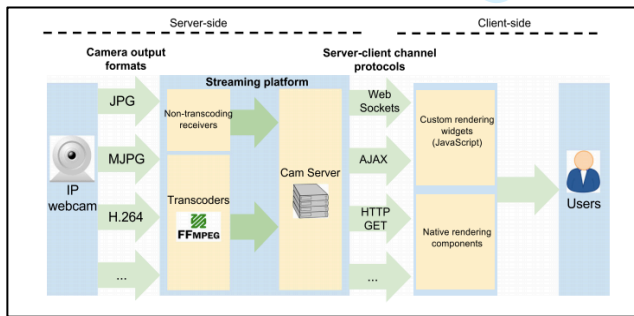


Fig. 3. LIVE STREAMING

## C. *CHAT ROOMS*

whenever, the user wants to prompt the knowledge for any content also the user can look for space by giving the keyword. A user searches for appropriate room and join the room.If no such room exists then one can create the room of appropriate name. After the users are joined within the dialogue room, users can decide whether the dispatches they shoot will be displayed alongside the sender name. If the user wants to be visible to other users also the dispatches are displayed with the username. If the user does not want the factual name to be visible also a singular id is generated as a username for that specific dialogue room and thus the dispatches are displayed with the generated name. Chatrooms can be deleted by Admin as well as a User who created it. The subsequent flowchart explains the dialogue room workflow [3]
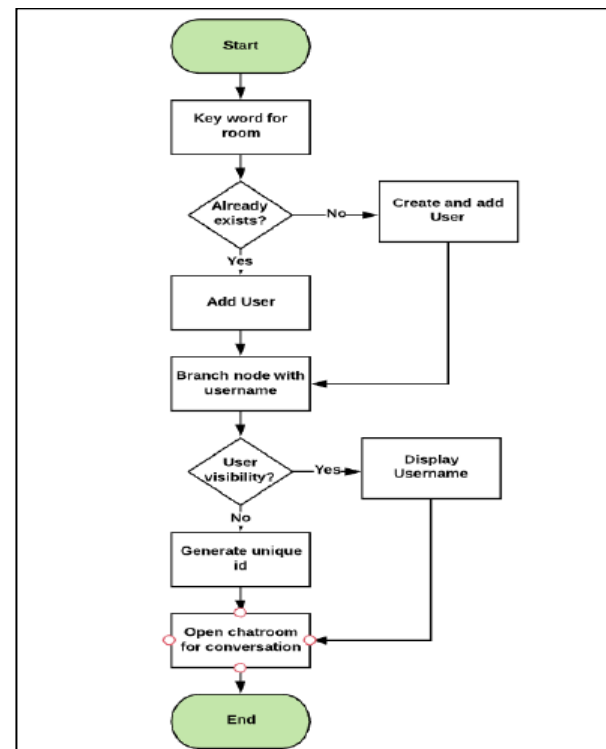


Fig. 4. Algorithm for Chatrooms

## IV. COMBINED STUDY

Listener audiences are increasing day by day. The users are increasing from Radio rather than Music. Until now, it was considered that radio listeners are decreasing day-by-day instead it is still increasing. [5] Fig 6 gives us the details of overall users listening to Podcasts till 2019. Fig 5 concludes that 63% of users listen to podcasts at home, 11% at work, 11% at car/truck,4% while walking and 8% others.[6]
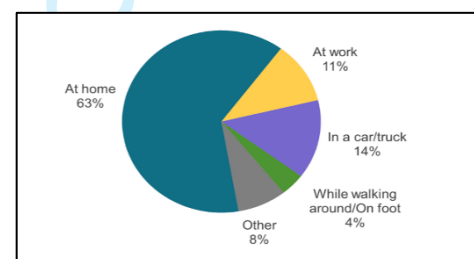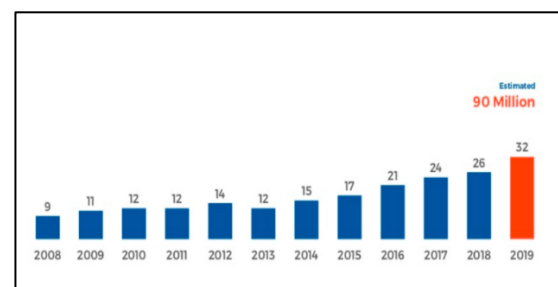


Fig. 5. Podcast Users



Fig. 6. Overall Listeners of Podcasts

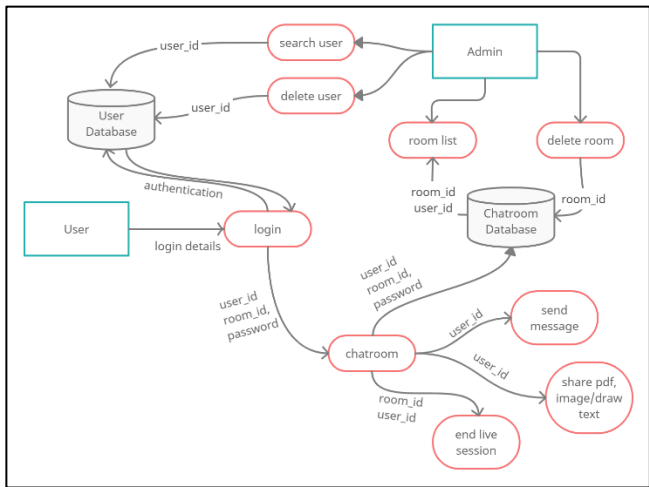## V.  PROJECT IMPLEMENTATION

### A.  Data Flow Diagram



Fig. 7. Dataflow Diagram

Data Flow Diagrams in simple words shows how the overall data flows within the system. As our system provides the user with the rights of creating a live chatroom, which can consist of 'n' number of participants, it assigns them with rights of logging in the system and then they can Share PDFs, send messages and also end the live sessions once they enter the chatroom.

The data flow for admin is also very simple, where all the data flows through the admin panel where the room list, user-ids, room-ids, and various other uniquely generated information is then forwarded to the database which the admin can access.

### B.  Activity Diagram

The below Activity diagram i.e. Fig 8. The diagram shows the flow of the system that is how it will work, What will be the result, what is the flow of the projects. The word "data" is plural, not singular.
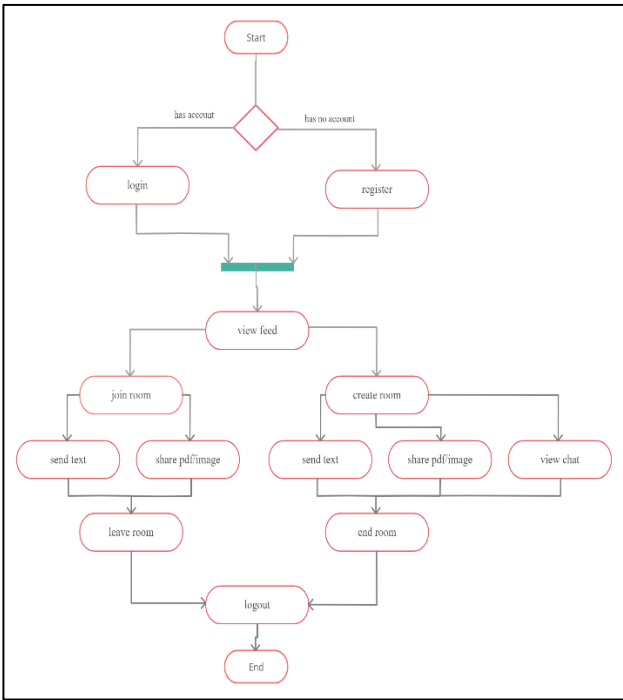


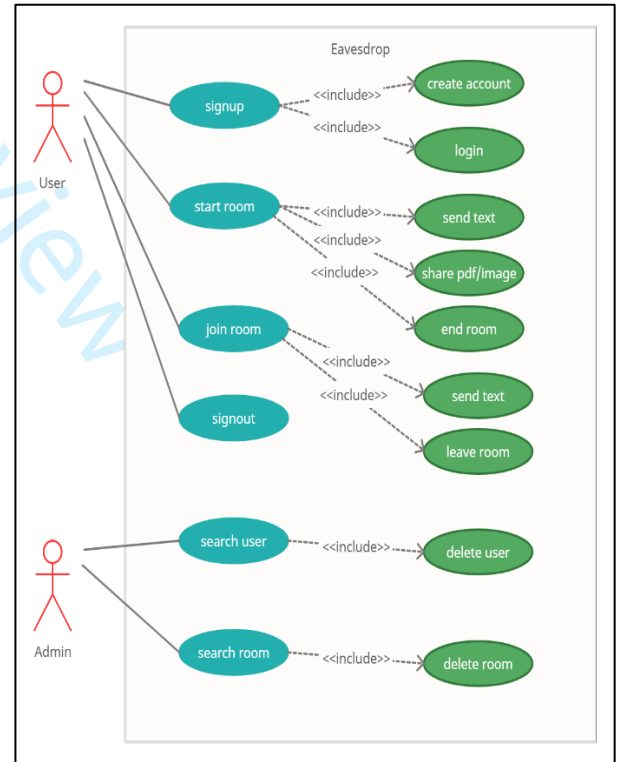Fig. 8. Activity Diagram

### C.  Use Case Diagram



Fig. 9. Use case Diagram

The above Use Case Diagram represents the users and their respective jobs, as well as the unique duties assigned to them which they have to carry out.
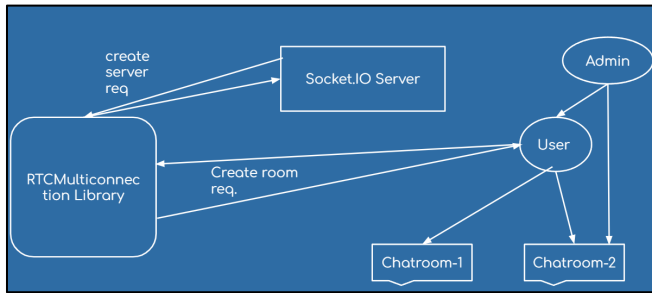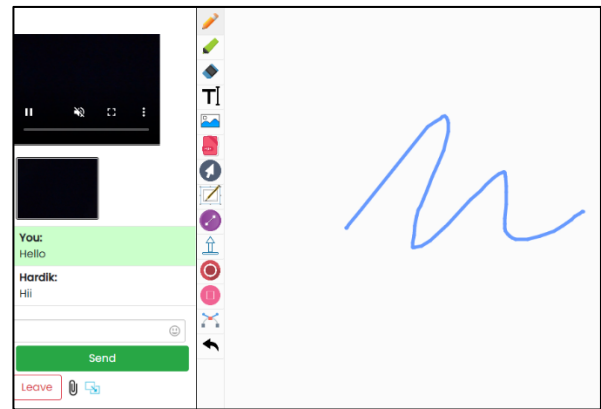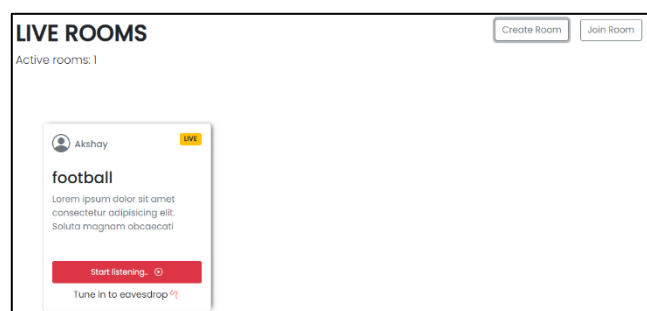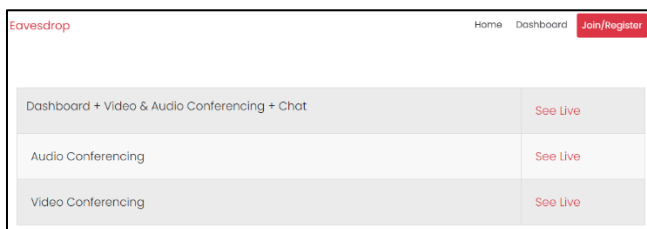
## D. Block Diagram of the System



Fig. 10. Block Diagram

As shown In this block diagram, there are four main components of the system.

- Server Socket.io
- RTC Multiconnection Library,
- User,
- Admin

The administrator holds together the user as well the chatrooms they create which obviously puts the admin in a very commanding position. The work of Creating Rooms, Handling them and Distributing them in two components i.e. Server Socket.io and the RTC Multiconnection Library respectively. The User when press create room button a request is processed from RTC Library which in return processes a request to the Server Socket.io that means it Requesting for Particular Socket on The Internet that it streams Audio and Video. The request when accepted is then processed towards the user which results in room creation and various ID generation.

## VI. RESULT SNAPSHOTS.









## VII. CONLCLUSION

The system will result as a real time communication application for a user so that they communicate with ease to make sessions more interactive. The User can create their Chatrooms for exchange of content and information. This Day-by-Day the video Conferencing Technology is increasing due to rise of COVID-19. Currently System includes the audio and video medium of interaction with audience. Where audience interaction plays the negligible role. With understanding Related Works and demerits of existing systems the system is designed. The application includes three main features

1. Live streaming
2. Audience Interaction
3. Chatrooms

The goal is to implement the EAVESDROP streaming application with audio, video, live facilities for the Audience, where WEBRTC helps the developer to build Better Real-Time Communication Systems. The System can be further improved, like streaming the audio interactive sessions on various platforms and much more.

## REFERENCES

[1] Rodriguez-Gil, L., Orduña, P., García-Zubia, J. *et al.* Interactive live-streaming technologies and approaches for web-based applications. *Multimed Tools Appl* 77, 6471–6502 (2018). https://doi.org/10.1007/s11042-017-4556-6

[2] G. Suciu, S. Stefanescu, C. Beceanu and M. Ceaparu, "WebRTC role in real-time communication and video conferencing," 2020 Global Internet of Things Summit (GIoTS), Dublin, Ireland, 2020, pp. 1-6, doi: 10.1109/GIOTS49054.2020.9119656.

[3] https://www.ijrte.org/wp-content/uploads/papers/v7i5s2/ES2063017519.pdf

[4] http://www.scirj.org/papers-0715/scirj-P0715273.pdf

[5] https://techcrunch.com/2019/08/21/after-a-breakout-year-looking-ahead-to-the-future-of-podcasting/

[6]   https://www.edisonresearch.com/the-podcast-consumer-canada-2018/

[7]   Duan Quanfeng and Liang Zhenghe

[8]   Amaral V., Lima S.R., Mota T., Chainho P. (2014) Exploring WebRTC Technology for Enhanced Real-Time Services. In: Rocha Á., Correia A., Tan F., Stroetmann K. (eds) New Perspectives in Information Systems and Technologies, Volume 2. Advances in Intelligent Systems and Computing, vol 276. Springer, Cham. https://doi.org/10.1007/978-3-319-05948-8_5