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Real Time Information and Communication Center based on webRTC

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Abstract- This paper presents how to do Advance communication between two or more browsers without having any issue. Wieb RTC API (Real-Time Communications) and Asterisk virtual PBX, Using P2P (peer to peer), Real time outbound calls, video conferencing with two or more browsers and Live chatting, File sharing and also QoS Characteristic Asterisk Call center Engine for handle the SIP calls which are coming outside to the network, Using AGI Scripts for handle the route of the inbound calls and also give some instructions to the call flow. Handling files and support to download files automatically.

Index Terms- HTML5, VoIP, WebRTC, QoS, Asterisk, SIP, File sharing, Real Time Communication

I. Introduction

Real time communication (RTC) allows to find information areas like social media, live chat, video conferencing and unified communications and collaboration are all possible today because of developments in real time communications. Analog phone becomes to the smart phones for make communication easier, also created SIP phones to make calls with IP addresses through the Internet, For the real time communication between internet browsers, there is a open source project called webRTC. WebRTC enables mostly peer-to-peer communications over the web without the installing a client software or plug-in. This is done by leveraging HTML5 and JavaScript in the browser. Asterisk also an open source project which perform as virtual PBX (Private Branch Exchange) and found by the Organization called Digium. This PBX used to mainly handle calls which are coming from digital or analog Lines. There are some advance methods and configurations have to do in this virtual PBX before the run time.

By using these two open source projects and give a solution to make peoples day to day works get much easier by accessing a single web site. On this website people can make calls or video calls, Live Chatting ,File Sharing and also without typing any single word in Search Engine and can find the documents that want (eg: Passport Application, ID Application, Some Government related Exam Applications) by making simple call through the web browser. That call directly come to the IVR or Live Agents. After that Person can talk with this Agent and tell their requirement. Agent can track the calling party. Otherwise Form the IVR that call automatically goes to the correct location and download the Application or the Document to their pc asking the location that wants to save it.

In the sever side Asterisk can handle more than 5000 inbound calls simultaneously, each and every call it get a recode, That recode include, IP address of particular computer , Location (Address), Who is Answered, What are the documents did they get , Duration and a voice recode.

Addition to that System provides file sharing between browsers, do Live Chatting, get video calls or Conference calls. Without installing different kind of software in PC People can do Communications with their favorite web browser. This will help to fulfill people needs without wasting time in Queues or searching on Internet. All communication can do with this system. And only thing that people want is internet connection with a computer, Laptop or Smart Phone Etc. that can run your web browser

In parse II describes about previous works that was done by the researches Parse III describe our approach to implement a complete call center solution with using webRTC it includes Generate SIP calls, Handle the SIP calls, Live Agents, QOS characteristics and Automated downloads. Parse IV describe the conclusion and Pace V describes the future works .From this onwards II will describe background and related works , from III our approach will be explained and our approach has its sub categories as what are the features in client side , generate calls via SIP between web clients , webRTC and asterisk , QOS , asterisk and softphones , call flow , agents and track ,automated download . IV described conclusion finally future works

II. Background and Related Works

The pervasive increment of VoIP technology devices cause rapid explosion of various Communities applications, videos and online Television (TV) channel designed for users to increase their links in any network with the development of this technology introduces Web Real Time Communication (RTC). WebRTC allows No plug-ins, cross platform, low development and usage cost , browser to browser communication using peer to peer connection then it will avoid third Party plug-in dependency[1][8][12].

Instead of installation of separate software's, the desired communication features can be used in the browser immediately. This program built on JavaScript Sockets programming. Communication held on between two networks with real time video streaming feature with help of special protocol as well as reliable communication with protocols such as Session Initiation Protocol (SIP), Session Description Protocol (SDP), Hypertext Transfer Protocol (HTTP) and the Real-Time Transport Protocol

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(RTP) and the browser base open publishing(BOPlish) [2][10] [11].

Instead, the desired communication features can be used in the browser and immediately two browsers to communicate without any servers and transmitted data between two peers is encrypted by Datagram Transport Layer Security (DTLS) protocol to ensure the security of data [3][12] . A Web based real-time communication system, Hypermodal, based on the concept of temporal linkage between resources [4].Consumers and enterprises alike are rapidly adopting voice-over-IP (VoIP) technologies, which offer higher flexibility and more features than traditional telephony infrastructures[5][10].

Application of technologies for reception and transmission of streaming data in web browser: definition of the most popular technologies among the major participants of market of streaming data transfer, working with a web browser, the possible scope of application of streaming reception and transmission from a web browser, as well as the prospects for availability of a technology for end-users [6]. The advanced services like media mixing, session recording regards to videoconference must be included in the WebRTC. Furthermore, technical issues like bandwidth availability of networks, differences in networks and devices, screen size problems and processing power issues need to be solve immediately. The multiple control unit can be used to overcome above issues very effective manner [7].

The Information technology (IT) technicians need to prepare the systems before the session and need to check the availability of latest plugin within the existing system. The web browser cannot handle everything alone in videoconferencing. Furthermore, all the parties who join for the videoconferencing also need to do the same procedure again and again. The WebRTC make above complex steps to the task of button click event. The evolution made by the WebRTC is filled a huge gap in modern videoconferencing regards to traditional videoconferencing methods [9].

Asterisk is an open source project which is done by the Digium. This actually a virtual PBX and can handle SIP Calls which are coming from internet and can handle Calls which are coming from Analog Phones. This Asterisk sever can send DTMF (Dual Tone Multi Frequency) input to the computer that would serve as commands for execution of different programs on the computer and relay voice the output in the form of narrated voice to the user [13] Asterisk based voice exchange makes it possible for people to use their phones either fixed or mobile to call others in the country and internationally for a fraction of the prices they pay now[14]. In Asterisk There is very interesting part Calld IVR (Interactive voice Respond) from this system can route the call Automatically and find the destination of the call[15].

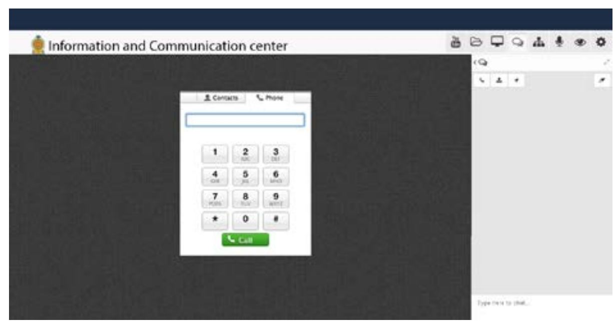
III. Our Approach

What are the features in Client side

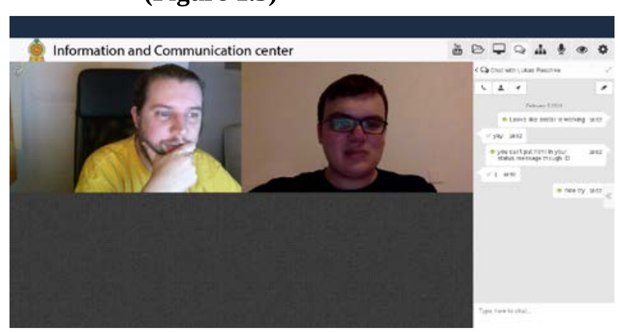
There is a small inbuilt Application in web browser (It can be an Application or simple website). From this Application client has different communication options Such as Live Chatting, Video conferencing, Talk to Live Agents with virtual softphone and virtual extension also client name and Get the related documents (Automatically Downloading). This is a simple Application and easy to use without having any pervious knowledge of handle sip calls. Figure: 1 explains the web Client Interface

(Figure1.1)

(Figure 1.2)



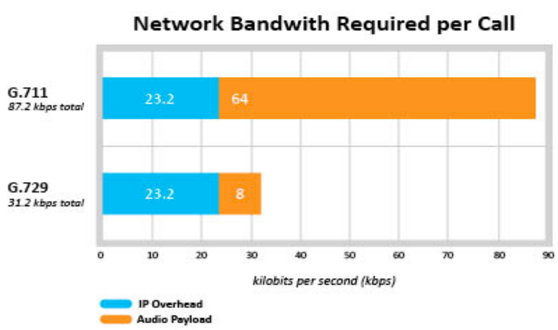
(Figure 1.3)



Generate Calls via SIP between web clients

For Generate SIP calls client should have working head phone and working Microphone. After client connect to the Application they can check their devices and make sure they are working. From User interface there are some options as explained in Figure: 1. Client click the button which is in the interface sip call will automatically generate from the browser (webRTC enable)

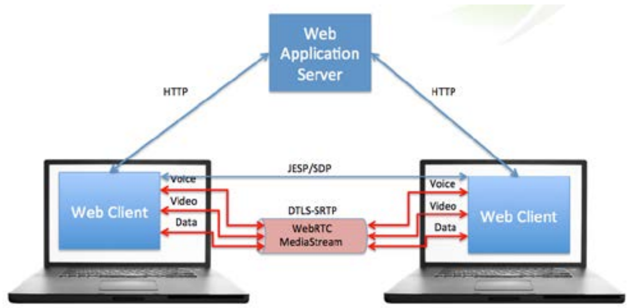
In default webRTC is supporting G.711 and Opus codes(Opus can handle a wide range of audio applications, including Voice over IP, videoconferencing, in-game chat, and even remote live music performances) for sending Audio stream. G.711 will give the best call quality for VoIP on the basis that it uses no compression at all, and as a result, the call quality sounds like using a regular ISDN phone. This Codec is supported by most VoIP providers. But when using G.711 codec it provides an uncompressed high quality voice, but uses a lot of bandwidth. To ignore this issue System use G.729 codec, G.729 is considered to offer a good level of call quality at a low bit rate of 8Kbps (kilobits per second), which would mean that you would be able to get more calls through your bandwidth that if you were to use the G.711 Codec. Different between G.711 and G.729 explain in Figure: 02



(Figure: 2)

For Video webRTC has VP8 and also H.264 codecs, VP8 has no limit on frame rate or data rate. It utilizes 14 bits for both width and height, which makes the maximum resolution 16384x16384 pixels. VP8 is often compared with H.246, a popular video codec that requires licensing royalties. Some commenters have noted the two codecs are highly similar in quality. When comparing h.264 and VP8, to get h.264 have to paid but VP8 is free

Using these codes (G.729 and VP8) and without using any plugins first implement a P2P communication system. Webserver doing the communication. And also Live chat system can be implemented by using Data stream without using any plugins as describe in Figure: 3



(Figure: 3)

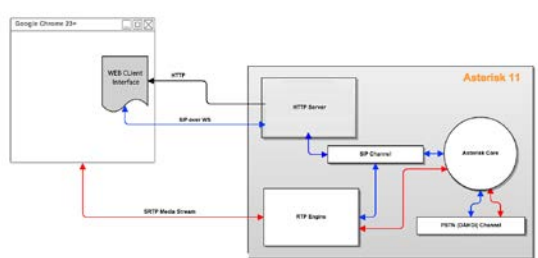
WebRTC and Asterisk

Combining WebRTC and Asterisk Call center Engine together it can make a good communication with web clients and Agents (who are provide a services to the clients). Asterisk is a virtual PABX and it can be hosted. When select the Asterisk version, 11 is better than other versions. And before install the Asterisk should build with

- res_http_websocket,
- res_crypto
- chan_sip.

After that need to configure Asterisk's built in HTTP server. Then select whatever bind port or bind address,. Port address that is used as bind port and address are used for talk over when using a Web Socket transport. After, configure the range of ports to use for RTP media. Then have to create a dial plan to handle the inbound call flow. Asterisk can handle SIP traffic which are coming from HTTP server and RTP engine, as a service System do not use any analog phones for the communication.

Through the firewall these ports should open. 5060 for SIP calls, some routers block the 5060 port. TO handle SIP calls and default port of SIP is 5060. 8088 For TCP and 1000-2000 UDP/RTP port range. In Figure: 4 describe this scenario



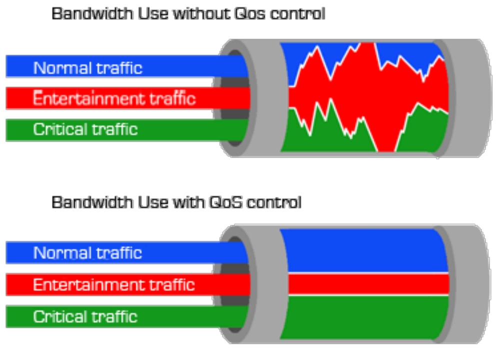
(Figure: 4)

(QOS) Quality of service between web Client and Asterisk

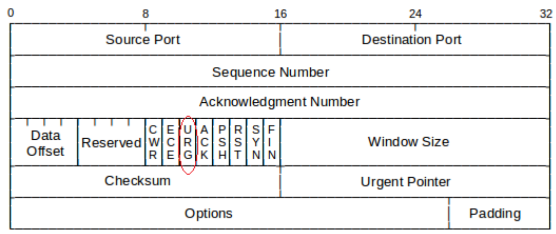
User open the browser log the site in any deferent environment and any kind of the device. When Request any information to the office talk with webRTC. Client provided their public IP address to QoS provider before. Router can access Asterisk sever IP address and Client IP address

If router verify IP address it will provide Quality of service. When data packet will going through in wire (layer 7). Increase rate of data packet throughput at packet go through wire. Delay time will reduces.

Asynchronous Transfer Mode (ATM), Ethernet and 802.1 networks, SONET, and IP-routed networks may use this technology. QoS arrange traffic in same band width. Three of traffics consider. This scenario is explaining in Figure: 5



(Figure: 5)

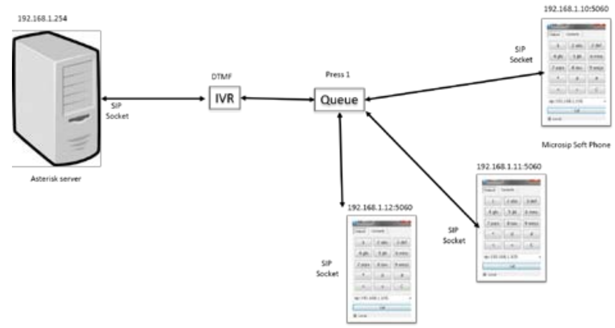


(Figure: 6)

If user log the web site and request information. Packet use TCP/IP protocol for voluble and reliable. If when any data packet will send add 1 bit to "urg" flag. In sender it is considered as Urgent pointer on TCP header in receiver's transport layer. Therefore packet will no wait receiver or sender buffer. It will gives huge priority in end devices. If data will wait application layer buffer or application layer buffer unauthorized percent will can receive refer to Figure: 6

Asterisk and Soft Phones

To communicating between Asterisk and Agents system using Softphones. There are many soft phones but system using Micro sip Soft phone. This soft phone also an open and customizable. Default this phone Support G.711 Codec but need to change it for G.729 codec. And block the video codes. This soft phone specially design for the windows platform. Connecting Agents soft phones to Asterisk server describes in Figure: 7



(Figure: 7)

Call Flow, Agents and track

There is some number of Agents for handle the calls which are coming from the web. Using dial plan and AGI Scrip system planned to handle calls. IVR (Interactive voice respond) is used for automated the system. When SIP call comes to the Asterisk server it can track the call using IP address and get the

(Figure: 8)

```
Dial plane
extention.conf
[general]

[globals]

QUEUE_1=Information
QUEUE_2=Documents

[default]
exten => 1011,hint,SIP/1011
exten => 1012,hint,SIP/1012
Handle inbound SIP call
exten => s,1,Answer()
exten => s,n,Monitor(wav,,m)
exten => s,n,Background(prompt press 1 for press 2 for)
Agent Queues
exten => s,n,Verbose(2,Call queue as configured in the QUEUE_${EXTEN} global variable)
exten => s,n,Set(thisQueue=${GLOBAL(QUEUE_${EXTEN})})
exten => s,n,GotoIf($["${thisQueue}" = ""]?invalid_queue,1)
exten => s,n,Verbose(2, --> Entering the ${thisQueue} queue)
exten => s,n,Queue(${thisQueue})
exten => s,n,Hangup()
Automated file sharing
exten => 3,1,AGI(Connect to the DB and get the informatin and send back to webcliets browser to download)
exten => 3,n,Play(Thank you)
exten => 3,n,Hangup()
```

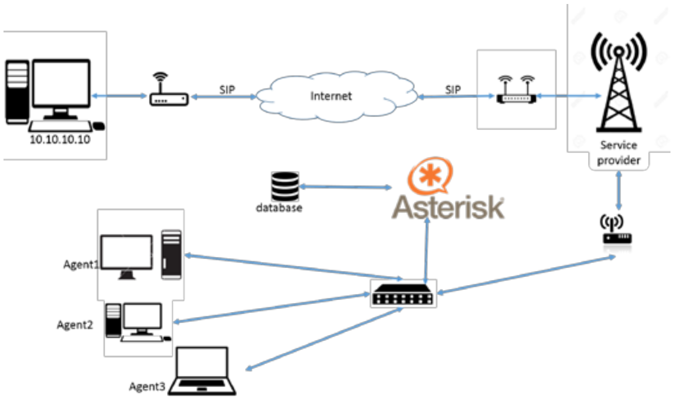
Location and customer provide details. Then save that details in to the database. According to the route of the call run a shell script or AGI scrip and get the documents which are request by web client, from the database. Then send those Files to the customer's PC for the downloading. Without getting a call. Web client can do a live chat with Agents and then also Agent can send the information or the documents that client requires. Tested Dial plan shows in Figure: 8

Automated Download

This is something interesting part which planned to integrate with the system. After SIP call coming to the Asterisk Server It can route to the IVR or Live Agents. When calls goes to the IVR client can get the

Information or the document that he want by Pressing virtual soft phone which is in the web Application, Asterisk can Identify DTMF digits and route the calls according to the DTMF inputs. Using AJAX SIP call permissions and AGI scripts system

try to get information from the database and send it to the browser and automatically download to the client PC
Real time Communication and Call center shows in Figure: 9



(Figure: 9)

IV. Conclusion

This paper introduces Using Asterisk Call center, webRTC build a complete call center solution. Any web Client coming to the system can simply click the buttons and get calls, this is a totally free service. Providing this solution to the web clients who are busy with their schedules can collaborate people who are in live and get information and documents that they want. Not only can that most of the organizations private companies Hospitals use this solutions for make their client's satisfaction.

V. Future Work

In future works system hope to hair Agents As part time workers by providing an Android Application with inbuilt soft phone. Then Agent can handle All SIP calls which are coming from the Asterisk server and send the files that client requesting from them.

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