Campus Radio Using Internet

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***Abstract*— Internet is a global source of communication with client server based application to access any type of data through the network. The goal of the application is to give the radio based solution through Internet. The software based radio application provides the radio activities in the campus using Internet with android application configured in clients. The mobile clients from android mobile phone must have installed the application and configure with the server by its respective port. The clients are configured with server by socket programming.**

At server side real time voice is recorded in .wav format at a sampling rate of 8KHz and converts to .mp3 format. The audio .mp3 packets are chunked and executed at the server. The executed packets are converted to User Datagram Protocol (UDP) packets and transmitted with the rate of 256BPS to destination port address of all configured mobile clients. The UDP stream packets are received and arranged at the client side with the sequence number and play the audio. The application can be used in the college to broadcast the voice notification to the students in the campus. Also it can be deployed in the marketing places like malls to broadcast the voice based advertisement to the customers. Therefore Radio in campus gives a cost effective solution to build private radio using Internet.

***Keywords*- Digital Audio Broadcasting, Frequency Modulation, Internet Protocol, Quality of Service, Real-Time Transport Protocol, User Datagram Protocol, Voice over IP.**

I. INTRODUCTION

Now a day's audio streaming plays an important role to play audio from different servers. Initially the FM radio where used to broadcast audio from different location. In FM radio the digital audio need to convert into analog signals and that signals was broadcasted using Frequency Modulation technique. And the hardware used was FM transmitter to transmit signals. At receiver side the frequency of the station should be tuned to enjoy music from FM station. But to build our own private radio station is cost effective to broadcast signals. So in recent technologies Internet radios are built to avoid cost & can build our own private radios. To broadcast the audio packets the socket programming is used to develop campus radio application. The server listens to the client connection request and accepts the connection if everything goes well. This application helps to broadcasting audio using internet in the field of marketing, education and to build other private networks. If the college needs to build Campus Radio, this application can be used to broadcast the audio notification using Internet and the clients can access using android mobile application. In marketing or malls the advertisement can also broadcasted using this system.

The microphone is connected to the server to record the real time voice and transmit the voice data. The socket is created between server and Client. The client request the server by using the port address and the connection is established between the server clients. The client is served by using IP address and port address. At the server side the authentication is done to record voice notification of college from specified users. Then the record window is appeared to capture voice and broadcast using the sockets.

The Campus radio application works with respect to OSI reference model. There are seven layers in OSI model. The Physical layer deals with physical connection of the device. The signal transmitting and receiving takes place from physical layer. The data link layer provides the reliable audio data transmission between the systems. The Network layer take the complete audio file and divide into datagram or the packets. The Transport layer provides the audio packet delivery to the system form server to client. The Session layer keeps the clients audio streaming in synchronized passion by creating the session between server and client. The presentation layer, provide the compression of the audio data packets and then convert the data network format. The application layer helps the user to interact with the network for broadcasting the audio.

The packet header format used for broadcasting the audio is the RTP format. It consists of the minimum 12 bytes header and after that it may contain the extension header. Then followed by the payload and used to determine the class of the application. The header consist of the version of the protocol used for the transmission, padding to indicate the end of the RTP packet, payload type to know broadcasting the audio or video and the timestamp to know the length of the data packets or the audio file.

II. LITERATURE SURVEY

The below shows some existing technologies which does the audio broadcasting by using wireless networks. These technologies are having some advantages and disadvantages.

A. FM Radio

FM Radio [21] was the first technology used to broadcast the audio. The audio file is converted from digital to analog and then the analog signal is modulated by using frequency. The transmission takes place by using FM transmitter hardware. The client can tune the frequency and listen to music. But to build own FM station the cost is more and there are some legal issues in transmitting FM signals.

B. PHP Radio

PHP Radio is audio broadcasting through internet. The Internet is used to connect from client and server [4]. The client need to request the server through HTTP request and audio packets are sent through HTTP response to client. The server side scripting languages is used to server clients. The problem with this is if the internet connection is lost and server will be busy when more clients are connected.

C. Digital Audio Broadcasting

In this analog audio signal is converted into digital audio files [10]. This type of audio broadcasting is done for multiple clients which provides a good audio quality. The system needs to maintain an input signal strength if it falls below the threshold than the system fails to broadcast the audio.

D. Voice over IP

The recorded voice can be transmitted using the Internet Protocol which provides voice communication peer to peer using internet. The Internet calls are carried out by using VoIP protocol [20]. The voice packets are formed and transmitted over network. The drawback of this protocol is that low quality audio are broadcasted using IP it can broadcast high quality audio files.

From the literature survey, it is observed that to build the private radio like FM and DAB is not cost effective, because the hardware used to deploy the base station is of high maintenance and high implementation cost. Also radio application using socket programming through internet is not carried out.

III. PROPOSED SYSTEM

The proposed system can broadcast the audio file of large size (.mp3) over internet network. The input of the system is to audio file which is processed by dividing into small chunks [14] and by using UDP protocol the packets are transmitted to clients who are connected to network. The below figure shows the components of proposed system.

The Components at server side consists of the Playlist where the admin adds the songs. The playlist wise the broadcast songs are divided into chunks and datagram packets are used for transmission. The client side component consist of the buffer to store the datagram packets form streaming packets and the datagram are converted into audio file format to play the radio in client mobiles. Once the stream bytes are played at client side the stream bytes are deleted from buffer and allocate space to incoming packets.

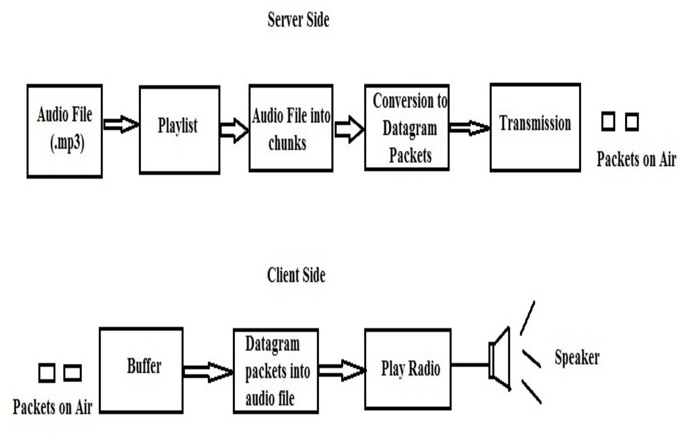

Figure 1.Components of Campus Radio

A. SYSTEM VIEW

The below figure shows the system architecture for broadcasting audio using Internet.

The implementation steps are :

- The admin from server record the real time audio to send the voice notification in the campus.
- The audio files are divided into chunks, if audio file is of mp3 format then we need to compress the file and divide into chunks . For wav format the audio can be divided directly from chunks.
- For transmission of audio chunks we use UDP protocol which does the transmission of data packets.
- Before the connection between client and server need to takes place before the transmission of voice bytes.
- The client send the request to server to connect to the using socket port and internet protocol address(IP).
- The server creates the connection and then transmit the voice byte to client buffer.
- The client streams the voice bytes and listen the voice which is broadcasted from server.

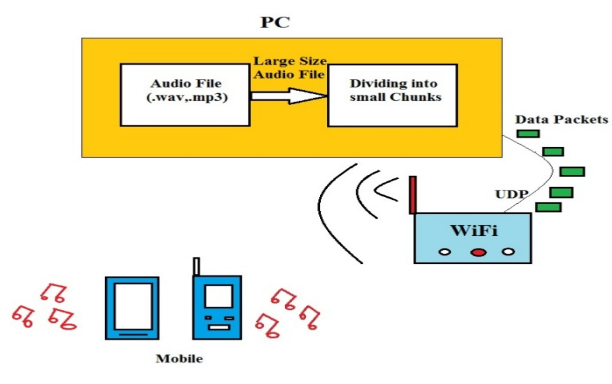

Figure 2.System Architecture

The Campus Radio uses the OSI reference model for broadcasting of audio packet in the network. There are seven layers in the OSI model, the physical layer, defines the physical medium used for communication like cables. In proposed system the wireless medium WiFi is used for the communication, this works in physical layer. The Data Link Layer take input as audio file and converted in to network suitable format, i.e data gram packets. The network layer is responsible for transmission of audio packets from broadcasting server to the multi clients connected the radio server. The Transport layer is responsible for the ordering of the audio packets in the transmission queue at server side. The session layer is responsible for creating the session between broadcasting server with the multiple clients. The presentation layer transmits the datagram packets to the application layer suitable format. The application layer interacts with the users and presents the audio data to the user.

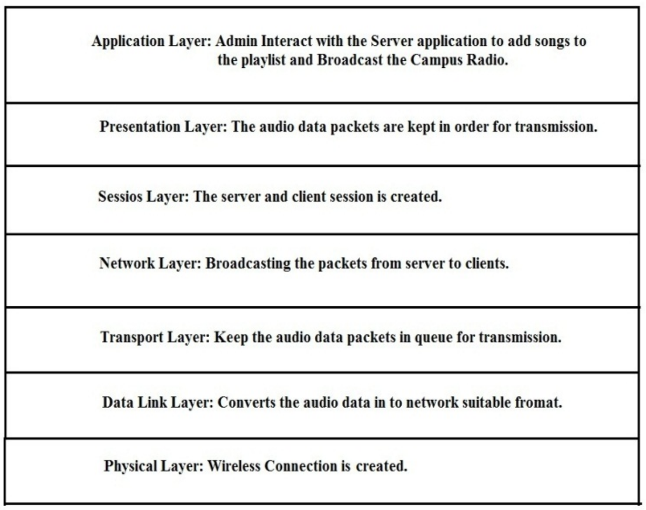

Figure 3.Working of Campus Radio with OSI model

A. System Modules

The software based radio over internet consists of software modules to help broadcasting the real time voice over the IP network and Android application. There are three modules, they are Server module, Client Module and Voice data conversion into UDP packets module. These modules work together to complete the system transmission of voice packets over network. The Client module consists of android application that is connected to the server to listen the real time voice notification in the college campus. The voice packets are in the form of digital the packets need to converted data packets to travel through network.

The Server side uses the all the seven layers for communication. The UDP protocol works to create the datagram packets in the network layer and the IP works in the network layer. The socket creation is done on top of the IP at server module. At mobile client side module operation takes place in three network layers, they are physical link layer, Transport Layer and the Application layer. The Physical link layer is the combination of physical, data link and network layers. The connection between server module and client mobile is created using physical layer. And the conversion of datagram packets to mobile suitable format takes place in this layer. The Transport layer maintains the connection, level of service and status with the server module. The mobile Application layer is combined with the session layer and presentation layer. The wireless session with the server and transport the order packets to the application layer. The application layer plays the songs receive from the presentation layer in sequence.

1)Server Module: The server module is connected to the microphone and only authenticated person can record voice for transmission. The admin user needs to enter the username and password. If they are correct then the server side application gets started. The Configuration at server side is done using the below methods

- Server Socket create ( ): The server side socket port is created on top IP and IP works in the networking layer. Socket are created in the network layer, the configuration is done with the combination of server IP address and socket port.
- Accept ( ): The next operation performed by the server is the accept method. This method used to accept the request from the clients using the socket port.

2) Client Module: The android session is created by using the socket configuration method used for the android mobile clients. The method used is shown below

Socket(String HostIP address, int port);

- The String HostIP is the IP address of the broadcasting server.
- int port is the socket port address of the server.

The Client module consists of android application that connects the server application. The socket port is used for transmitting the voice bytes. The Client sends the IP address and port number to the server for connection. Then the connection established between client and server. The client receives the data from the server port and the output is listen in the mobile speaker.

3)Voice Data Conversion into UDP Packets Module: The voice from the microphone is in the form of analog. The conversion of analog voice into digital data done. But digital data cannot be transmitted in the network, before the digital data need to convert into bytes data or chunk data bytes. This conversion is done by using the User Datagram Protocol (UDP), this protocol helps to convert the digital data into datagram packets which can be sent in the network. The UDP protocol is more used for the transmission of real time voice or video data in the internet.

The method used for the creating the datagram packets using port is shown below

```
socket = new DatagramSocket(port address);
in = new BufferedReader(new FileReader(audio.mp3));
```

The first method creates the datagram packets Datagram Socket at the port address and the input audio.mp3 is read using the BufferedReader method. Then the datagram stream bytes are transmitted in a network.

```
byte[ ] buf = new buf[256];
DatagramPacket packet = new DatagramPacket(buf,
buf.length);
```

socket.receive(packets);

The above method shows to receive datagram packets at the client side. The buffer size is define to 256 bytes to store the stream data in clients buffer. The constructor DatagramPacket takes two arguments, a byte array and the buffer stream length. The byte stream are played in sequence from the buffer in client mobile. The datagram socket is closed using the socket.close( ) method.

IV. APPLICATION DESIGNED

The java netbeans is used to build server application and android eclipse is used to build android application.

A. Server Application

The below screen shots show the campus radio server application running. The server side user can add the songs to the playlist by pressing the Add Audios button. The Path of the song is stored in the playlist and the playback begins.

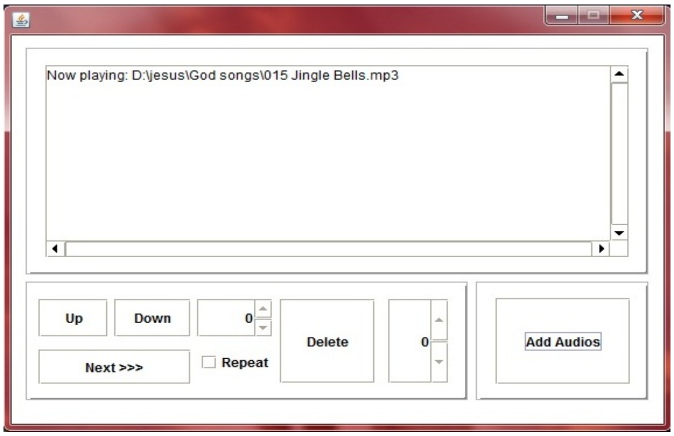

Figure 4. Server Application

B. Android Client Application

The android application consists of the two buttons to play and stop the radio. If the user press the play button the radio gets auto tuned to the radio sever and the buffering takes places, played sequentially.

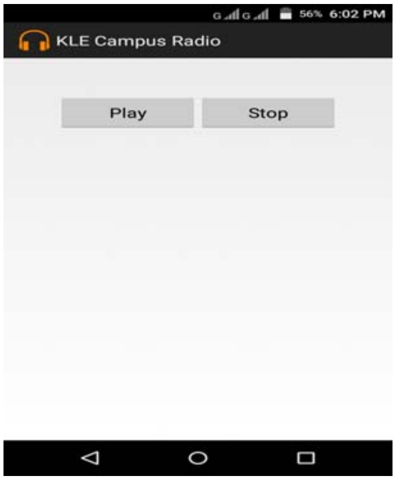

Figure 5. Android Client Application

V. RESULTS

Radio in Campus using Internet is tested in real time with the android clients and the broadcasting server running on the laptop. The application was tested for the twenty clients and the delay caused for the establishing the connection to the server.

The Networx tool is the freely available software for windows, which does the calculation of the number of bytes sent or received. The tool also has a graphical view and the normal digital view to analyse speed of the network.

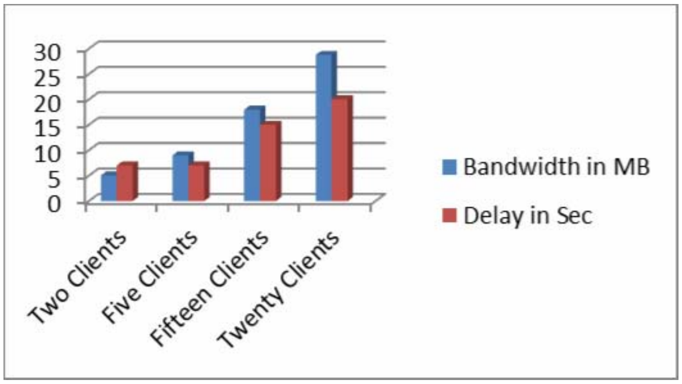

Figure 6. Bandwidth and Delay graph

In the above graph blue colour bar indicates the Bandwidth Analysis of the server and the test was made with different number of client's connection to the server. The server speed was 72Mbps with Wifi Hotspot form the Android Mobile. The Test was made for 1minute duration for all the clients connected.

The bandwidth usage is increased by two percentages as the number of clients goes on increasing. The bandwidth usage can be decreased by using the unused bandwidth in the college campus. For example the wifi routers placed at canteen or parking lot, the bandwidth is not used more. So we can use the bandwidth to broadcast the radio from less crowded places.

The server only keep on broadcasting the tracks and the clients need to click the play button. Then the client application takes time to establish connection and starts buffering. So the delay is occurred to buffer the stream packets.

In the above graph the red colour indicates the delay in the client application to start the radio. The delayed caused for two clients was less than the delay caused for Twenty clients. Because the bandwidth is shared among the clients and the request sent to the server will be delayed.

The Result Analysis of the Campus radio system, shows the increased performance and throughput of the system, when there is no bandwidth scares for all the clients connected. If the clients connected are more and bandwidth is less, then the jitter, traffic and packets loss is more.

If the clients are using the Campus Radio and at the same time using the internet to browse the web pages or downloading the files, at that time the packets are lost due to heavy traffic caused at client application.

The solution is that the Bandwidth used for transmission of Campus Radio is from the area where the students use the college wifi less, for example the parking lot or canteen area where the internet used is less. Then the will be no issues of Bandwidth problem.

VI. CONCLUSION AND FUTURE SCOPE

The goal of the application is to give the radio based solution using Internet. The developed application can be used in the colleges to broadcast the voice notification to the students in the college campus. And also it can be used in the marketing places like malls to broadcast the voice based advertisement to the customers. Therefore Radio in campus gives a cost efficient solution to build private radio using internet.

The Future scope of the application is to test the unused bandwidth in the network and use that resource for transmission of the radio over internet. This will help to broadcast the radio with less traffic, without the packet gets corrupted.

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