Bandwidth Optimization Through Real Time Audio/Video Codec Switching By Using IMS

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***Abstract*— The diverse access technologies used IP Multimedia (IMS) as a solution for providing robust multimedia applications and bandwidth optimization. Information sharing by the audio / video call through ubiquitous innovative applications and bandwidth utilization is gaining more popularity in the mobile world. This paper provides a novel solution for seamless connectivity through codec switching and bandwidth optimization by using an IMS server in real time. The mobile user will switch from audio mode to video mode and vice versa without losing connectivity. The technique used in our developed application will optimize the bandwidth utilization in an efficient way.**

***Keywords*—IMS, SIP/UDP, RTP, Codec Switching, Bandwidth Optimization**

I. Introduction

Innovative mobile applications provide opportunities to strengthen the fast growing and globally connected society [1] [2] Information dissemination through these applications has a huge impact on mobile users. The mobile community not only creates a ubiquitous environment for novel and useful applications, but also generates revenue for the companies. With advancements in this field, developers are now focusing to tackle the challenges of bandwidth optimization [3], high information costs and device processor limitations [4]. The emphases are about making applications more attractive and interactive according to the mobile clients' requirements, so that the mobile community can enjoy their benefits. Multimedia applications utilize optimal bandwidth [5] user interaction, elaborate and improve communication efficiency, and act as a milestone in viewing optimization [6]. For seamless connectivity, especially for the time critical services like media streaming, sophisticated protocols are needed [7] [8]. Different techniques for bandwidth optimization are used for better quality of services [9].

Many researchers have investigated about seamless handovers on different layers [10] [11] and worked on the SIP / SDP codec switching [12]. The impact of viewing is always more than a voice; hence developers are putting more efforts improving it by making it more interactive. Now the mobile users not only enjoy the services but also have an impact on different service providers through interactive video exchange with each other [13]. Easy access to internet on mobiles plays a pivotal role in developing user interaction applications. The rapid development in the processing power of mobiles and system architecture gives easy access to the network and provides a ubiquitous environment for development of novel applications. Mobile phone applications are providing many exciting services, like chatting, multimedia applications and information sharing on real- time basis. The presence of internet connected devices is beneficial to the users in many ways. Interactive application services are gaining more popularity due to easy approach to the internet. The new paradigm of mobile application reaches far beyond the capabilities of the Plain Old Telephone Services (POTS) [14].

Current systems of open IMS/SIP have not implemented audio to video switching and vice versa in an efficient way to gain seamless connectivity. Traditional telephony services quickly change their trends in the market to cope up with required modern services and functionalities. The development of the next generation network supports the new trends of the communication world and commences new audio and video standards. It is gradually overtaking the traditional telephone service because of its flexibility and cheaper costs although the mobile processing time and bandwidth issues are still big challenges to conquer. The introduction of innovative mobile application development by using IP Multimedia Subsystem Architecture for audio-video switching during the call would be an important milestone in the communication field. There is a need for users to share voice and multimedia (picture) data with each other, the solution to which has not been yet focused in an efficient way.

II. Module Introduction

As shown in figure 1 that by pressing the button, it initiates a switch call which is responsible for obtaining the required media CODEC. The button triggers the process of codec switching when required. The screen module at a lower level calls functionalities of another module. This module performs the operation of making calls for switching. It also creates the outgoing session with a server for negotiation and tries to establish the session with other mobile client. This module is also responsible for generating audio call at first stage, if the

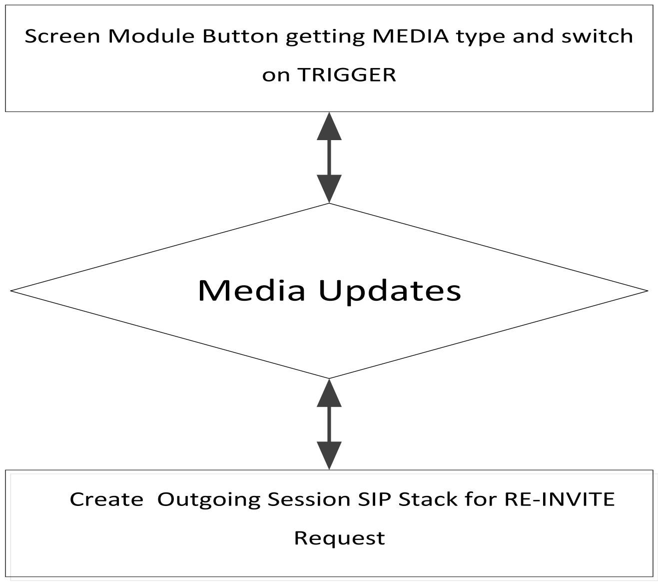

Figure 1. Block Diagram Of Proposed Module Of Seamless Audio/Video Switching

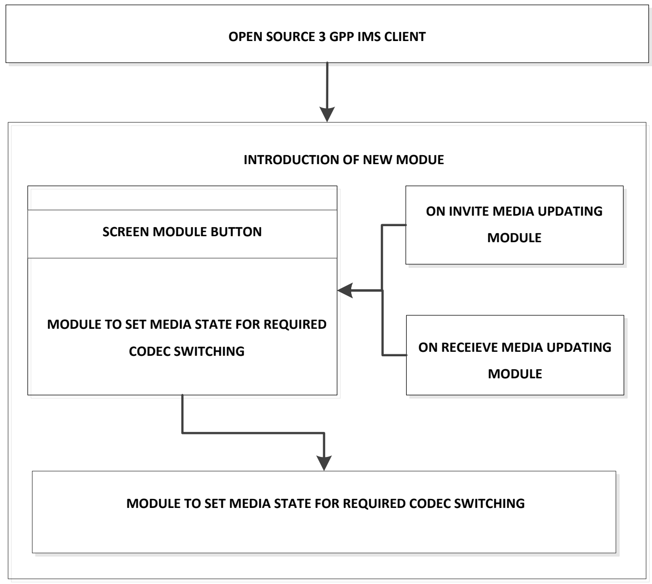

Figure 2. Audio-Video-Audio Call Flow

mobile client needs to switch from audio call to video call after establishing voice call, then it also requests for another function which generates call with other mobile clients through the IMS server. The responsibility includes calling functions which creates an outgoing session SIP stack for RE-INVITE request. The mobile clients keep sustaining the current voice or video call and simultaneously a request is generated to the IMS server for switching the audio or video codec. The IMS server then sends video codec to the clients, if they need to switch to video mode. During a video call if any mobile client wants to switch to audio mode then after pressing the button, the IMS server will stop the video codec and the mobile users will have a voice call session only. The resource allocation is a major issue for seamless connectivity. If the resources are not present for audio or video call then it is not possible to make or shift the call. It can also handle audio/video mobile clients for sufficient allocation of required resources. With the help of this newly developed module, seamless connectivity and switching between audio and video call is possible. This developed android application does not depend on the nature of the codec used in making audio or video call. The problem of codec compatibility or use of particular audio or video codec is not an issue by using this module. So this module provides assurance of seamless connectivity by switching of any audio or video codec during ongoing real time audio or video call according to the requirement of mobile users. Figure 2 shows the addition of new audio/video codec switching module. When mobile client wants to switch its audio/video mode then the client presses the button present on the screen. This will initiate the request of media update. The process is handled by a method present in the top layer of application which receives initial data from mobile client and then sends it to Screen module. The screen module handles incoming audio and video call by SIP event and Media event functions. The internal graphical user interface (GUI) is responsible to load the mobile view.

Different methods presented in this Screen module are responsible for establishing sessions, receiving calls and also handle media and sip event. It also initiates the GUI to load the mobile View to handle and set media type updates for switching state. The event INVITE functionalities are handled by event invite module and implements media updating to receive audio/video switching. On receiving media, this Screen module requests the lower level functionalities of the service implementation module and it’s responsible for updating the media through SIP event updates. It is responsible to set the media state accordingly for required code switching. The new module successfully integrates with other module [15] and the seamless connectivity is established without losing the connection between two mobile clients.

III. INSTRUCTION SEQUENCE FLOW

A complete description of the system and the flow is shown in Figure 3. In order to develop the connection between two clients, the caller first sends the invitation to IMS and IMS server sends the invitation to Callee (Instruction 1, 2 in Figure 3), while IMS server sends the "trying" message to the caller (Instruction 3 in Figure 3). The recipient accepts the invitation request and sends a confirmation message which shows that the callee is ringing (Instruction 4, 5 in Figure 3). The caller also receives back the ring tone. Both the caller and the recipient then send OK message and because of the handshaking, both establish a connection (Instruction 6, 6A, 6B in Figure 3). This call can be set up audio or video. In our case the Caller initiates a voice call (Instruction 7 in Figure 3). The Caller again sends RE-INVITE request to the IMS server (Instruction 8A in Figure 3). The IMS server then develops a route by handshaking on both sides (Instruction 8B in Figure 3). The Callee receives switch INVITE request from IMS server, and switch request is also sent to Caller for a handshake (Instruction 8C, 8D in Figure 3). Now the video call has been established between Caller and Callee (Instructi-

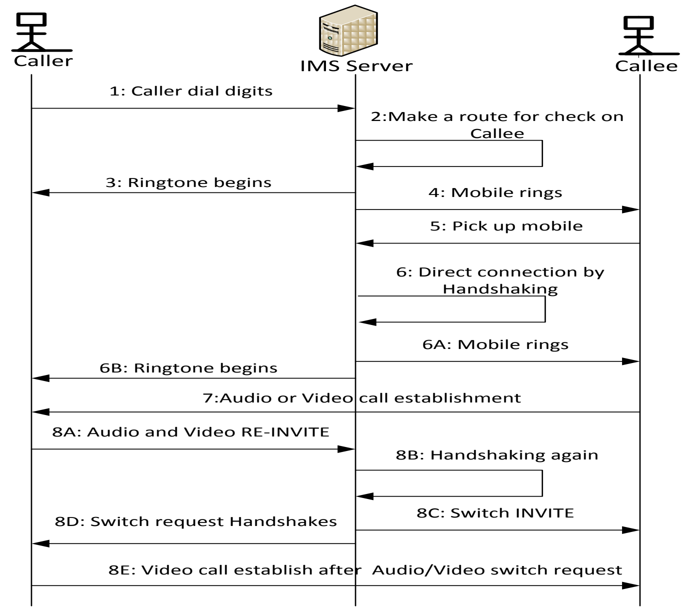

Figure 3. Flow Diagram Of Call Switching Between Caller and Callee Through IMS Flow At Different Stages

on 8E in Figure 3). The call can be switched to voice call from video call by repeating the same procedure.

IV. Required Components Of IMS Used In Our Developed System

When a telephone client dials to another client, an ad hoc connection is established between them on the IP network, but this facility is only offered in isolated and single service provider environment on the internet. Any two IP network terminals should have some medium of connection to communicate with each other. Therefore, there is a need of a system to connect the clients globally, and the IP Multimedia Subsystem Architecture (IMS) provides such a facility. It allows IP-enabled devices to connect peer-to- peer and peer-to-content in a very secured and easy way. IMS can access the service providers independently and hence it is not only global, but also provides standard IP-based connectivity. This architecture controls services and enables different types of multimedia services to the end users using internet based protocols which are common between them [16] [17]. Integration of voice and data services enhances overall services enhances overall productivity and effectiveness during the call. The development of innovative mobile applications which combine and integrate data, voice and multimedia will increase and create the demand of new mobile application services like push to talk, presence, conferences and multimedia chat etc. [18]. The combination of multimedia and IP networks will play an important role in the success of these services in future.

V. Flow Of Registration And Invite Request

Figure 4 shows the registration procedure from the IMS client to IMS Network. End User B sends the SIP register request to P-CSCF. The request then goes to I-CSCF from where the request is entertained by S-CSCF. Finally the registration request is stored in HSS.OK response is generated after successful registration by HSS and forwarded through S-SSCF. The OK response then moves through I-CSCF to P-CSCF and finally reaches to UE. IMS Registration is accomplished by a SIP Register request. By this stage, the registrar accepts the registration and creates a registration state.

Figure 5 shows INVITE request and its Message Body made up of SDP consisting of media information to up of SDP consisting of media information to establish a video call session during the call. The INVITE request is generated by UE1on SIP traffic ports to P-CSCF. From P-CSCF the INVITE is forwarded to S-CSCF on Diameter traffic ports. The S-CSCF gives OK replies. At this stage the audio call has been established the RE-INVITE 2 is sent to S-CSCF for video codec and then received Ok message. The RE-INVITE 2 is also sent to UE 2 and received the OK message. The results show that during the call the user can switch from audio call to video call and vice versa. When the call switches from video to audio then UE 1 and UE 2 can share audio codec only and video codec will be blocked. With the help of this test, it has been proved that seamless connectivity through audio, video codec switching is possible.

VI. EXPERIMENT FOR SYSTEM VERIFICATION

This application solution has been tested physically by using OpenIMS [19] server and two android mobiles. Wireshark [20] data analyzer has been used to analyze the data. The applicatio-

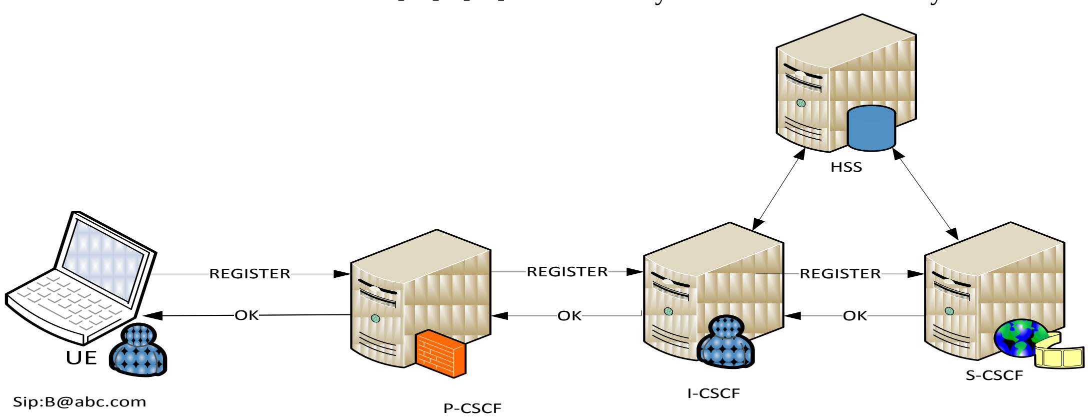

Figure 4. Registration From IMS Client To IMS Network

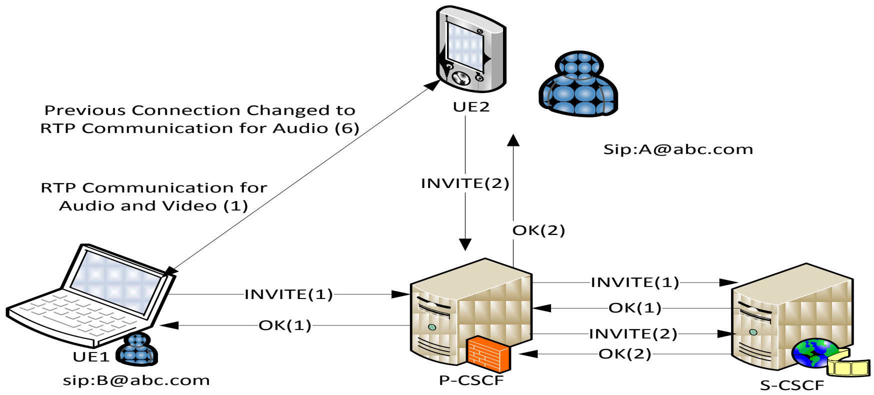

Figure 5. Flow Of Invite Request For Audio/Video Call Establishment

n has been installed on both mobile phones. To test its functionality and remove any doubts, Asterisk [21] and OpenSIP [22] solutions have been used. Here our developed mobile client and myMONSTER [23] clients are being used to test and verify the solutions with Wireshark and sip as packet analyzer. OpenIMS is an IMS core reference implementation for IMS technology testing and IMS application prototyping for research purposes, typically performed in IMS test-beds. Before any operation begins, there are several prerequisites that need to be finished to start the test.

Installed IMS server is used as a central data center and an application router. SIP Services used as authorization, conference, billing, presence, and interactive voice response with OpenIMS. OpenIMS has been interoperable with Asterisk and OpenSIP and if OpenIMS is used as first the point of contact, then this solution works out audio and video switching on open source IMS and VoIP / SIP services. As shown in

Figure 6 that IMS server will connect two mobile clients. When one mobile client will try to communicate with other mobile client then all the packets can be analyzed by data analyzer .Wireshark is acting as a data analyzer and is integrated with IMS server. This will show and give conforma-

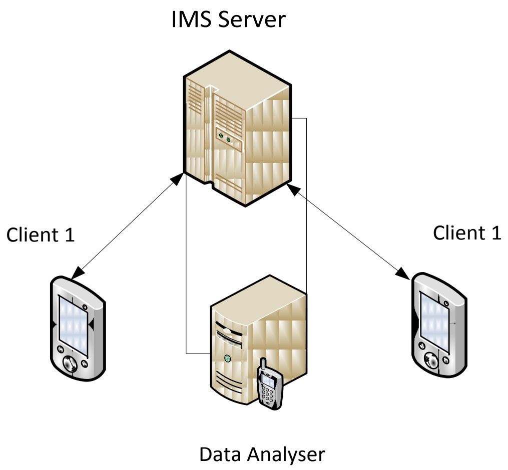

Figure 6. Test Components for Verification

tion of the codec switching. For evaluation purpose, OpenIMS Core is using abc.com as domain and A and B as two users to talk in-between. User A is a computer/mobile user, who is calling User B. User B is a computer/mobile to receive a call. At the start, user A calls user B through voice call. After some time, user A switches voice call continues. This shows a complete switching of audio to video at codec level too.

Figure 7 shows an overview of IMS architecture used in this codec switching verification test. Additionally, there are many components which cannot be added due to a main focus on the Components which cannot be added due to a main focus on the test.

Mobile User – A

Laptop User – B

SIP Application Server is used as OpenSIP or Asterisk or Application Server (AS) depending on research requirement.

Identification is the key to operate functionalities in a network. IMS has used similar identity profile from SIP URI. There are two different types of user identities

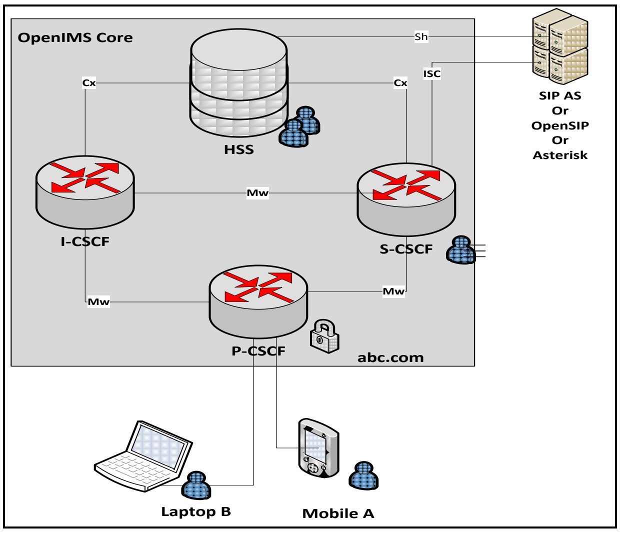

Figure 7. IMS Testing Components For Codec Switching Verification

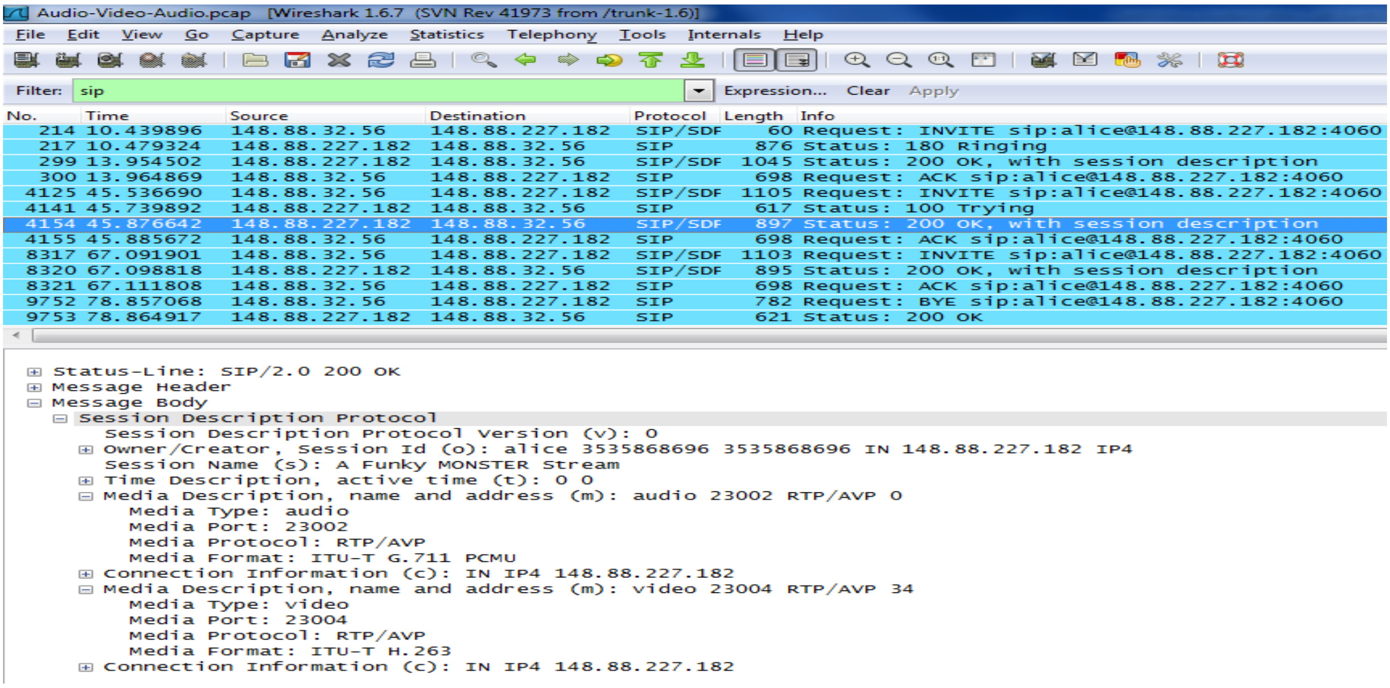

Figure 8. Data Of Voice and Video Codec Sharing During Voice To Video Call Switching

1. Private User Identities

2. Public User Identities

It is also possible to use a TEL URI. There are server identities similar to SIP URI to be used in AS.

In this test, DNS Server runs locally as abc.com which has its own configuration. After successfully running that DNS server, the other can be run on OpenIMS components pcscf.sh, scscf.sh, icscf.sh and startup.sh. There might be dependent requirements for running and configuring them which need to solve properly to run correctly on OpenIMS. Before starting test operation, it is necessary that an IP connectivity access network such as GPRS, ADSL, WLAN or LAN is connected to IMS Server for its end users to communicate between them. Here, this project works on Mobile Radical Wireless Network and University Local, Area Network. Both of them are already in the same domain on University Network. The ports can be monitored as SIP traffic ports and Diameter traffic ports, using Wireshark and SIP. After fulfilling the prerequisites

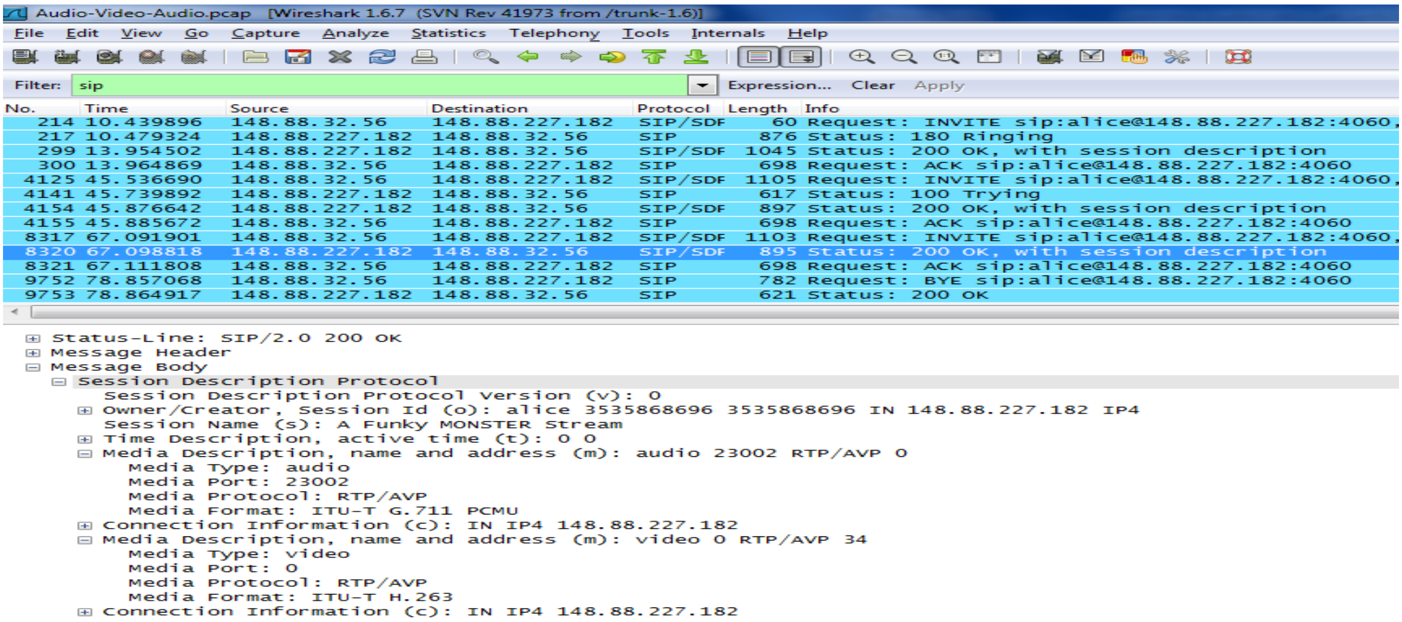

Figure 9. Data Of Video Codec Block, During Switching From Video To Voice Call

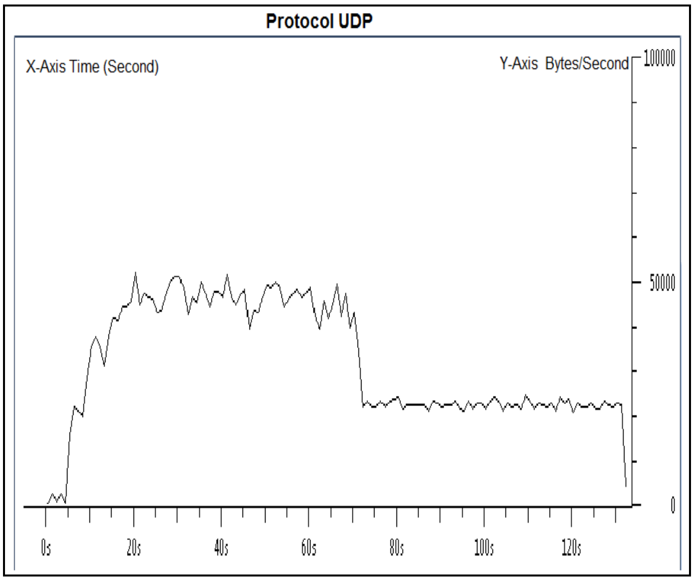

Figure 10 UDP Bandwidth Optimization Graph Of Our Proposed System

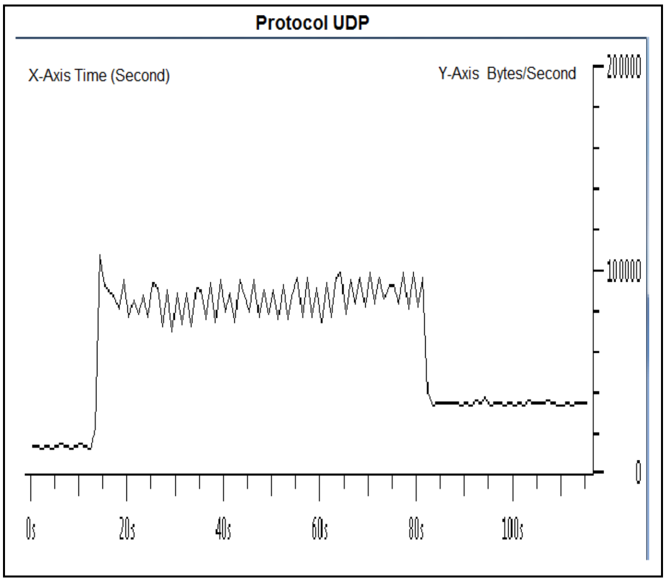

Figure 11 UDP Bandwidth Optimization Graph Of Other System

completely, end users can use IMS client terminal to connect the IMS network.

The data shown in Figure 8 explained the sequence of call information. It starts from voice call and then switch to video call without losing connectivity. Figure 8 shows that G.711 PCMU and H.263, audio and video codec are coming respectively. So video call established at this moment.

Data in Figure 9 shows that when call switched back from video to voice call only audio codec G.711 PCMU is coming while video coded H.263 is not coming.

Video packets are blocked when switched to voice call again. Different stages of information like INVITE, Ringing, Trying , OK and Ack shows that when the voice call is established then only the audio codec is coming and when the call switch to video call then video codec also start coming. At the end when call switch back to audio call then video codec blocked and only audio codec coming. The results showed that seamless audio/video call switching through IMS during the call is possible.

VII. BANDWIDTH OPTIMIZATION

After analyzing the data obtained after performing this test, it has been observed that the optimization of the bandwidth is also achieved with the help of this developed application. An efficient seamless connectivity through switching of the audio/video codec is possible. In Figure 10 a graph of UDP protocols is shown. The graph values are obtained by using our proposed system in above mentioned experiment. The X-axis on the graph is representing time in second while the Y-axis is showing bytes per unit second (byte/tic). On the X-axis From 0 to 6 seconds there is no call but the IMS server is connected with both of the clients. Initially video call is initiated at 7 second and we can see the change in UDP protocol as shown on the graph. It reaches up to 5000 bytes/tics. The graph shows deflection at the point where the change occurs. The video call continues around 72 seconds. At around 72 second one of the clients want to switch into audio mode, to accomplish this, the client just clicks it to audio call and IMS server will stop video codec. At this stage, only the audio codec is coming.

Figure 11 also shows the UDP protocol graph of other application [23] that has been used in this paper for comparison. It is obvious from the graph that on X-axis from 0 to 13 second there is no change in UDP protocol because there is no call established between two clients. At first when one client is trying to establish video call, we can see the change in the UDP protocol. From Figure 11, it is obvious that both audio and video codec are shared between two clients till 85second. During this period the clients are utilizing bandwidth for audio and video call and on the Y-axis of the graph, it reaches till 10000 bytes/second. . The change in UDP protocol is observed at 85 seconds when it is required to switch from video mode to audio mode. There is a difference of 5000 bytes/second between our proposed system and other application used in the experiment. By comparing the two graphs it is clearly shown that our proposed system optimized the bandwidth utilization whenever there is a change from video mode to audio mode or vice versa. As shown in the graphs that by using our proposed system, once the call is established, the application switches the required codec seamlessly and without any change in bandwidth. This is a novel and an important feature for service providers. They can increase the no of mobile customers by optimizing bandwidth and providing them good service.

VIII. CONCLUSION

The proliferation of bandwidth optimization and the ubiquitous development in mobile processing technology have engaged and enable the design and development of real time efficient and economically feasible multimedia application and bandwidth optimization. Bandwidth utilization and multimedia information processing has been thoroughly integrated many computational devices and systems simultaneously. Seamless connectivity is always an important issue in the mobile communication world. Processing time of the devices is also a major hurdle for developing future application which can be improved by proper bandwidth optimization. Bandwidth utilization played a pivotal role in modern communication world. Bandwidth can be optimized by using this novel application.

When compared with other existing applications our proposed solution can optimize approximately 50% of bandwidth utilization.

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