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Peer-to-Peer Conferencing using Blockchain, WebRTC and SIP

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Abstract—The owner of the centralized video platform has more control over uploaded content than the content producer does. But the other Blockchain-based decentralized video services are attempting to reduce ad pressure and get rid of middlemen. The article suggests a combination of a safe encryption technique and an access control mechanism created "with technology" to create a successful decentralized video streaming platform built on the Blockchain. Peer-to-peer (P2P) overlays are one of the complicated network applications and services that have been migrated to the Web as a result of the increasing support for Web Real-Time Communication (WebRTC) standard in modern browsers for real-time communications. The expansion of access networks' bandwidth also makes it possible for end users to start their own content businesses. This paper presents a preliminary proposal of metrics and technologies to move toward a decentralized cooperative architecture for large-scale, real-time live stream content de- livery based on WebRTC, without the requirement of a Content Delivery Network (CDN) infrastructure. The paper takes into account the light of the aforementioned aspects [6].

Keywords-WebRTC, Session Initiation Protocol, blockchain, token, conferencing, peer-to-peer.

I. INTRODUCTION

Due to its openness, immutability, and decentralised nature, blockchain has been an emerging technology for more than a decade.

The primary problem with popular video platforms is that one company controls and finances everything. The platform makes the decisions regarding what content should be kept and what should be removed, stifling the originality of the content provider. As a result, a user is unable to select their own material.

Users can interact with shared objects in a shared workspace typically provided by real-time groupware systems. A shared workspace, however, frequently falls short in keeping users informed. By providing a visual context, video conferencing can make it easier for users to communicate and comprehend one another. Additionally, users can work at any time and from any location thanks to contemporary communication methods and flexible working arrangements. A groupware system should therefore

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The Web Real-Time Communication is a new communication technology candidate that promises to overcome many of the limitations mentioned and to bring a genuine and native real- time communication experience to the web. Today, after years of development, it appears that we have a final solution for our needs (WebRTC). Without the need for a specialised client, extra plug-ins, or extensions, WebRTC technology enables native integration of multimedia communication elements into modern web browsers, enabling peer-to-peer conversations using voice, video, text, and data. Several standardisation groups, including the World Wide Web Consortium (W3C) and the Internet Engineering Task Forces, are interested in the design evolution, standardisation, and technology integration of WebRTC (IETF). Given its capabilities, WebRTC is attracting increasing attention. It has a high potential to improve and grow a variety of existing businesses, with education and training appearing to be one use case that makes the most sense. The interaction between students in person during class and online learning can both be greatly enhanced via WebRTC. The continuing WebRTC effort is largely focused on peer-to-peer browser-based multimedia communication in order to create a ubiquitous, open, and browser-based communication frame work without the need for specialised communication end devices. The idea behind WebRTC is simple yet ground-breaking, advancing communication toward a browser-based, clientless future for specialised applications.

A number of currently utilised and implemented communication solutions are pushed into the cate gory of "ancient" technologies by this new method. One such widely used option is the SIP-based enterprise and telecom solutions for IP real-time communication. As a result, interaction and integration between the current SIP communication solutions and the upcoming WebRTC standard are required [1].

This project's goal is to propose a mutual authentication system for secure key distribution through the blockchain network before the multimedia communication of P2P applications. It establishes a peer-to-peer, secure communication environment built on the blockchain with the goal of entirely decentralising the authentication of conferencing applications. Data latency is the amount of time it takes for data to move between a source and an endpoint. Low latency results in higher visual quality and fewer interruptions during video conferences. In an effort to bring the data source as close to the end user as possible, centralised services and solutions try to achieve this by establishing more and more data centers [3].

Decentralized blockchain-based authentication creates a conducive atmosphere for safe audio communication. The suggested approach, which embodies cost effectiveness, ease, and trust, is a new generation of authentication model that improves the online experience of users of conferencing applications.

II. TECHNOLOGY AND PROTOCOLS

A web browser equipped with real-time communication capabilities, or WebRTC, enables the creation of peer-to-peer chat sessions. The primary signalling mechanism, nevertheless, is absent and essential for exchanging media properties and connection data. Which signalling technique a WebRTC application chooses to utilise is up to it. The use of SIP and its protocol family is one of the useful options, reflecting its adoption by telecommunication solutions. WebRTC and SIP integration can take place directly in the browser or with the aid of an intermediary device, such as a translation gateway. A WebRTC application needs a reliable web-based transport method and the ability to interface with SIP entities. The transport method ought to offer a quick and effective conduit for two-way communication. The WebSocket protocol is offered by these characteristics.

For the purpose of identifying and briefly de scribing the essential protocols and technologies needed for the integration of WebRTC and SIP, we provide some theoretical background in this chapter [1].

A. WebRTC

A native web browser-based technology was a key motivation behind the development of WebRTC. The solution should be built on well-established web technologies that enable users to easily insert and use real-time communication media (such as audio, video, and data transmission) into web pages. Real-time communication can then be incorporated into a web page with the use of HTML and JavaScript, two common online technologies. Although other web companies (Mozilla, Opera), as well as manufacturers of telecom equipment (Ericsson, Cisco, Alcatel Lucent), have joined Google in leading the development of WebRTC as an open source project, the endeavour has the support of these companies. Real-Time Communications (RTC) functionality for web browsers is now provided by the free and open-source We bRTC project, eliminating the need for any additional plugins or outside software. The different WebRTC standardisation initiatives are under the control of the IETF and W3C. Standardization distinguishes between internal communication (media and

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signalling exchange) and a single node's duties and obligations (application level duties).

Together, they are concentrating on developing the HTML JavaScript APIs (W3C) and the underlying networking communication protocols for setting up and managing secure peer-to-peer multimedia connections between pairs of browsers (IETF).

With functionality including managing browser-to-browser connections, negotiating, controlling media, encoding/decoding, and more, the WebRTC API makes it possible to create browser- based client-side RTC programs. Three APIs, MediaStream, RTCPeerConnection, and RTCDataChannel, are implemented by WebRTC. Within a larger WebRTC architecture, they are included. Web browser apps can access the media streams coming from the user's camera and microphone thanks to the Media Stream API. RTCPeerConnection enables the setup of an audio or video connection (offer/accept negotiation), with bandwidth management and encryption features. Although a connection control mechanism, also known as signalling, is necessary, the RTCPeerConnection API does not offer one [1].

The functionality of existing defined internet session control protocols like SIP and XMPP (Extensible Messaging and Presence Protocol) may be reused by WebRTC developers. The RTCDataChannel API offers capability for bidirectional data channel peer-to-peer communication exchange. The creation of a variety of secure communication channels is natively supported by RTCDataChannel API. WebRTC utilises a number of communication protocols and communication services, all of which web browsers are required to support from the perspective of network communication. As is customary for contemporary technologies, communication protocols are divided into those for the signalling transfer and those for the media/application data transmission.

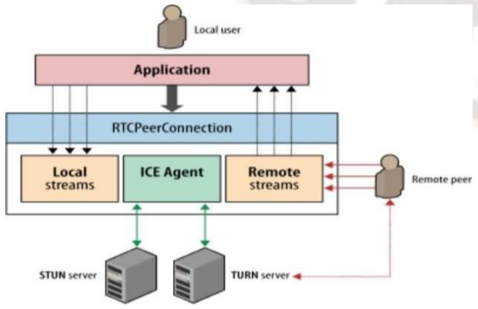

Fig. 1: WebRTC Architecture

Real-time media transport is accomplished via the User Datagram Protocol (UDP). In order to implement WebRTC, an unencrypted UDP data- gram stream must be protected using encrypted UDP communication. WebRTC complies with these criteria by utilising the Datagram Transport Layer Security (DTLS), Secure Real-Time Transport (SRTP), and Stream Control Transport Protocol (SCTP). SRTP is used to protect large media outlets (e.g. audio, video). In order to interact with key management protocols, SRTP needs this feature. The key management protocol enables the safe exchange of encryption keys for media and application data encryption. This is DTLS's responsibility. The protocol SCTP, which builds on UDP and is used to multiplex several application data streams, is the last but not least [1].

WebRTC needed to adopt a session setup or a signalling control mechanism in order to provide a peer-to-peer protected media channel. Media features and connection parameters are negotiated through an offer and response process using the signalling mechanism. The Session Description Protocol, another protocol, describes these parameters (SDP). The WebRTC does not inherently de- fine a new signalling protocol, nor does it specify the usage of any particular, already established signalling protocol (SIP or XMPP). But a signalling server of some sort must be present for WebRTC to function. The signalling server must unavoidably comprehend SDP.

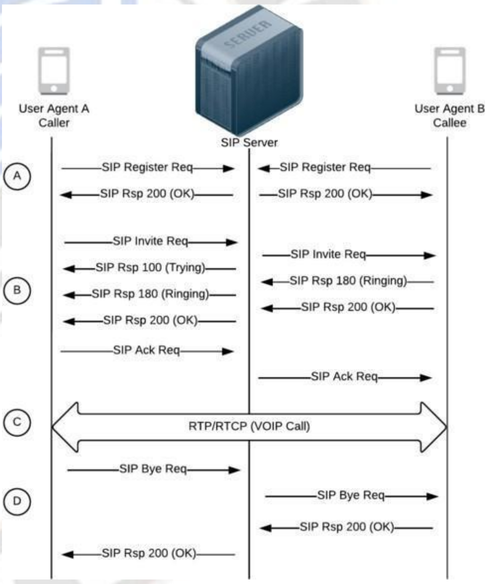

Fig. 2: Session Initiation Protocol.

The handshaking and connection establishing processes for peer-to-peer multimedia capabilities are also aided by the signalling server. JavaScript Session Establishment Protocol-based control method is used by WebRTC as the session setup mechanism (JSEP). An SDP abstraction is offered by JSEP, which is used to streamline the building of WebRTC applications. The DTLS/UDP or TLS/TCP protocol stacks are sup- ported by the WebRTC signalling mechanism for the transfer of encrypted and secure signalling. Any network communication topologies and scenarios must permit the successful formation of peer-to-peer sessions.

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The deployment of public/private network scenarios, established firewalls, and network address translators (NATs) must be reflected in the signalling and soon after media transport. WebRTC must therefore contain NAT and firewall traversal methods. The Interactive Connectivity Establishment (ICE) framework, a VoIP-inspired approach, is used by WebRTC. WebRTC peers can gather and test their preferred "candidates of communication" with the aid of ICE's tools (STUN/TURN). Candidates for communication who have been carefully chosen are increasing chances of success [1].

B. Session Initiation Protocol

SIP is an IP-based application protocol that was created by the IETF as an open signalling system for multimedia communications formed over packet IP networks. For a wide variety of multimedia communications, including voice, data, pictures, messaging, presence, file transfers, etc., SIP offers signalling and control functions. The SIP protocol is regarded as a crucial component in network convergence. SIP specifies a number of logical functional units, each of which has a distinct role and permits the establishment of SIP-based communication infrastructure that offers consumers personal, terminal, and service mobility.

The entities fall into one of two categories: SIP servers or SIP endpoints. User agents (UA), back-to-back users, and SIP endpoints are SIP or agent (B2BUA) gateways. SIP servers might include a Registrar, proxy, or redirect server roles are all used. SIP is the signalling protocol in terms of the protocol. SIP is the preferred protocol from an implementation standpoint. technology for communication that is being used and adopted a larger selection of technologies, protocols, and security mechanisms (Fig. 2) [1].

It primarily covers, among other things: RTP/RTCP for media. SRTP for its media security, transport, and DNS for SIP addressing; media signalling and SDP for features handshaking, protected by employing the Protocol stacks using DTLS/UDP or TLS/TCP; SIMPLE, for presence and instant messaging, use MSRP and XCAP; for NAT and firewall traversal, use ICE/STUN/TURN. SIP as although the technology is still developing, the fundamental requirements have been extensively embraced, thoroughly standardised, and adopted and put to use [1].

III. OVERALL WEBRTC AND SIP SYSTEM ARCHITECTURE

A. Integration of WebRTC with SIP

WebRTC technology enables browser peers to start RTC communications across web browsers. The SIP is not strictly necessary for WebRTC, as was previously mentioned, but it does need some sort of signalling system. JavaScript WebRTC application developers are free to choose. Building a WebRTC communication environment, though, raises the question of how WebRTC and "old" SIP systems will work together. A real-time multimedia connection can be started immediately from a web browser through a WebRTC application interface as a result of difficulties being resolved, which gives the ability to integrate SIP with WebRTC. This WebRTC application enables browser-to-browser and browser-to-SIP RTC sessions, in addition to backend server communication entities [1].

The JavaScript-based WebRTC application is simple to integrate into any type of web page, web based e-learning LMS system, etc. Using additional programming APIs, the entire communication environment can be simply modified to meet user demands and expectations (HTTP or SIP APIs), In order to merge WebRTC and SIP, a number of challenges pertaining to interworking at the media and signalling plane must be resolved. Each plane anticipates a different approach. The difference between the WebRTC decentralized application and centralized aspect is highlighted in table 1.

Table 1: Difference between decentralized and centralized conferencing

Decentralized Peer to Peer conferencing	**Centralized conferencing**
Current operational paradigm does not exhibit any form of incorporation of extraneous software entities.	The convoluted and intricate software is surrounded with the implication and entanglement of tertiary software programs.
The software can implement a steady increase or reduction in scope and efficiency.	Centralized conferencing applications would require integration of various products like Flock, Discord, or Slack for any changing circumstance.
Charges levied by intermediaries for the provision of services related to the transfer of goods, services or assets between parties is minimal.	Charges levied by intermediaries is high due to involvement of more number of mediators.
The metric denoting the typical time interval required for a given.	The metric denoting the typical time interval required for a given

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Through a meticulous implementation of sophisticated protocols, it is ensured that the fundamental right of privacy is duly preserved, while simultaneously delegating a commensurate level of control to the end-users.	The depiction of matters concerning privacy and control evinces a complex problem where the fundamental right of privacy is not preserved, and fails to delegate a majority of control to the end-users.
Basement consists of a system is designed to harness the power of a complex network of cutting-edge smartphones and PCs, meticulously coordinated and synchronized with a series of decentralized servers.	Centralized conferencing application necessitates the utilization of highly sophisticated centralized servers, which are specifically designed to process and store vast amounts of data with the utmost efficiency and reliability.
system or network to transmit and receive information, commonly known as latency, has been determined to possess a mean value of 100 milliseconds.	system or network to transmit and receive information, commonly known as latency, has been determined to possess a mean value of 300-450 milliseconds.
Ex: Proposed App – Basement	Ex: Zoom, Google meet

B. Identification of system entities

A number of system components are needed for the WebRTC and SIP technologies. We must combine them and address their interoperability when creating a complicated integrated communication environment. SIP is a well-known technology.

There are a number of ideas on how to construct a sophisticated RTC SIP platform using either free source or for profit SIP components. One such SIP communication solution which is primarily composed of open source software components is described. The underlying layered redundant network design and high availability algorithms make the proposed SIP communication platform redundant. The platform enables the design, development, and deployment of new integrated services. It can deliver communication services including voicemail, conferencing, voice and video over IP, and presence-aware instant messaging. The system permits signalling and media flows to get through NAT and firewalls (STUN, TURN) [1].

Based on IETF standards, the SIP communication platform makes use of open source software elements. The topic of extending a SIP infrastructure with components to enable the integration of SIP with WebRTC will be the subject of our next discussion. Three key components are needed for the WebRTC technology itself:

• RTC sessions can be started using a WebRTC client.

• During the beginning of a WebRTC session, the signalling server manages the exchange of session control information.

• Web server that houses the application code for the primary WebRTC client.

The WebRTC client should either be directly accessible via a URL or integrated within a web page or web portal. We must make sure that the signalling and media planes can communicate with one another in order to realise the WebRTC/SIP integration. As we have already stated, SIP will be the primary signalling protocol used. The request consequently calls for

• The signalling features of the WebRTC client will be implemented using the WebSock SIP API. This enables the start of sessions between WebRTCs or between WebRTCs and SIP.

• SIP signalling server with support of the Web Socket protocol.

The WebSocket protocol will be used by a WebRTC client as the signalling transmission mechanism. As a result, we require a SIP entity that combines the SIP protocol with the WebSocket protocol. A SIP proxy server with WS interface can be solved by this need. Since both technologies use SRTP and its AVP characteristics, the ideal scenario doesn't call for any specific media plane components. However, the WebRTC standard mandates the support of protocols that are currently not extensively included into SIP vanilla clients. In this situation, a translating mechanism and some sort of media gateway functionality are required [1].

C. System architecture

With no servers in between and latency cut to one- fifth of server-based solutions, we aimed to create a peer-to-peer video conferencing system named as Basement. On the other hand, users can utilise Basement as a tool for video conferencing, as network miners, or as both. In the future, we intend to offer its users financial incentives via cryptocurrencies in order to support the calls on its network. A network like Basement gains value as more people join it, which also raises the value of the tokens on that network [2].

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D. Current Implementations

In favour of promoting WebRTC and SIP as plat- forms for advanced learning, communication, re- search, and experimentation, we neglected commercial products. We are examining the supported protocols and the products that are offered for free download or distribution under an open source licence. New technology called WebRTC has varying levels of support across the major web browsers. Chrome, Mozilla, and Opera web browsers all support WebRTC. Support for its products from firms like Microsoft or Apple has not yet been announced. Microsoft offers a different approach. The WebRTC for all project aims to address general interoperability issues and ensure that WebRTC is supported by all web browsers [1].

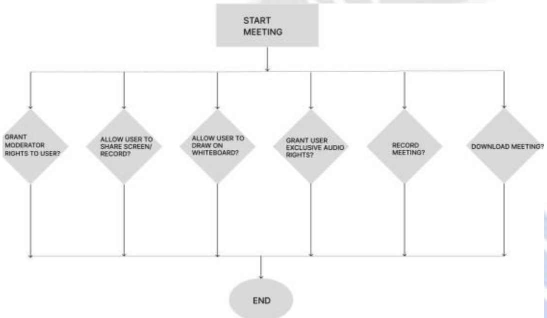

Fig. 3: Meeting Room Display.

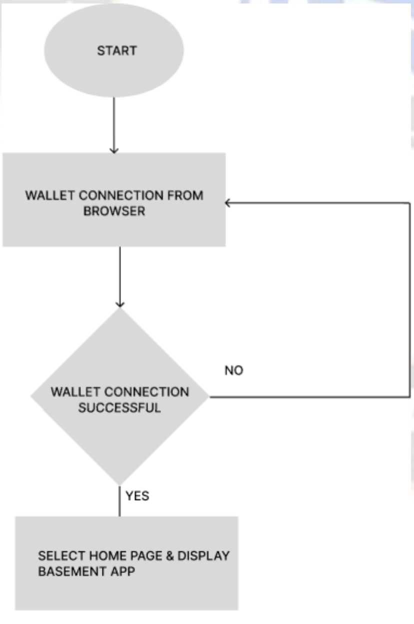

Fig. 4: Establishing Wallet Connection.

IV. CONCLUSION AND FUTURE WORK

The internet is currently in a centralised era. Being run by a small number of the enormous corporations that have complete control over your data, conversations, and thoughts. The goal of Basement is to user in the era of real-time communications on the web 3. The goal is to safeguard your digital identity while maintaining a secure on- line presence. We think that everyone should have access to a digital place where they can freely express themselves and let their ideal speech emerge. Our goal is to develop a cutting edge technological platform that will shape real time online participation in the future [5]. WebRTC is a novel and forward thinking real time communication technology, along with other pertinent technologies like SIP and WebSocket, in the paper. Then, we went over a notion that resulted in the expansion of a SIP platform with the WebRTC communication functions, leading to a new type of integrated communication environment [1]. Due to the deployment of peer-to-peer or peer-to-group connections in real-time communication, anybody can construct their own website or application, such as a real-time file sharing platform, real-time messaging environment, or video/audio conferencing platform. This made it possible for programmers and developers to compete with proprietors of social networking sites on the real job market. With the help of straightforward JavaScript APIs and Node JS, we can build a web-site with the most advanced capabilities, enabling each user to connect to one another via text messages and video/audio calls. the computer hosting Google's STUN and TURN servers [3].

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