VISVESVARAYA TECHNOLOGICAL UNIVERSITY

"JNANA SANGAMA", BELAGAVI-590018, KARNATAKA

Project Report On

"eKaksha – Live Video Conferencing Application"

Submitted in the partial fulfillment of the requirement for the award of degree of

BACHELOR OF ENGINEERING
in
COMPUTER SCIENCE AND ENGINEERING

Submitted By

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DEPARTMENT OF COMPUTER SCIENCE AND ENGINEERING

SAI VIDYA INSTITUTE OF TECHNOLOGY

(Affiliated to Visvesvaraya Technological University, Belagavi | Recognized by Govt. of Karnataka | Approved by AICTE, New Delhi)

Accredited by NBA, New Delhi (CSE, ISE, ECE, MECH, CIVIL)

RAJANUKUNTE, BENGALURU – 560 064

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CERTIFICATE

Certified that the Project Phase -1 work entitled ***"EKAKSHA – LIVE VIDEO CONFERENCING APPLICATION"*** carried out by**), Shubham Kumar(1VA18CS046), Mohd. Asad Qatib(1VA18CS024), Chirag Patidar(1VA18CS008) and Saikat Maity(1VA17CS043)** a Bonafide students of **SAI VIDYA INSTITUTE OF TECHNOLOGY**, Bengaluru, in partial fulfillment for the award of Bachelor of Engineering in Computer Science and Engineering of **VISVESVARAYA TECHNOLOGICAL UNIVERSITY**, Belagavi during the year **2021-22.** It is certified that all corrections/suggestions indicated for Internal Assessment have been incorporated in the Report deposited in the departmental library. The Project Phase – 1 report has been approved as it satisfies the academic requirements in respect of Project Phase – 1 work prescribed for the said Degree.

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Project Phase – 1 Marks Allotment

Introductory (20)	Intermediate (20)	Final (60)	Total (100)	Signature of Guide

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ABSTRACT

EKaksha – Live Video Conferencing Application is a web application using which the user can attend the live classes in virtual manner. It can also be used by individuals to connect to other peers in real time.

The next feature is to capture the attendance who are attending based on this list the attendee will be able to download their certificate.

EKaksha – Live Video Conferencing Application is a great solution for the pandemic world. EKaksha – Live Video Conferencing Application is an all-in-one solution to virtually connect to large number of people or individuals. The key idea of virtual connect software EKaksha – Live Video Conferencing Application is to allow users to create, host, and manage live events.

Depending on the size of the company and business demands, the number of attendees may be different. Therefore, hosting an online event with Virtual Event Software is even easier than organizing an offline event.

You need to create an event and become its host, live discussions, and video calls on the same platform.

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CHAPTER 1

INTRODUCTION

As everyone is aware of the pandemic condition that is currently in place, social estrangement is the most important course of action to take. So, utilising an application, we can communicate with consumers via live video calling. When you choose to meet the person who wants to get, it provides the best option and remains at home. Therefore, a video calling software is urgently needed.Types of Events that can be conducted in EKaksha – Live Video Conferencing Applications.

Webinars, Online classes, Conferences, Broadcast to the event, Workshops, Panel discussions,meetings, Q&A, Live streaming.

For both hosts and participants, EKaksha - Live Video Conferencing Application is a one- stop shop. The user base of this platform is able to locate and participate in any events pertinent to their interests. However, this platform is used to organise events and attend meetings on the same platform, rather than only discussing these interests online.

The application will be having a user-friendly interface that is not only very easy to use but also adds to the user's readability. It is a fundamental rule to ensure that the application is easily learned, experienced and used. This includes both the visual and behavioral aspects ofthe design.

With a flexible online community, we hope to enable users to discuss their interests, find others who share them, and participate in activities that are related to those interests. We are taking these steps to improve the current system, which makes it difficult for users and event organisers to interact with one other at the same time. It is an excellent tool for networking, getting more involved in your community, and pursuing an interest you are passionate about.

1.1 Problem Statement

The Web Real-Time Communication (WebRTC), WebSocket protocol and Node.JS We have todevelop a live video calling application right from scratch which is easy, secured, fast and very reliable to use. The main aim is to design0and implement. WebRTC video conferencing between browsers in real-time. Also, an evaluation of CPU performance, bandwidth consumption and Quality of Experience (QoE) must be taken into consideration. The best feature of any highly complex software is that it should be user-friendly, and we too will take proper care of it.

1.2 Solution for the problem

EKaksha – Live Video Conferencing Application is a great solution for the pandemic world. EKaksha – Live Video Conferencing Application is an all-in-one event management platform for conducting online events.

EKaksha - Live Video Conferencing Application's main goal is to enable users to plan, host, and manage live events. The number of guests may vary depending on the extent of the host's requirements. A virtual event management system makes holding an online event even simpler than. We need to create an event and become its host later chat and perform live discussions, and video calls on the same platform

Everything happens in real-time the speaker or host streams the event; the attendees get engaged in the discussion via the platform

The host will create an event and share the meeting ID with all the attendees and attendees will enter that particular ID to connect the corresponding chat room.

1.3 Existing Techniques

Zoom

Zoom is a cloud-based video conferencing service you can use to virtually meet with others either by video or audio-only or both, all while conducting live chats - and it lets you record those sessions to view later.

Features:

- Screen-sharing to present documents, spreadsheets, presentations,
- Two-way and multi-way audio and video calls
- An accompanying chat

Disadvantages:

- Attendees can't search for their interesting event or meeting and join
- Only if the link/Meeting ID is present with the user only then he can attain the eventor the meeting
- User privacy issues has been arising lately.

Cisco WebEx

Teams can stay connected using the cloud-based productivity toolset Cisco WebEx. The package enables unified communications for any organisation, from SMBs to enterprise-wide demands, with video meetings, file sharing, and team messaging. With WebEx, either the Cisco WebEx Meetings or WebEx Teams apps may support all internal meetings.

Features:

- Two-way and multi-way audio and video calls
- An accompanying chat
- Screen-sharing to present documents, spreadsheets, or (if using a browser) other browser tabs

Disadvantages:

- Attendees can't search for their interesting event or meeting and join
- Only if the link/Meeting ID is present with the user only then he can attain the event

Microsoft Teams

Microsoft Teams is a platform for collaborative work that uses persistent chat and includes several incredibly practical capabilities for business interactions, such as document sharing and online meetings.

Features:

- Call encryption between all users
- Screen-sharing to present documents, spreadsheets, presentations,
- Two-way and multi-way audio and video calls
- Ability to raise and lower hand

Disadvantages:

- Attendees can't search for their interesting event or meeting and join
- Only if the link/Meeting ID is present with the user only then he can attain the event or the meeting

Google Meet

Google Meet is a video conferencing service from Google. It's a great solution for both individuals and businesses to meet on audio and video calls. It was born from Google Hangouts but boasts some unique features

Features:

- Two-way and multi-way audio and video calls
- An accompanying chat
- Call encryption between all users
- Screen-sharing to present documents, spreadsheets, presentations, Ability to raise hand

Disadvantages:

- Attendees can’t search for their interesting event or meeting and join
- Only if the link/Meeting ID is present with the user only then he can attain theevent or the meeting

WhatsApp

WhatsApp is an application that is free to download messenger app for mobile phones. WhatsApp uses the internet to send messages, images, audio, or video. The service is very similar to the other text messaging services

Features:

- Send text messages and voice messages
- Make voice and video calls
- Share images, documents, user locations, and other content

Disadvantages:

- The user has to be in contact list to be added to a group.
- Limited number of people can be added to a group.

Google Duo

Google Duo is an application developed by Google which allows users to make audio and video calls from one user to another. Google duo unlike Apple’s FaceTime can be used on both Android and iOS devices. There is also Google Duo for PC if you use your computer frequently. Google Duo is working like Facebook’s messenger. If both parties are in the Google Duo then through the internet they can contact each other anytime.

Features:

- Offers very high-quality video calls
- Knock Knock services
- Can work on various devices
- Secured communication

Disadvantages:

- The sign up is mandatory
- No way to reject a call with a message
- Some Wi-Fi connectivity issues in testing
- No way to accept a call without video

Skype Business

Skype Business is paid version of skype which is used by many companies and organizations around the globe. This is because it offers all the customization any big organizations need for its employees.

Features:

- It is almost ubiquitous (over 600 million subscribers) and it's easy to deploy (assuming yourIT policy doesn't prohibit it) you can be up and running within minutes.
- It is user friendly, even an inexperienced users can easily use the application

Disadvantages:

- Skype does not work well for multi-party calling
- Does not support remote desktop while video conferencing
- Should not be using Skype to view several people in a meeting room
- Not clear audio for multiple people in audio conferencing

GoToMeetings

GoToMeeting, a web conferencing software as a service solution created by Centrix, enables you to host business meetings online. GoToMeeting makes it simpler to convey demands instantly over the globe if you have employees, contractors, or clients who are in different timezone.

Features:

- It allows users to have full control over the equipment of a participant.
- You can share documents in real-time.
- All you need is a data connection to have a meeting
- It permits large meetings
- You don't need to attend to experience the meeting

Disadvantages:

- It requires a stable internet connection
- The quality of the equipment matters
- It can be difficult to get the call-in process right
- There is a cost to access the premium features you may require.
- It can take up a lot of bandwidth

Google Meet

Google Meet is a platform for video conferences. The ability to meet via audio and video chats is a fantastic solution for both individuals and corporations. Although it was derived from Google Hangouts, it has various distinctive features.

Features of Google Meet:

- Two-way and multi-way audio and video calls
- An accompanying chat
- Call encryption between all users
- Screen-sharing to present documents, spreadsheets, presentations,
- Abilityto raise and lower hand

Disadvantages:

- No file sharing during the meeting
- Only if the link/Meeting ID is present with the user only then he can attain the event
- Any random user can join the meeting using the link
- No attendance system

1.4 Proposed Technique

EKaksha – Live Video Conferencing Application is a web application using which the user can connect in virtual manner.

Our application will have some additional features. We will implement exam mode, in which host can conduct exam virtuallyand can keep an eye on malpractices practiced byanyattendee.

The next features we will add are screen sharing, screen recording, file sharing, whiteboard and text-chat and all these will work in real time with minimum latency.

The most important feature is that it will be end-to-end encrypted and will have facial attendance.

An additional plan is to add push notification.

1.5 Objective

As from the past few months, there is an increase in the virtual webinar and events which are conducted byvarious organizations and colleges from different departments on trending topics but the problem is many are using different platforms to share their details like WhatsApp, Facebook, Instagram, etc. but it is not reaching the maximum number of users(attendees) Also users must download various video conferencing apps like google meet, zoom, teams, etc. for different webinars. Every time the user wants to connect, he/she just have to initiate the even and share the meeting ID.

1.6 Scope of the project

The data is very accurate and authentic as we take all the data from users and hence the application is reliable for the users. The user can attend or also conduct virtual sessions. Attendee can attend the meeting or conference from anywhere in the world. user can explore all the events, webinars, workshops, courses, etc. Teacher can conduct their classes at their convenience and need not take attendance manually.

1.7 Organization of Report

The rest of the report is organized as follows:

1. Chapter 2 contains a literature survey on:
2. Zoom, Cisco WebEx, Microsoft Teams, Google Meet, WhatsApp,
3. Chapter 3: Requirements Specification
4. Chapter 4: Gives the complete system design analysis and design, it explains the high-level design of the project, the block diagram, high level design, and low-level designs.

CHAPTER 2

LITERATURE SURVEY

2.1 Zoom

Zoom is a cloud-based video conferencing service that enables you to hold live discussions while digitally meeting with others. Zoom also allows you to record those meetings for later viewing.

A proprietary video telephony software product created by Zoom Video Communications is called Zoom Meetings (sometimes abbreviated to Zoom and stylized as zoom). The free plan has a 40-minute time limit and supports up to 100 concurrent users. Users can choose to upgrade by signing up for a premium plan. The most expensive package allows for meetings to run up to 30 hours and up to 1,000 people at once. Zoom was heavily used for distant work, distance learning, and online social interactions during theCOVID-19 epidemic. the growth

Features:

HQ audio and video

The video conferencing app from Zoom can broadcast HD video and audio at high quality, providing participants excellent picture quality. Although most high-speed internet connections can stream high-quality video, you can downgrade the picture quality if you are having trouble getting a stable connection.

Observer camera feed

The main function of Zoom is video conferencing. Participants have the option to share their video but are not required to. Zoom will request authorization to use your computer or device's camera before using the camera to show other callers video.

Only-Audio conferencing

Zoom can be used for audio-only conferences as well, even though it is primarily advertised as a video-conferencing solution with many video-related features and advancements. Simply disable video when attending a meeting and continue using the app normally without sharing any footage. Additionally, you can utilise your phone to join Zoom meetings.

Encryption from end to end

AES-256 encryption is used for shared content, while 256-bit TLS encryption is used for communications. Zoom is working to implement end-to-end encryption as part of its security updates in May of 2020.

Encryption (E2EE) is a feature that may be turned on and off. While end-to-end encryption will be accessible for all Zoom meetings and services, some communication platforms will perform better with this encryption turned off.

Records of meetings

You can locally record meetings on your computer. You can upload files to a file storage service like Google Drive, Dropbox, or any alternative to share these files with others. Meeting recording privileges can be restricted or extended to attendees and other Zoom users by the meeting host.

Attention Signal

The host (and co-hosts) can keep an eye on how attendees are interacting and paying attention during meetings and webinars by using the Attention Indicator function..

Texting immediately

Within the Zoom Chat app (desktop, mobile, and internet versions) and in-meetings, you can use their instant messaging service. Converse with attendees of meetings and other Zoom users. Participants can successfully communicate during meetings using this without disturbing the presenter. Outside of calls, Zoom Chat can be used for instant messaging. This can be used by teams for effective communication that is simple to record. Other team members can use this as a tool to pass along instructions because they have a record of your request.

Disadvantages:

- Attendees can’t search for the event or meeting of their interest
- The meeting ID must present with the user

2.2 Cisco WebEx

A cloud-based set of productivity tools called CiscoWebEx keeps teams linked. The package enables unified communications for any organisation, from SMBs to enterprise-wide demands, with video meetings, file sharing, and team messaging. With WebEx, either the Cisco WebEx Meetings or WebExTeams apps may support all meetings inside a corporation.
An American firm called Webex by Cisco creates and markets applications for online conferencing, videoconferencing, unified communications as a service, and contact centres. It was started in 1995 under the name WebEx and was acquired by Cisco Systems in 2007.

Features:

Full HD Video Conferencing

Six people can simultaneously participate in HD video conferencing with Cisco WebEx. Each conference participant has a personal video feed that is accessible in the private meeting space. Using Active Speaker technology, the main visual stream automatically changes to the speaker. It makes keeping track of talks simple.

The video conferencing software also works with any USB-connected webcam or camera. Regardless of their hardware or operating system, everyone will be able to participate in the debate.

Platform compatibility as well as data security

Cisco WebEx allows simple cross-platform and practically universal device operation, regardless ofwhether teams are using desktop or mobile, Windows or iOS applications.

When sharing documents and holding private conversations, Cisco's unmatched secure data encryption, conference password protection, and network security measures keep businesses safe from crucial data leakage.

Utilizing Cisco also offers the advantage of utilising its globally dispersed data centres. This means that virtually no network lag will result in video disruptions during any video conference sessions, which will all feature high-bandwidth transmissions.

Convenient online meetings at any time and location

Companies with employees from different parts of the world and clients from other countries are constantly seeking for ways to improve communication and reduce unneeded travel costs. Cisco WebEx makes it possible for staff members and clients to communicate and share presentations on a single digital platform, maintaining relationships and projects on schedule and within budget. Additionally, it lessens the number of emails and voicemails that go unanswered, which slow down work and weaken outcomes.

Comprehensive Communication

Even those without visual capabilities can easily join video conference sessions thanks to Cisco WebEx. One of the extra features of the software is teleconferencing, which enables participants to request call-backs for simple admittance into the virtual conference.

Meeting participants can easily connect through VoIP and participate in audio-only sessions using built-in computer or mobile device microphones, headsets, or both. The web interface gives the conference host complete control over all interactions, including the ability to mute specific members or the entire group if necessary.

Desktop and document sharing

The ability for users to quickly share their desktops or documents in real time is one of Cisco WebEx's most potent capabilities. The WebEx conference host can provide specific sharing permissions for programmes or documents displayed on their screen. After that, they might assign a new degree of anyone in the group is given control. Additionally, a virtual whiteboard can be used to construct note-taking and brainstorming tools with annotations. The requirements of business owners are always changing. But the significance of implementing effective team collaboration systems doesn't change. Cisco WebEx offers a sophisticated way for organisations to stay in touch with their staff and customers no matter where they are, what they are using, or when.

Disadvantages:

- Users can’t search for their interesting meeting or event andjoin
- Link/Meeting ID is a must for the user to connect and join the event or a meeting.

2.3 Microsoft Teams

Microsoft Teams is a platform for collaborative work that uses persistent chat and includes several incredibly practical capabilities for business interactions, such as document sharing and online meetings.

The Microsoft 365 family of products includes the proprietary business communication tool known as Microsoft Teams. Teams offers workspace chat and videoconferencing, file storage, and application integration, and largely competes with the similar service Slack. Teams will eventually take the place of Microsoft's current enterprise communications and collaboration tools, like Skype for Business and Microsoft Classroom. As more meetings moved online during the COVID-19 pandemic, Teams and other programmes like Zoom and Google Meet attracted a lot of attention. It has over 250 million monthly users as of 2021.

Features:

Screen sharing and online video calling

Take advantage of smooth and quick video chats with staff members inside your company or clients outside your company. A collaborative platform would benefit greatly from having a good video call capability. Additionally, one can take advantage of quick and easy desktop sharing for multi-user real-time collaboration and technical support.

Online conferences

Through the use of an online meetings function that can accommodate up to 10,000 participants, this feature can improve your communications, company-wide meetings, and even training. Anyone inside or outside of a firm can participate in online meetings. A scheduling tool, a note-taking software, file uploading, and in-meeting chat messaging are also included in this function.

Audio conferencing

This is a feature you won’t find in many collaboration platforms. With audio conferencing, anyone can join an online meeting via phone. With a dial-in number that spans hundreds of cities, even users that are on the go can participate with no internet required. Note this requires additional licensing.

A chat functions

A basic chat feature that enables communication between teams, groups, and individuals is present in the majority of collaborative apps.

Keeping documents in SharePoint

Every Microsoft Teams team will have a site in SharePoint Online that comes pre-configured with adocument library folder. Across all discussions, all shared files will be automatically saved to this location. For critical information, permissions and security options can also be modified.

Disadvantages:

- Users can’t search for their interesting meeting or event andjoin
- Link/Meeting ID is a must for the user to connect and join the event or a meeting.

2.4 Google Meet

Google Meet is a platform for video conferences. The ability to meet via audio and video chats is a fantastic solution for both individuals and corporations. Although it was derived from Google Hangouts, it has various distinctive features.

Google developed Google Meet, a video-communication service that was formerly known as Hangouts Meet. It and Google Chat together make up the two apps that take the place of Google Hangouts. On mobile devices, it will also take the place of the user-facing Google Duo.

With 100 million users accessing Meet every day during the COVID-19 pandemic, as opposed to Zoom's 200 million daily users as of the last week of April 2020, the use of Meet increased by the factor of 30 between January and April 2020.

Features:

Templates for Video Conferencing

People are attending video meetings more frequently, which means that they are gathering in conference rooms less frequently. Google Meet already allows you to view up to 16 people simultaneously.

But you can now use several layouts that allow you to simultaneously see up to 16 people and the supplied content in order to create the most accurate representation of a real-time meeting. This will soon be expanded even more, allowing 49 participants to be shown simultaneously.

Hands raised

It frequently happens during video calls that people speak at the same time or are unable to appropriately interrupt due to the conversation's flow.

To help the conversation flow more easily when someone has a question or something to say, Google is adding the "Raise hand" button..

Capture a video conference

Video calls can be recorded by moderators for subsequent viewing by others.

Take a poll during a video call.

You have the ability to design polls as the meeting moderator. A report of the poll results is automatically sent to the moderator via email following the meeting. The report includes the participants' names and responses.

Limiting the length

Individual and group calls can last up to 24 hours between users.

Disadvantages:

- Users can’t search for their interesting meeting or event and join.
- Link/Meeting ID is a must for the user to connect and join the event or a meeting

2.5 WhatsApp

For smartphones, WhatsApp is a free communication app to download. When sending messages, photos, audio, or video, WhatsApp uses the internet. The offering is extremely comparable to text messaging services.

Features:

- Send text messages and voice messages.
- Make voice and video calls.
- Share images, documents, user locations, and other content

Disadvantages:

- If the user is not in the contact list or a group the details cannot be reached to thatuser.

26 Google Duo

Google developed the Google Duo software, which enables users to make audio and video chats to other users. Unlike Apple's FaceTime, Google Duo may be used on both iOS and Androidsmartphones. If you frequently use your computer, you may want to consider Google Duo for PC. Google Duo functions similarly to Facebook Messenger. If both participants are using Google Duo, they can communicate with one another at any time online.

Features:

Make dependable, safe calls.

When you can't be together in person, private, high-quality video calling is essential. Your video calling experience will be crystal clear and uninterrupted thanks to end-to-end encryption provided by Duo, and we're always making improvements. Duo currently makes use of AI to lessen audio hiccups, and next week we'll be introducing a new video codec technology to boost video call quality and dependability even on extremely slow connections.

Utilize the Picture-in-Picture (PiP) mode of Duo.

You can utilise another app while using PiP mode to minimise your video calls to a smaller screen. This only functions on iPhones running iOS 14 or later and Android smartphones running Android
8.0 or higher.

Tap the home button or swipe up from the bottom while you are on a video call. You'll only see asmall window during your video call.

Utilize the Picture-in-Picture (PiP) mode of Duo.

With PiP mode, you can minimize your video calls to a smaller screen and use some other app simultaneously. This only works on Android smartphones running Android 8.0 or above, and iPhones running iOS 14 or above. While on a video call, tap the home button or perform a swipe up gesture from the bottom.Your video call will minimize to a small window.

Make Use of Duo's Data Saving Mode

For users with metered data connections, Google Duo provides a Data Saving mode. By lowering the video quality during video calls, it saves data. Google Duo automatically selects a 720p resolution for HD video calls.
Open Duo and tap the button with three dots in the top-right corner to activate Data Saving mode. Next, activate the Data Saving mode by going to Settings > Call Settings.

Disadvantages:

- Fewer features
- High Data usage

2.7 Skype Business

Many businesses and organisations utilise Skype Business, a commercial version of the software, all around the world. This is due to the fact that it provides all the employee customization that big businesses might possibly require.

Through Skype, users may message people on their contact list instantly. The recipients of messages sent to offline users will receive them as soon as they connect to Skype online because the messages are saved on Skype servers. When a user joins in with the same Skype account, their chat history and message status are synced across all of their supported devices.

Features:

Customize Participant Views

Both the hosts and the participants can take advantage of intriguing features offered by Skype for Business. Anyone you wish to participate in your business meeting through phone or video conferencing can join and see it using Skype for Business..

- Outsiders can watch both you and the meeting content when using gallery view.
- You can view the content of your meetings using Presentation View..
- You can display the images of participants using Compact View..
- Speaker View allows visitors to present their views on meeting content.

Broadcasting

You can broadcast meeting content to up to 10,000 attendees using the Skype Meeting Broadcast feature. It is simple to distribute material to a huge audience because you may adjust the viewing count whenever you want to suit your demands.

Recording Feature

This is yet another amusing trick that Skype For Business provides. The presenters have the ability to record a variety of meeting elements, including handouts, sharing sessions, video, and audio. Additionally, you can choose the recording's resolution..

Meeting Content Retention

Users of Skype for Business can upload a tonne of business meetings in the form of PPTs, Excel files, notes, etc. thanks to meeting content retention. The type of meeting determines how long its contents are retained..

Disadvantages:

- Skype does not work well for multi-party calling
- Does not support remote desktop while video conferencing
- Should not be using Skype to view several people in a meeting room
- Not clear audio for multiple people in audio conferencing

2.8 GoToMeetings

GoToMeeting, a web conferencing software as a service alternative created by Cintrix, enables you to host business meetings online. GoToMeeting makes it simpler to convey demands instantly over the globe if you have employees, contractors, or clients who are in a different time zone.

A group of computers connected to a host computer over the Internet can access the desktop of the host machine via GoToMeeting. High-security encryption and optional passwords secure transmissions. Transmissions can get past extremely limiting firewalls by using a web-hosted subscription service along with host computer software..

- It gives users complete control over a participant's equipment.
- Documents can be shared in real-time.
- All you need is a data connection to have a meeting
- It allows large meetings
- You don’t need to attend to experience the meeting

- **Screen sharing**
  - Easily share the desktop, smartphone or tablet screen with anyone and everyone. Present,collaborateand keep everyone on the same page.

- **Meeting Drawing Tools**
  - Draw on your screen to collaborate, brainstorm or present in real-time.

- **Commuter Mode for mobile meetings**
  - While travelling, call in with no interruptions. Commuter mode uses up to 90% less bandwidth byturning your phone's full screen into a color-coded audio/mute button. our mouse or keyboard.
- **Custom backgrounds**
  - Customize your webcam background and truly work from “anywhere.”

- **Presenter control**
  - Assign presenter control to meeting attendees so they can share their screens and participate inmeeting management. Even your keyboard and mouse can be shared.

Disadvantages:

- It requires a stable internet connection
- The quality of the equipment matters
- It can be difficult to get the call-in process right
- There is a cost to access the premium features you may require.
- It can take up a lot of bandwidth

2.9 IEEE Paper (Design and Evaluation of Browser-to-Browser Video Conferencing inWebRTC)

WebRTC was developed as a new standard to facilitate RTC between users over different web browsers without the requirement for further installation. Due to the incompatibility problem with multi-browser connections, WebRTC is difficult to implement and prevents WebRTC from functioning effectively. This is because WebRTC does not specify the communication standard0between different browsers.
It overcame the aforementioned issue by using the WebSocket protocol as a server to fully communicate between two different browsers. It also created and put into use WebRTC video conferencing, which allows for two-way communication and makes use of a number of networks, including wired and wireless LAN and WAN networks. evaluation of the

physicalImplementation was prioritised over CPU usage, bandwidth use, and QoE. In the future, we'll try to use Socket.io to build a WebRTC signalling system that supports an infinite number of peers and can be used with mesh and star topologies. Will also contrast WebRTC videoconferencing with the most used VoIP protocols, including SIP or IAX2 protocols.

Signalling

The signalling implementation in this study makes advantage of the WebSocket mechanism. Four different sorts of control messages were created in our signalling system: initiator, getting media stream, peerChannel, and exchange SDP. Peer A will initially submit the server a "request," and once it receives the control message "getting access media stream," it will produce the user media on the server's behalf. Peer A now watches for Peer B's response. Peer B will send a response signalling message up until the point at which it is activated, and when it realises that it is not the initiator, it will first get "peer Channel" and then "got access media stream." Following that, a peer-to-peer connection can be established between the two parties. The SDP (Session Description Protocol) is used to transport the offer and response messages.

Client application

A test-bed lab was built to create WebRTC video conference apps as clients and servers. The linkwas established via the Local Area Network and Wide Area Network (wired and wifi). This implementation's main HTML (web page) has been built and set up to support a number of functionalities, such as starting a conversation between two different users (PCs) using Chrome, pausing music or movie, using full-screen, adjusting the volume, and taking a screenshot. Before a connection was established, local and remote descriptions were exchanged (the audio and video media information).

The following describes the signalling offer and answer exchange using the SDP (Session Description Protocol). Peer A first uses the "setLocalDescription()" function to set the local description before calling the RTCPeerConnection construct Offer() method and sends this session description to Peer B via the signalling server. Second, Peer B sets the description that Peer A gave as a remote description using the "setRemoteDescription()" method. Peer B runs the RTCPeerConnection "createAnswer()" method first, then uses the "setLocalDescription()" method to set the local description, sends Peer A the session description, and finally executes the RTCPeerConnection "createAnswer()" method. After receiving the session description from Peer B, Peer A uses the "setRemoteDescription()" method to set it as the remote description.

Quality of Experience (QoE)

Actual users who responded to questionnaires to express their individual viewpoints on the perceiveduser experience were a part of this implementation. The audio and video quality were excellent, it was found. WebRTC was developed as a new standard to facilitate RTC between users over different web browsers without the requirement for further installation. Due to the incompatibility problem with multi-browser connections, WebRTC is difficult to implement and prevents WebRTC from functioning effectively. This is because WebRTC does not specify the communication standard between different browsers. In order to completely connect with the devices in this study, the WebSocket protocol was employed as a server.

Two unique browsers Additionally, it developed and implemented WebRTC video conferencing, which enables two-way communication and utilizes various networks, including (Wired and Wi-Fi) LAN and WAN networks. The physical implementation was thoroughly assessed in terms of CPU performance, bandwidth usage, and QoE. In the future, we'll try to use Socket.io to build a WebRTC signaling system with limitless peers that we can use with mesh and star topologies. We'll also contrast WebRTC video conferencing with the most widely used Voice over Internet Protocol (VoIP) protocols, such SIP or IAX2..

Experimental Result

As a new standard for supporting RTC between users through various web browsers without the need for additional installation, WebRTC was created. The fact that WebRTC does not specify the communication standard between various browsers makes it difficult to implement and hinders WebRTC from working properly owing to the incompatibility issue with multi-browser connections.

In order to address the aforementioned issue, this article used the WebSocket protocol as a server to fully interact between two different browsers. Additionally, it created and implemented WebRTC video conferencing, a two-way communication tool that makes use of LAN and WAN networks (Wired and Wi-Fi) as well as other networks. The CPU performance, bandwidth use, and QoE of the physical implementation were all carefully evaluated. In the future, we'll try to use Socket.io to create a WebRTC signalling system that can be used with mesh and star topologies and supports an infinite number of peers. We'll also compare the most widely used Voice over Internet Protocol (VoIP) protocols, such as SIP or IAX2, to WebRTC video conferencing.

2.10 IEEE Paper (The WebRTC-based Peer Assisted Framework for HTTP LiveStreaming)

The main method for broadcasting live video material on the Internet is now HTTP-based adaptive streaming technologies. The WebRTC stack has made it possible to developAn entirely browser-based, plugin-free method of using peer-to-peer upload bandwidth to support and scale HTTP-based streaming applications. As a result, we've developed a peer- assisted WebRTC framework for HTTP live streaming. We have also empirically assessed its functionality with various peer densities. Our test findings show that by fulfilling segment requests from nearby peers, our architecture reduces server load by more than three-quarters.It is now more crucial than ever to create effective and scalable systems, especially for providing video content to end consumers, due to the skyrocketing volume of Internet traffic. Recently, it has become commonplace to transmit video information through HTTP. The advantages of HTTP over older streaming protocols include its compatibility with NAT/Firewalls and its capacity to scale.

Networks for delivering content (CDN). Furthermore, HTML5 players don't require any additional plug-ins or software to play the video material that is streamed over HTTP. Better RTC on the web is made possible by a group of protocols and APIs together referred to as WebRTC. Using only JavaScript code that is executing on web browsers, WebRTC's Data Channel API enables peer-to-peer transport of arbitrary data across web browsers. A signalling server is required for the peer-to-peer connection setup in order to exchange data such IP addresses and port numbers. WebRTC employs complex methods to get around any NAT or firewalls in the connection path during setup. Once the setup process is complete, all data is sent immediately across peers.

Foundation

With WebRTC's Data Channel API, peer help may now be introduced to HTTP-based streaming systems. Users of such a system can connect to one another via peer-to-peer data channels and utilise them to share downloaded video snippets while they are both watching a stream. As a result, it will reduce both the stress on the streaming server and the time needed to acquire a segment. The best feature of these kinds of systems is also how easily they can be deployed. Participants in the streaming service can access peer support without downloading any additional plug-ins or software.

There hasn't been much focus in the literature on the design and study of these kinds of systems. In this study, we therefore created a WebRTC-based peer-assisted system for HTTP live broadcasting. To provide peer assistance, a P2P controller has also been included into the clients. We have carried out some exploratory testing with various peer densities to evaluate the peer contribution during the streaming. Our testing demonstrates that our framework can cut the amount of segment requests the streaming server receives from clients by more than a third without degrading the stream quality.

Implementation

We briefly go over the WebRTC-based peer aided framework's implementation as pacts and the specifics of the P2P controller. A visual representation of our WebRTC-based peer aided framework with two peers may be found in Fig. 1. As an adaptive bitrate streaming technology, we have picked HLS because of how straightforward its specification and format are for live videos. The streaming server keeps a sliding window with a set number of video streams.segments. New video clips are released on a regular basis at a set interval (around 2-4 secs).The server-maintained window advances by one segment duration whenever a new video segment is published. A metadata file in the M3U8 format contains information about thestatus of the sliding window and the segment URLs that are present in the window. Additionally, there is a master playlist file with information on various bitrate streams and URLs for their respective descriptors..

Experimental Result

The tests were carried out on our campus LAN during various times of the day with regular background traffic. The trials are conducted with different peer densities, namely 5, 15, and 30. The average outcomes from numerous tests conducted under each peer density scenario are tabulated in Table I. In all three cases (i.e., with 5, 15 and 30 peers), we can see that the peers draw more than three-quarters of their segments from other peers rather from the server. resulting in a significant reduction in server burden. Our findings also show that as the number of peers in the system increases, a greater proportion of segments are being fetched from nearby peers. These promising early results would serve as a catalyst for our future efforts.

2.11 IEEE Paper (The Peer-to-Peer Adaptive Video Streaming System)

The study describes a peer-to-peer video streaming system that can adapt the encoded video resolution to changes in data flow. The investigation made advantage of low latency live video streaming based on the WebRTC specification. The P2P video streaming system was able to foresee how the receiver's data connection will perform in the future and respond by asking the sender to enhance or reduce the video resolution. Through the use of a network emulator and the activation of a 10 percent packet loss emulation, an experimental examination was carried out. When the packet loss rate rose, the technique suggested in this research was able to reduce the video resolution. Additionally, the system was able to recognise an increase in data.

Video-on-demand (VOD) is how adaptive video streaming systems are commonly run. Each video record contains multiple copies of the video encoded in different resolutions (e.g., 720p, 480p, 320p). The web browser's video player analyses the video playback buffer and switches between available video streams to prevent playback buffer overflow. The playback buffer is examined to find rising media stream data packet lag and packet loss. The video stream bitrate is decreased by changing to a lower resolution, and the data connection's throughput is altered as a result.

Foundation

For applications that transmit live peer-to-peer video, streaming strategy is unacceptable. In light of this, using WebRTC-based low latency video streaming solutions Specifications are preferable. It is challenging to stream live video from a mobile device in an environment with dynamically fluctuating data connection throughput because it is not viableto build numerous concurrent video streams on the device before broadcasting. Despite the fact that the technology might be able to encode four or five video streams concurrently in parallel, the energy consumption of the mobile device will be too high for the majority of practicThe study describes a peer-to-peer video streaming system that can adapt the encoded videoresolution to changes in data flow. The system that is suggested in this study analyses the WebRTC data connection statistics and responds by asking the video source to change theresolution of the video. To examine how the system would behave when the packet loss rate abruptly increased and then returned to normal after some time, an experimental inquiry was carried out. When the data connection's throughput improves, the suggested adaptive video streaming system can momentarily reduce the resolution of the streamed video and thenenhance it again.

Implementation

This study suggests a video streaming system with adaptive video resolution selection that is based on the WebRTC specification. The system consists of a mobile device that acts as a video streaming source (publisher), a signalling server for establishing peer-to-peer connections, and an HTML5-based video player (subscriber). An estimation of the data throughput possible on the current data connection can be obtained by studying the statistics provided for the WebRTC stream.

RTCP packets are sent via UDP in addition to RTP data packets during WebRTC video streaming to send extra information about media streaming-related events (e.g., times-tamp, accumulated number of packets lost from the beginning of the video stream). The proposed system makes use of a mobile video streaming application that has been upgraded to facilitate switching between three different video resolutions, including 1280 720, 720 480, and 320 240. The switching is accomplished without halting the video stream in real time by utilising the FFmpeg library's capabilities. The command to change the video resolution is sent from the web browser (which houses the video player) to the mobile application using a WebSocket-based data connection. Experimental connected d

WAN emulator was used in the experimental inquiry to raise the packet loss rate to 10%. Three video streams with resolutions of 320x240, 640x480, and 1280x720 were subjected to non-periodic packet loss at the same time. Figure 1 compares the distribution of dropped packets across three video streams with varying resolutions. The peak of the packet loss for the high-resolution video (640 480 and 1280 720) streams is seen at the start and monotonically lowers. A 320240 video stream's lost packet count doesn't show a distinct peak at the start of the distortion application.

CHAPTER 3

SYSTEM REQUIREMENT SPECIFICATION

3.1 Hardware Requirement

- Processor Intel Core Duo and above
- RAM minimum size 512MB
- Internet supported device
- Operating System: Microsoft windows 7/8/10
- Minimum of 50MB Hard drive space

3.2 Software Requirement

- Microsoft Windows or Linux
- Internet Connection
- Web Browser
- Allow using your camera and microphone

3.3 Functional Requirements

- Create Event: The organizer can create the event and upload the information as a Post with all the necessary details
- Share webinar details.
- The event organizer can Conduct:
    - Video on/off: organizer can on and off the video
    - Audio on/off: organizer can on and off the audio
    - Screen share: organizer can share their screen with the participants
    - Record session: organizer can record the full session
    - Chat: View & send message
    - Leave meeting

    - Multi-person video call

- Store the recorded Webinars
- Capture participant attendance

3.4 Non-Functional Requirements

- Users can register and login
- **Usability**- They are written expectations and requirements created to guarantee a product is simple to use. This app offers alternatives that are enjoyable to use and approachable.
- **Scalability** – Algorithms/Data Handling should be done in such a way that it should handle a large number of requests without the app crashing or slowing down.
- **Maintainability**- One type of change that frequently occurs after software development is finished is maintenance. The needs alter as time does.
- **Availability -** The application will run 24 x 7 if an internet connection is available.
- **Security** –The application is encrypted and only authorized personnel can join and attend the sessions.
- **Reliability** – WebApp should be adaptable and readable in different screen sizes

Chapter 4

System Design

4.1 Basic Block Diagram

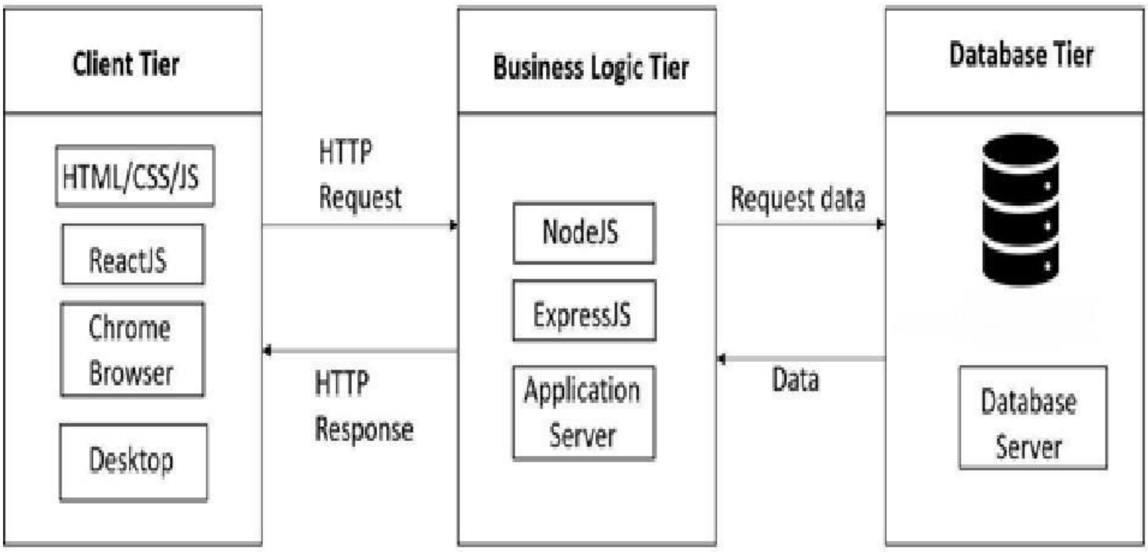

Fig. 4.1 Basic Block Diagram of eKaksha Application

Figure 4.1 shows the basic block diagram structure of eKaksha application in the figure above the client interacts with the application through presentation tier that is the GUI of our web application. The business tier handles the logic of the application. Through the presentation layer user makes the request to the business tier and through business tier data is fetched from the database tier. Our presentation tier is developed with html, CSS, JavaScript and bootstrap. The business tier is developed using node.js and also uses JavaScript code for the backend.

4.2 Protocol Architecture

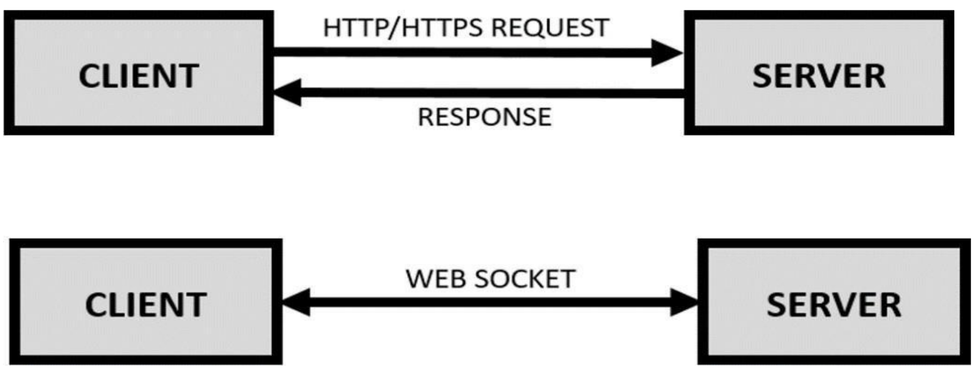

Fig. 4.2 Protocol Architecture of eKaksha Application

As observed in figure 4.2 client makes a http/https request to server. In response to the request made by client server responds with appropriate response and data/packets are served to the client. Client and server are connected to each-other through web socket. The data channel is secured and encrypted with built-in encryption system of the web socket.

4.3 Sequence Diagram

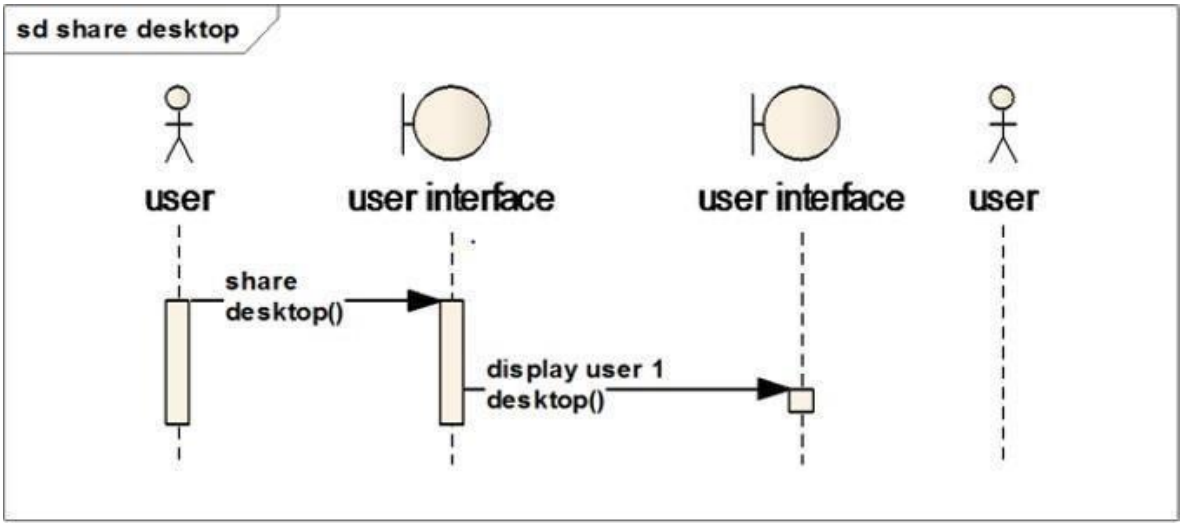

Fig. 4.3 A. Sequence diagram for sharing desktop of eKaksha Application

Figure 4.3 A shows the sequence diagram for sharing the desktop to other users while the meeting is running. When a user shares his desktop, all the others users can see the screen presentation of the user who is sharing the screen. The user shares the screen through sharedesktop() to other users in the meeting. Meanwhile, the other users can see the screen presented by displaydesktop(). The screen can be directly seen in the GUI of the application.

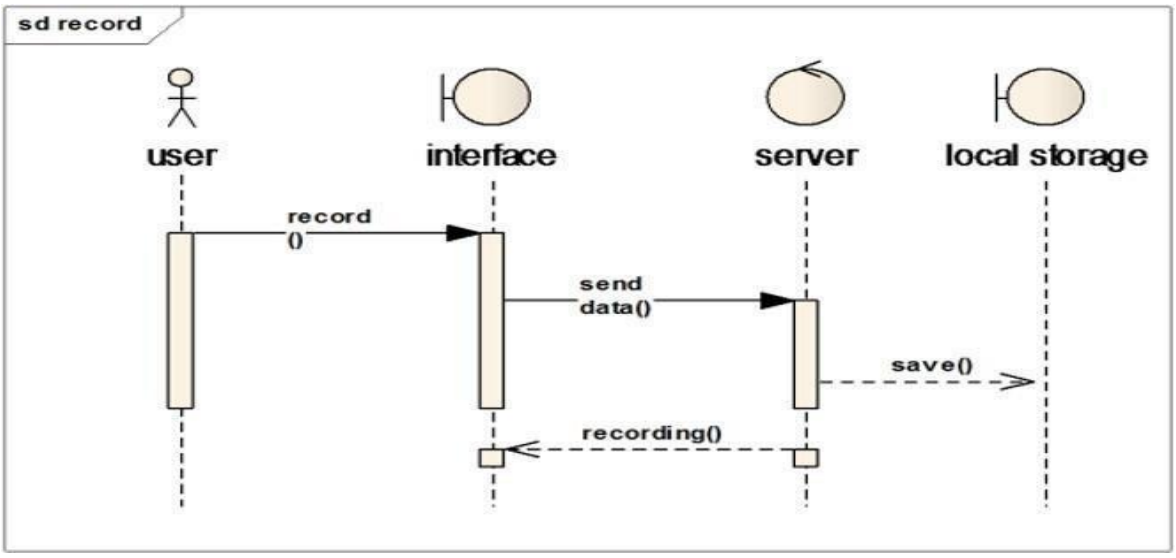

Fig. 4.3 B. Sequence diagram for Screen Record of eKaksha Application

Figure 4.3 B shows the sequence diagram for screen recording which can be done when the meeting is running and can be stored locally. Whenever a user wants to record a screen, the interface of all the things being displayed on his screen are recorded through recording(). And then the data is sent to the server by senddata() and thus the video clip of the whole recording can be accessed by the user and there is also facility to download the screen recording clip.

4.4 Flow Chart

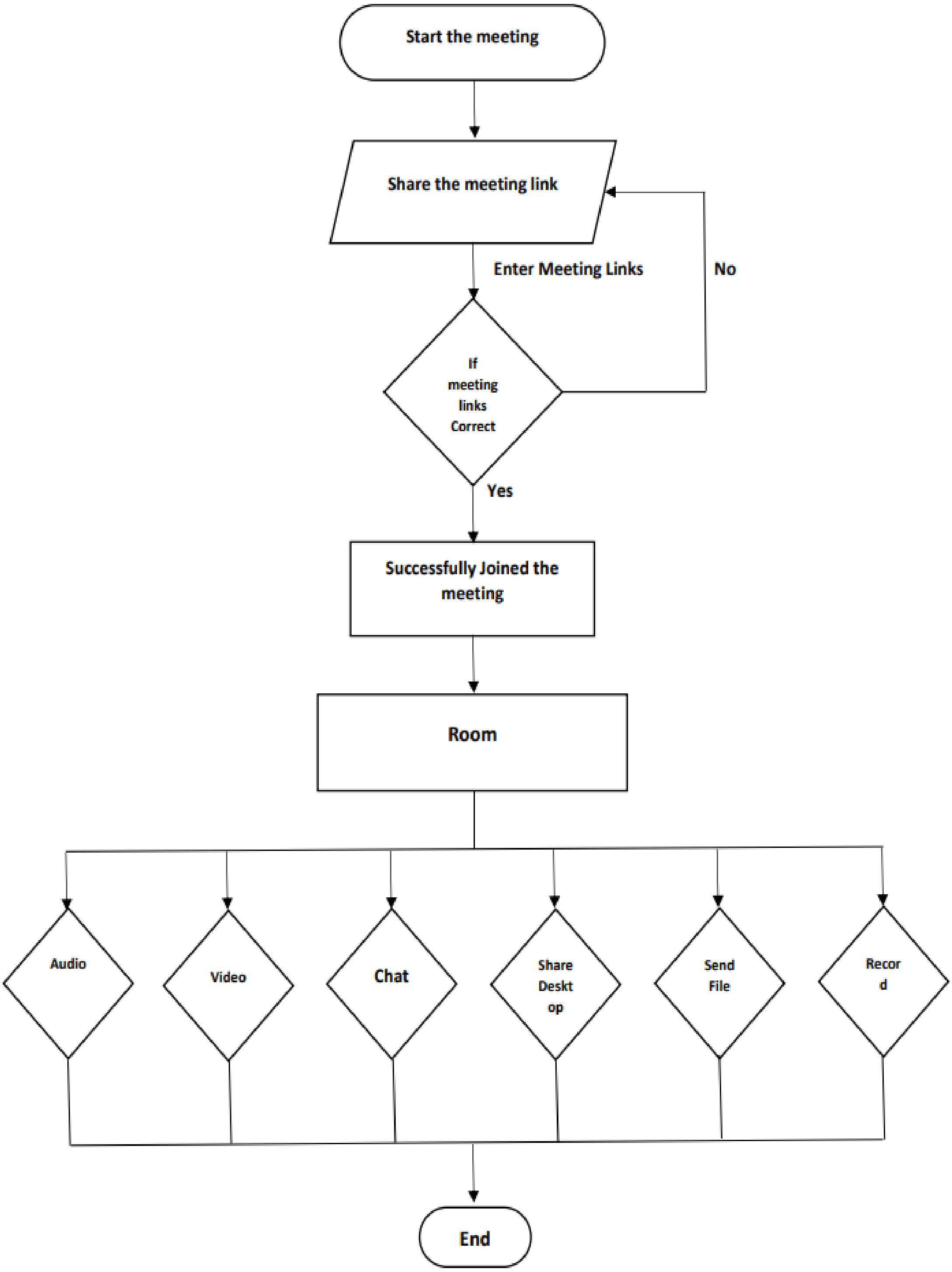

Fig. 4.4 Flow Chart of eKaksha Application

Figure 4.4 shows the basic block diagram of eKaksha application. Firstly, the host has to start the meeting. Then, the meeting link is shared with the participants of that particular meeting. When participants enter meeting link, it is checked if the meeting link is correct or not. If correct, participant can directly join meeting, if not they have to again enter correct meeting ID. When participants have joined, they can access the meeting room, which allows them for audio transmission, video transmission, chat box, share desktop, send files and also record the meeting. When willing to end the meeting, the process ends.

CHAPTER 5

TESTING

5.1 Unit Testing:

Unit Testing may be a level of software testing where individual function/method of software is tested. the aim is to validate that every function/module of the software performs as designed. Here, Unit testing conducted for every module of application.

5.1.1 Unit Testing Test Case for Landing Page Module

Table 5.1.1: Unit testing for Landing Page

Test Case	Objective	Actual Output	Expected Output	Remarks
1	Checking whether the contents are displaying properly.	Yes, displayed	Yes, displayed	Passed
2	Checking whether the application name and logo are displaying properly	Yes, displayed	Yes, displayed	Passed
3	Checking whether the navigation bar is displaying and working properly	Working Properly	Working Properly	Passed
4	Checking whether the Home page in the navigation working properly	Working Properly	Working Properly	Passed
5	Checking whether the buttons in the homepage are working properly	Working Properly	Working Properly	Passed

5.1.2 Unit Testing Test Case for Home Page Module

Table 5.1.2: Unit Testing for Home Page

Test Case	Objective	Actual Output	Expected Output	Remarks
1	Is the Logo and Title are displaying in home page	Yes, displayed	Yes, displayed	Passed
2	Is the Default image responsive on all the Screen sizes	Yes, displayed	Yes, displayed	Passed
3	Is uploaded images are displaying on the Screen	Working Properly	Working Properly	Passed
4	Checking whether all the buttons are working as expected	Working Properly	Working Properly	Passed
5	Is background animation working properly	Yes, displayed	Yes, displayed	Passed
6	Checking whether buttons on the Home page are working as expected	Working Properly	Working Properly	Passed

5.2 Integration Testing

Integration testing could be a level of software testing where individual units are combined and testedasa group. the aim of this level of testing is to reveal faults within the interaction between integrated units.

Table 5.2: Integration testing

Test Case	Objective	Actual Output	Expected Output	Remarks
1	Check whether all the images and text content is displayed properly	Yes, displayed	Yes, displayed	Passed
2	Check whether the data from the backend is loaded and displayed correctly.	Yes, displayed	Yes, displayed	Passed
3	Check whether images and logos are displayed properly for all the locations in the app.	Working Properly	Working Properly	Passed

5.3 System Testing

System testing could be a level of testing that validates the entire and fully integrated software package. the aim of a system test is to gauge the end-to-end system specifications.

Table 5.3: System testing

Test Case	Objective	Actual Output	Expected Output	Remarks
1	Logo is displayed when app is opened	Yes, displayed	Yes, displayed	Passed
2	Start Meeting event button takes to meeting page.	Yes, displayed	Yes, displayed	Passed

3	Only the host of the meeting initially has the meeting details	Working Properly	Working Properly	Passed
4	Is send files button working properly and printing the file name	Yes, displayed	Yes, displayed	Passed
5	video on/off button working properly	Working Properly	Working Properly	Passed
6	Audio on/off button working properly	Working Properly	Working Properly	Passed
7	Hand raise button is working perfectly	Yes, displayed	Yes, displayed	Passed

5.4 Functional Testing

The functional testing of android application normally consists within the areas of testing user interactions moreover as testing the transactions. the varied factors which are relevant in functional testing are

- Type of application based upon the business functionality usages.
- Target audience type.
- Marketing which is employed to spread the appliance.

Table 5.4: Functional Testing

Test Case	Objective	Actual Output	Expected Output	Remarks
1	Validating whether all the required mandatory fields are working as required.	Yes, displayed	Yes, displayed	Passed
2	Validating that the mandatory fields are displayed in the screen in a distinctive way than the non- mandatory fields	Yes, displayed	Yes, displayed	Passed
3	Validating whether the application works as per as requirement whenever the application starts/stops.	Working Properly	Working Properly	Passed

4	Validating that the device is able to perform required multitasking requirements whenever it is necessary to do so	Working Properly	Working Properly	Passed
5	Validating that the page scrolling scenarios are being enabled in the application as necessary.	Working Properly	Working Properly	Passed
6	Validating that navigation between modules in app are as per requirement	Working Properly	Working Properly	Passed
7	Validating whether the user interface is direct and understandable by the the end user	Working Properly	Working Properly	Passed
8	Validating the buttons and options are working as per the requirement	Working Properly	Working Properly	Passed

5.5 Usability Testing

The usability testing process of the Mobile application is performed to possess a fast and straightforward step application with less functionality than a slow and difficult application with many features. the most objective is to make sure that application has an easy-to-use, intuitive and the same as industry-accepted interfaces which are widely used.

Table 5.5: Usability Testing

Test Case	Objective	Actual Output	Expected Output	Remarks
1	Ensuring that the buttons have the required size.	Yes, displayed	Yes, displayed	Passed
2	Ensuring that the buttons are placed in the same section of the screen to avoid confusion to the end users	Yes, displayed	Yes, displayed	Passed
3	Ensuring that the icons are natural and consistent with the application.	Working Properly	Working Properly	Passed
4	Ensuring that the buttons, which have the same function, and also have the same color.	Working Properly	Working Properly	Passed

5	Ensuring that the text is kept simple and clear to be visible to the end users	Working Properly	Working Properly	Passed
6	Ensuring that the short sentences and paragraphs are readable to the end users.	Working Properly	Working Properly	Passed
7	Ensuring that the font size is big enoughto be readable and not too big or too small.	Working Properly	Working Properly	Passed
8	Ensuring that the application items are always synchronized according to the user actions.	Working Properly	Working Properly	Passed

5.6 Compatibility Testing

Compatibility testing on mobile devices is performed to make sure that since mobile devices have different size, resolution, screen, version and hardware therefore the application shouldbe tested a cross all the devices to make sure that the applying works as desired.

Table 6.6: Compatibility Testing

Test Case	Objective	Actual Output	Expected Output	Remarks
1	Ensuring that the text is readable forall users for the application.	Yes, displayed	Yes, displayed	Passed
2	Ensuring that the screen sharing is working properly for different screen sizes	Working Properly	Working Properly	Passed
3	Ensuring the application is able to receive notifications from the cloud.	Working Properly	Working Properly	Passed
4	Validating all the buttons are visible and are working as per therequirement.	Working Properly	Working Properly	Passed

CHAPTER 6

CONCLUSION

Adding a video chat feature to our application provides a powerful visual means for individuals to connect. No wonder then this feature is becoming increasingly popular in the world of business, to improve efficiency, enhance communications, and produce a satisfying user experience. Live video chat can be integrated into an existing communication platform, or built as a separate standalone application specific to your business needs.

The system is built on the MERN stack, which makes use of Bootstrap, a CSS library, for the front-end, NodeJs and expressJs for the server and backend processing, and MongoDB for our continuously growing and incredibly scalable database. This platform is the best for both the end-user, who is looking for events and friends that match their interests, and the event organisers as well because of how effectively and powerfully it provides the community users based on their interests and by using the most cutting-edge technologies

Future Enhancement

As e-Kaksha is video calling application lashed with all the appropriate requirements to handle user's requirements. Still there is lot of areas which can be further improved and developed. We can add background filter option, adding emojis in the chat. Implement a function to send chats to an individual member in the meeting. Introduce an admin type role I which host can kick any participants out of the meeting or control the participant's media devices.

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O. Boyaci, A. G. Forte and H. Schulzrinne, "Performance of Video-Chat Applications under Congestion," 2009 11th IEEE International Symposium on Multimedia, 2009, pp. 213-218, doi: 10.1109/ISM.2009.45.

4. D. Tang and L. Zhang, "Audio and Video Mixing Method to Enhance WebRTC," in IEEE Access, vol. 8, pp. 67228-67241, 2020, doi: 10.1109/ACCESS.2020.2985412.

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6. https://socket.io/

7. https://nodejs.org/en/

APPENDIX A

SNAPSHOTS

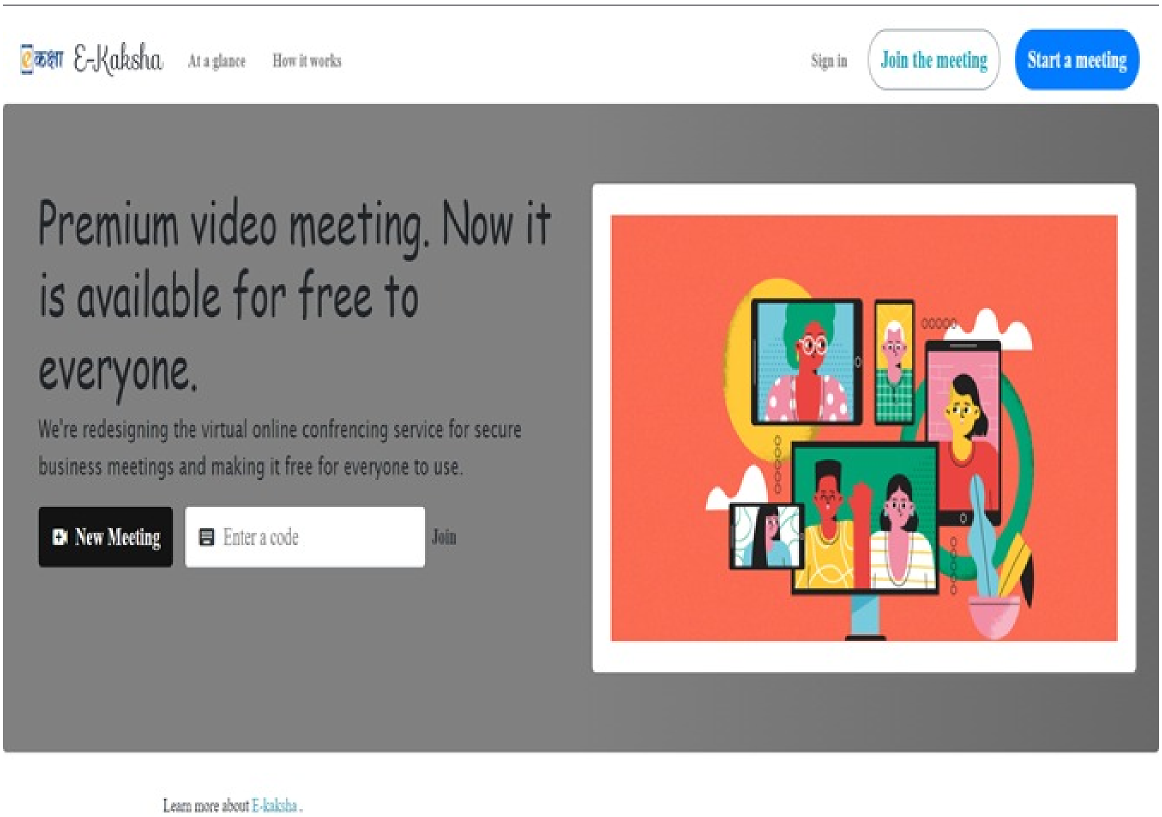

Figure A.1: HomePage

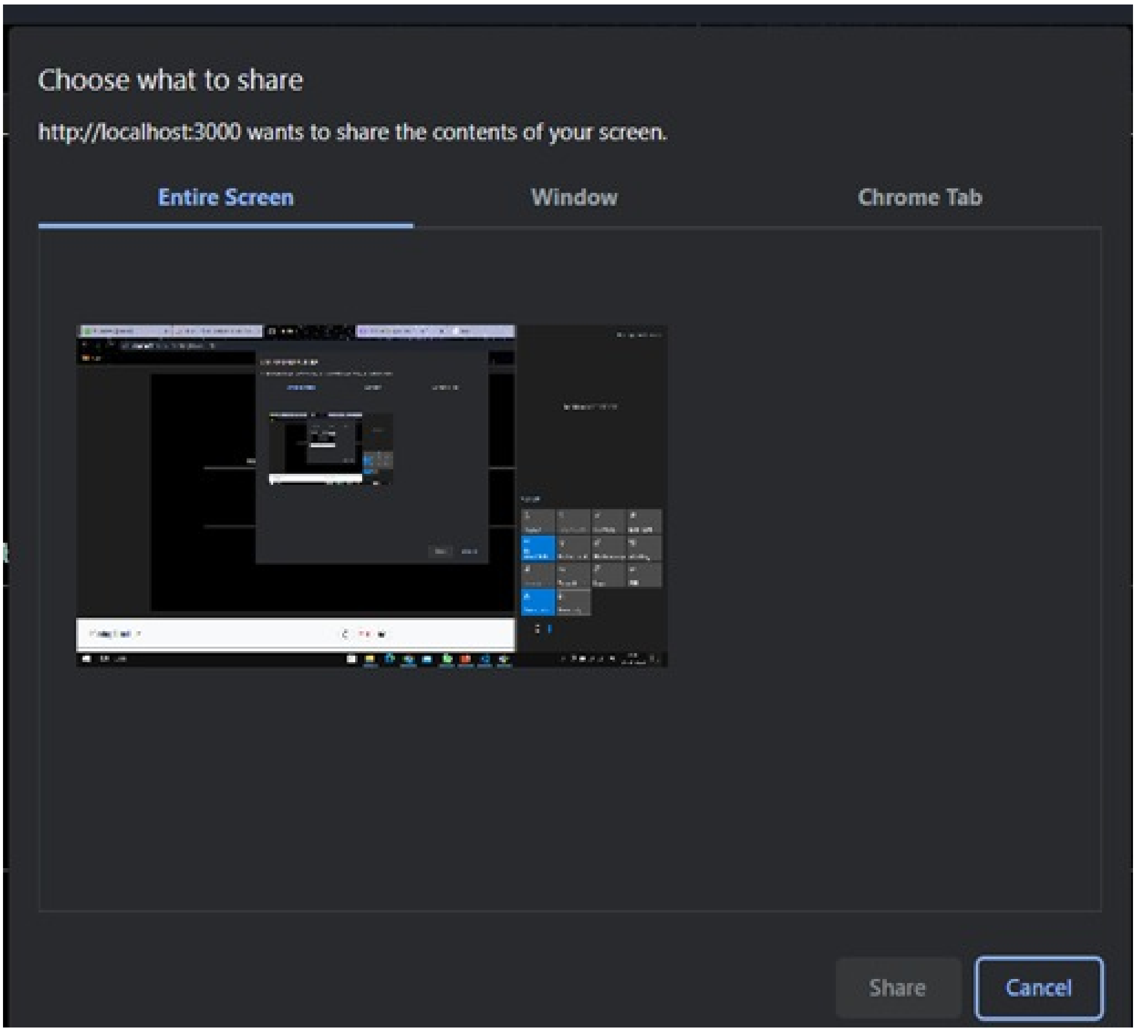

Figure A.2: Presenting Screen

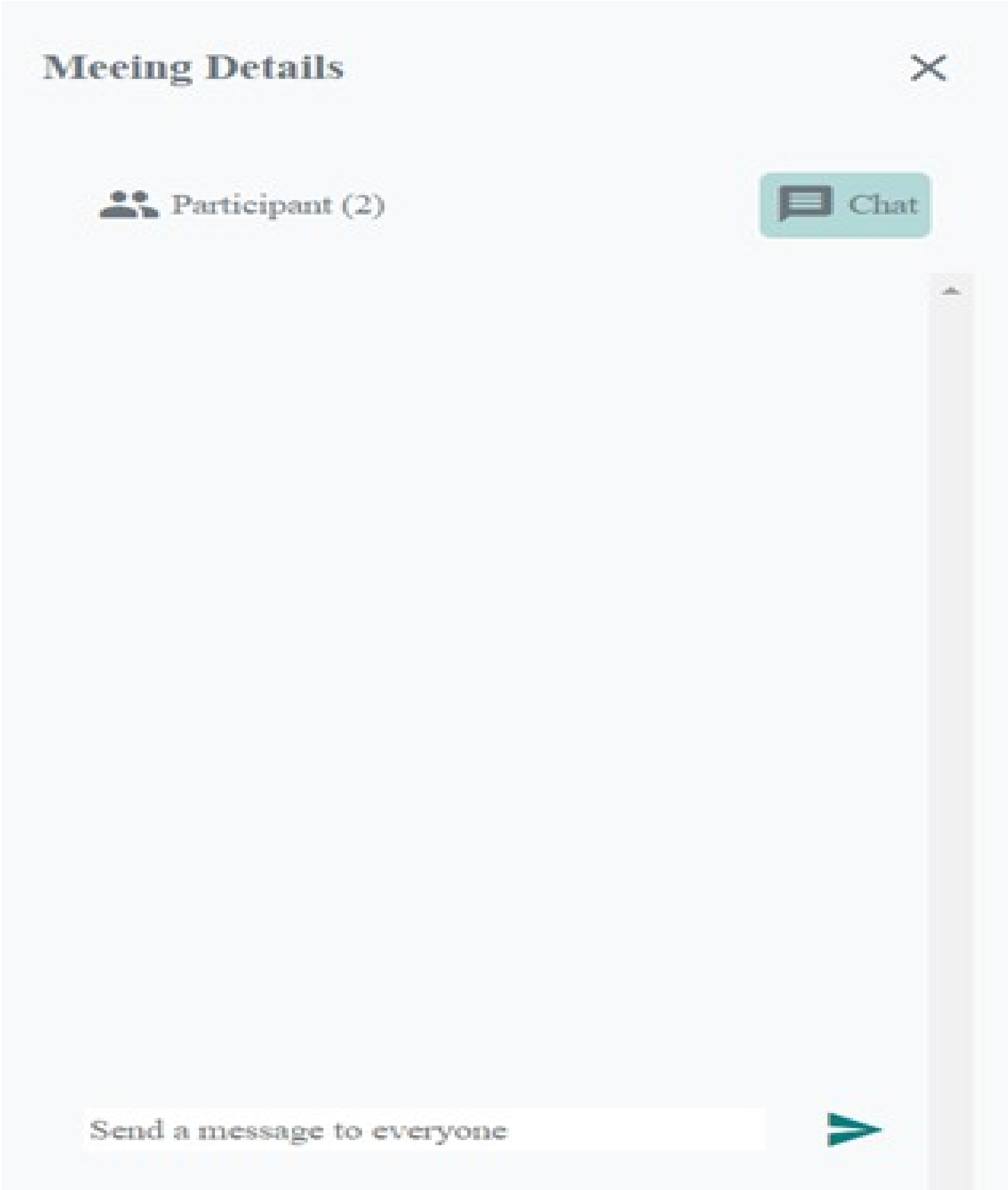

Figure A.3: ChatBox

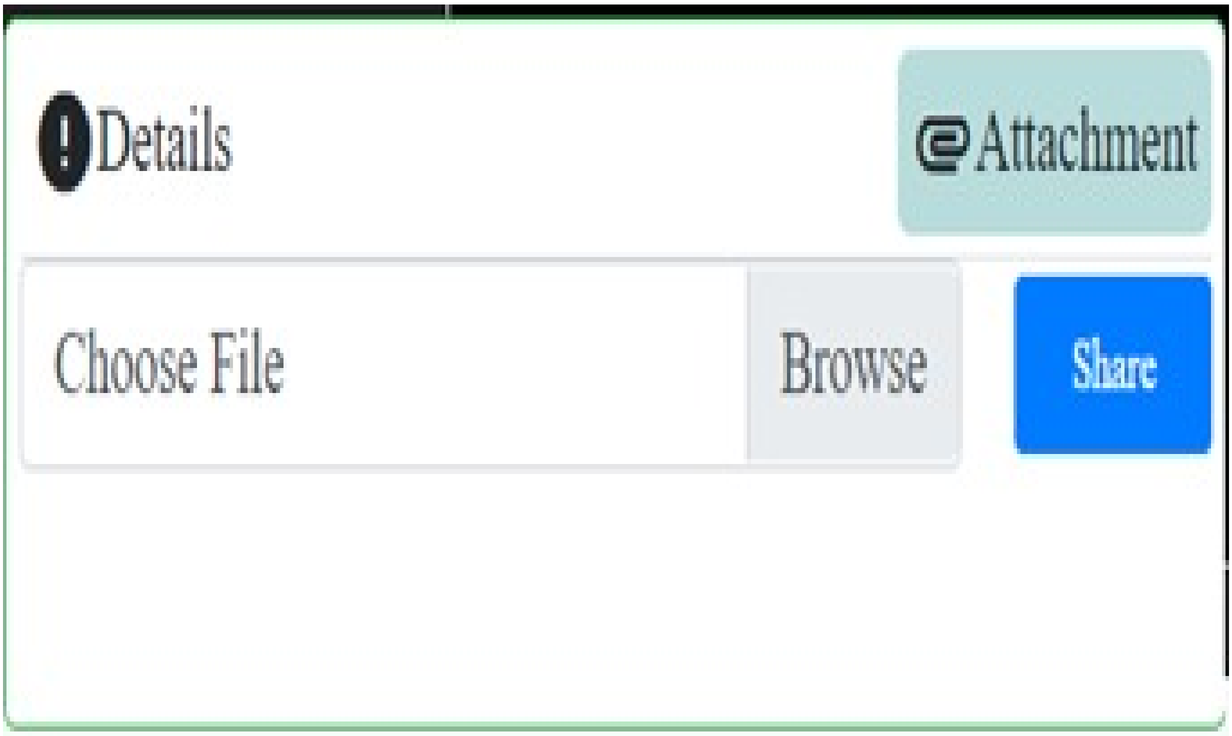

Figure A.4: File Attachment

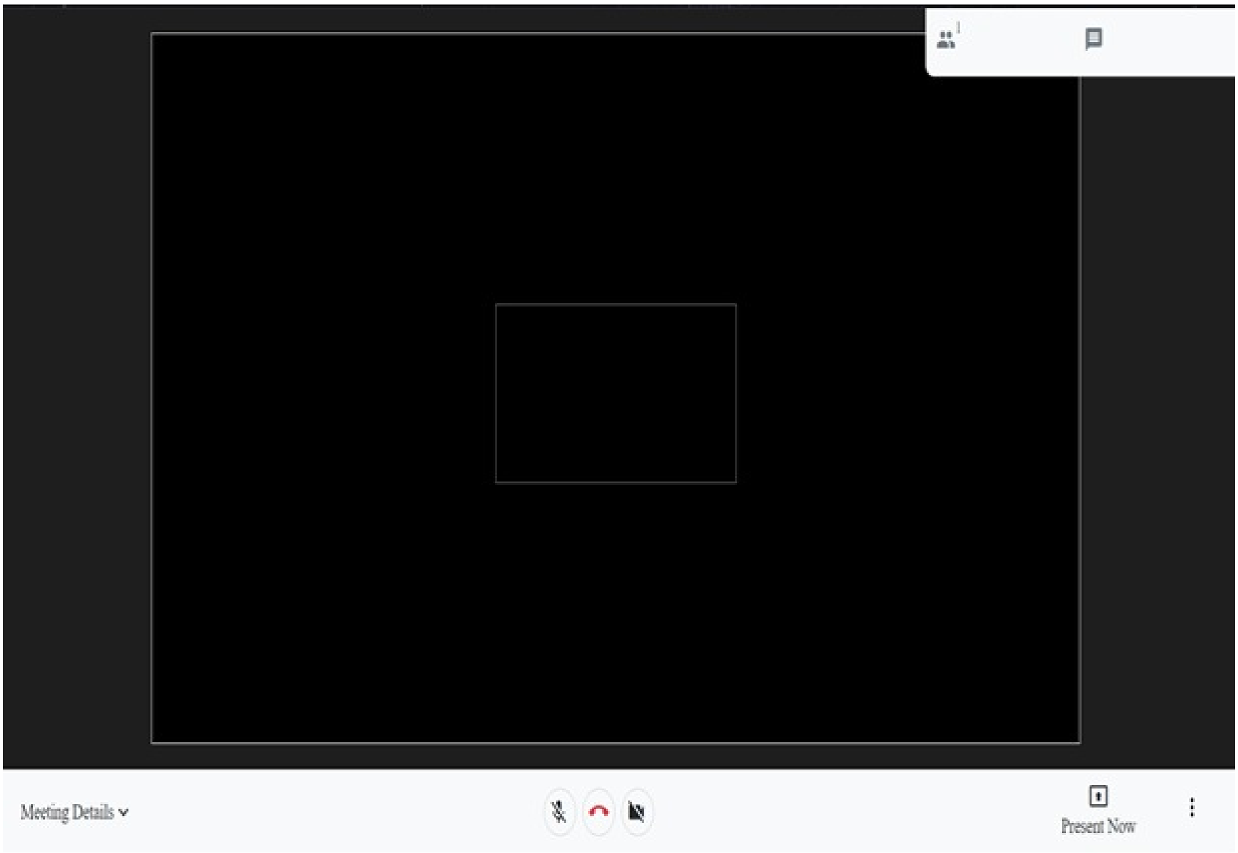

Figure A.5: Meeting View

Project Report

ORIGINALITY REPORT

20%	12%	8%	9%
SIMILARITY INDEX	INTERNET SOURCES	PUBLICATIONS	STUDENT PAPERS

PRIMARY SOURCES

1	nectar.northampton.ac.uk Internet Source	3%
2	Paulius Tumas, Arturas Serackis. "Peer-to-peer adaptive video streaming system", 2015 IEEE 3rd Workshop on Advances in Information, Electronic and Electrical Engineering (AIEEE), 2015 Publication	2%
3	Midhul Varma, Hema Kumar Yarnagula, Venkatesh Tamarapalli. "WebRTC-based peer assisted framework for HTTP live streaming", 2017 9th International Conference on Communication Systems and Networks (COMSNETS), 2017 Publication	2%
4	Submitted to Visvesvaraya Technological University Student Paper	2%
5	www.makeuseof.com Internet Source	1%
	www.guru99.com	

6	Internet Source	1%
7	Submitted to Mancosa Student Paper	1%
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9	Submitted to UOW Malaysia KDU University College Sdn. Bhd Student Paper	1%
10	journals.grdpublications.com Internet Source	1%
11	www.compete366.com Internet Source	1%
12	www.softermii.com Internet Source	1%
13	Submitted to iGlobal University Student Paper	<1%
14	2021.co.id Internet Source	<1%
15	Submitted to CTI Education Group Student Paper	<1%
16	student.aut.ac.nz Internet Source	<1%
17	Submitted to Lithan Academy Pte Ltd Student Paper	

		<1%
18	techboomers.com Internet Source	<1%
19	Submitted to University of KwaZulu-Natal Student Paper	<1%
20	Submitted to MAHSA University Student Paper	<1%
21	scholarworks.calstate.edu Internet Source	<1%
22	www.etgroup.ca Internet Source	<1%
23	unrecoding.weebly.com Internet Source	<1%
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Acceptance Letter

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Dear Authors,

We are glad to inform you that your paper has been accepted as per our fast peer review process:

Authors Name: Shubham Kumar, Mohammed Asad Qatib, Chirag Patidar, Saikat Maity, Sreelatha P.K.

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e-Kaksha WebRTC based Video Conferencing App

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***Abstract* – eKaksha, a video conferencing app based on WebRTC is a multi-utility application that provides wide range of features. Recall the recent few past years and we can confidently say that it's pretty impossible to imagine our lives without video conferencing apps due to Covid-19. From a simple video call involving two close friends, to a meeting involving all the Chief Ministers of our states in India-all these meetings were based on Video Conferencing apps.**

Now, imagine video conferencing applications which all require logins to their own platforms, applications that are heavy, require high-end processors for the meetings, applications that demand expertise and are complex to use, this can be really frustrating for the users. Add privacy/security breaches to the list and users would rather prefer not using the applications.
e-Kaksha pretty much overcomes all these flaws/problems. WebRTC that stands for Web Real-Time Communication is a open source communication technology which is capable of overcoming issues and implement an application that is light-weight and has a very low latency. WebRTC is based on UDP and not on TCP, which means WebRTC does not require handshakes from the server and is also bi-directional in nature.

Keywords- Video Conferencing App, WebRTC.

I. INTRODUCTION

WebRTC, in a decade of its existence has pretty much changed the way video conferencing applications work and it has also increased the reliability over it. The fact that WebRTC is said to be P2P (Peer-to-Peer) service implies that there is no requirement for the presence of a server in the middle. But initially WebRTC does require a server to set up the connection between the peers involved in the communication. P2P connection between the peers present In the communication require a server initially, which otherwise is said to be a signaling server. This signaling server acts as a middle man that sets up connection and allows the service to be Peer-to-Peer after the first connection establishment between peers.

In WebRTC, initially one client sends the offer, also called as SDP (Session Description Protocol) to the signaling server. SDP is the media configuration of the client, which tells if the client wants to send video or it wants to send audio or it can also send both video and the audio. The signaling server when it receives the SDP from the client, it passes the same SDP to the other client who is to be involved in the communication. In response, the other client sends back the answer SDP to the signaling server and the signaling server send the answer SDP to the initial client.

Now comes the main part, where both the clients need to share their network configurations in order to establish the connection. Then client receives ICE Candidates (Stun and Turn) from the WebRTC API. The initial client sends these configuration over the signaling server to other client and vice-versa. Now that both clients have network information about each other, the connection is successfully established.

The connection is established and both video and audio can be seamlessly shared among the clients. Additionally, this eKaksha application supports file sharing where the participants can share files during the meeting. It also supports screen recording so that it can be used for future references. Whenever the host is willing to conduct a meeting or conference, he has to create and share the meeting ID with the participants. When the participants enter the correct ID, the connection is established and the meetings can take place. Using WebRTC, a system with low-configurations can also be involved in the meeting and no doubt, the privacy is intact and not compromised.

II. LITERATURE SURVEY

(a) P2P video conferencing system based on WebRTC

Pocket-friendly large-scale close to real-time video streaming technology has remained out of reach despite innovations and huge efforts over the past two decades. Introduction of Internet Protocol (IP) Unicast and Must architecture were earlier attempts to tackle this problem [1]. But few problems regarding scalability and difficult deployment have hampered these attempts. In recent years peer-to-peer (P2P) based broadcast is upcoming as a promising technique because this paradigm brings many unique advantages such as easy deployment, cost effectiveness, scalability and resilience. While P2P applications such as voice-over-IP and file download have gained a lot of popularity, video conferencing is still in its early stages [2]. The aim is to introduce P2P Video Conferencing System based on WebRTC (Web Real Time Communication). WebRTC provides the opportunity of establishing successful Peer to Peer connections within web, without third party software and additional plugins. The paper confers a scalable live video conferencing architecture designed based on Web real-time communication. These experiments show that Web Real Time Communication is capable of being a building block for expandable, robust live video-conferencing within a web browser.

(b) Design and Evaluation of Browser-to-Browser Video Conferencing in WebRTC

This design and evaluation of WebRTC narrate the Web Real-Time Communication (WebRTC) technology and the application of the server and clients. The main aim of this study is to design and imply Web Real Time Communication video streaming between the browsers in real time implementation using browsers (Chrome). Also, an evaluation of Quality of Experience (QoE) and CPU performance was achieved. Further, a signalling channel between the browsers using the Web-Socket protocol via Node.js platform has been setup. This study will give web developers an opportunity to snatch the WebRTC technology, and also to understand how to design a dependable WebRTC video conferencing

III. EXISTING SYSTEMS

(a) Zoom: Is one of the most famous video conferencing application. It is always recommended by the professionals all around the globe. Zoom for personal use comes with zero cost while for business, it offers a lot of features at the minimum cost and it has user friendly interface.

Features: ●Super easy to use. ● Supported on multiple devices. ●Budget friendly. ●Centralized platform.

Disadvantages: ●Delayed customer service. ●Vulnerable to cyber-attacks. ●Reports of data sharing with China. ●Compromised security.

(b) WebEx: Manufactured by Cisco Systems, one of the most used online meeting tools. It works on both windows, android as well as Mac operating system. It allows users to meet over the internet as well to share screens and speaking through phone.

Features: ●Simple user interface. ●Easy to share screens and documents. ●Provides whiteboards and pass keyboard and mouse control. ●Gives the recording option.

Disadvantages: ●Expensive as compared to other platforms. ●Chooses internet explorer as the default browser. ●Sometimes users face audio issues.

(c) Google Meet: It is a video conferencing application developed and maintained by Google Inc. It is so popular that is has already replaced two of the Google's owned applications, namely Google Hangouts and Google Chat. It is also set to replace Google Duo. It is available on web, windows, IOS as well as android.

Features: ●Free for all users. ●Flexible paid versions. ●It is integrated with other google services. ●Simple and straightforward user experience and user interface. ●Good audio and video quality. ●Strong security.

Disadvantages: ●Relatively limited features as compared to others. ●Heavy on hardware and system resources. ●User must create a google account to use the app. ●Screen sharing restrictions.

III.PROPOSED WORK

eKaksha is a web application using which the user can connect to anyone and anyplace, given both have internet access. In the year 2020, the global video calling apps market was whooping $5.77 billion, later in 2021, it reached $6.28 billion. In fact the market promises to reach $12.99 billion by the end of 2028, the numbers are impressive. eKaksha is a video calling application but it offers much more than just video calling. Through eKaksha we can connect through audio/video call, screen sharing, file transfer, record the meeting feature and save it to local storage, chat and raise the hand option is also included in it. It uses WebRTC and Socket.io to communicate in real-time using web browser. WebRTC and Socket.io is great when combined together and the results are impressive.

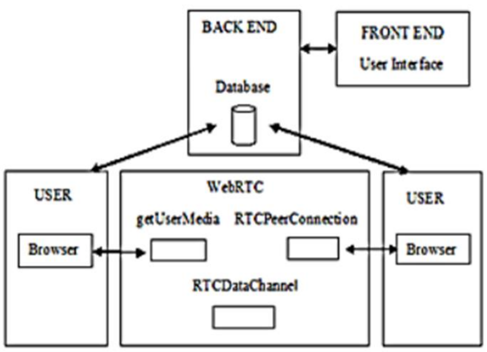

Fig. 1. SYSTEM ARCHITECTURE

The Fig.1 shows the architecture of our application which is based on a MVC model. Our Presentation tier (View) will be developed in JavaScript, HTML and CSS. Users will interact with presentation tier to use our application. The Business Logic Tier (Controller) will be developed using NodeJs and ExpressJS, and this tier represents the Server that will act as a crossover of interaction for the Presentation Tier and Database Tier. This tier will responds with the HTML pages to the user's device and accept HTTP requests from the user and follow with the appropriate response.

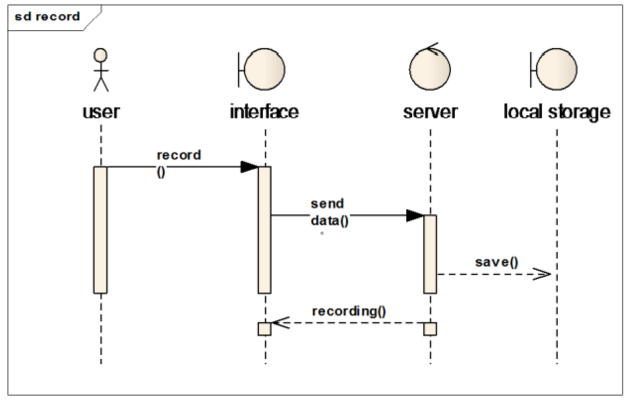

Fig. 2. SEQUENCE DIAGRAM FOR RECORDING

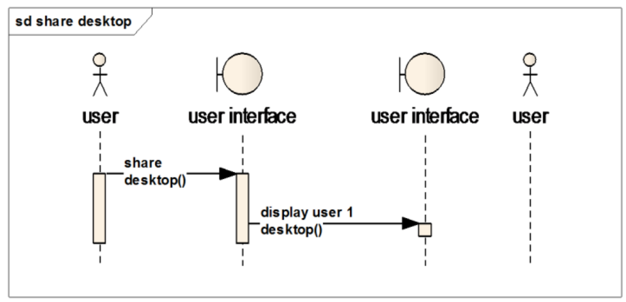

Fig. 3. SEQUENCE DIAGRAM FOR SHARE DESKTOP

The host can instantly start the meeting and share the auto-generated meeting link with other peers. The other peers has only to paste the link in their browser and they will be connected instantly. Any joiner can leave the meeting anytime they want just by a single click. Our application is totally safe to use as we don't hunt the data of the user. There is no need to provide any prior details to use this application. It is safely encrypted because of WebRTC and effective as well.

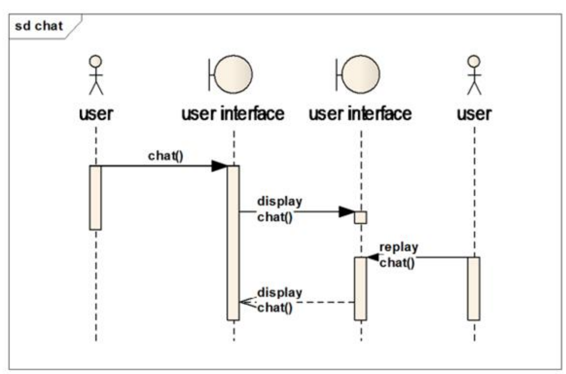

FIG. 4. SEQUENCE DIAGRAM FOR CHAT

IV.CONCLUSION

Adding the video chat feature in our applications gives immense visual means for the users to connect and interact with one another. This feature is becoming popular day by day and is on trend. The application proves to be robust in the world of business. It improves efficiency, enhance communications, and produce a satisfying user experience. Live video chat can be integrated into an existing communication platform, or built as a separate standalone application specific to your business needs.
The technology is based on MERN stack, and bootstrap is the commendable part of it. CSS library is for the front end and NodeJS and ExpressJS are for the server and backend handling. It is a powerful and efficient tool for community and professional users to find their best matched events and friends, and for the event organizers too.

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