

A Model ITTP Architecture over Heterogeneous Network for 5g Application

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ABSTRACT

The advent of 5G technology is believed to have mutual basic requirements such as high reliability and extremely high data rates. Meeting up with such demand changes the overall fifth generation (5G) network architecture. 5G is aim at interconnecting all wireless Heterogeneous Network (HetNet) for seamless and reliable Quality of Experience (QoE) where every network is in control of its UE's mobility, while the network providers manages the operation at the terminal. The proliferation of Voice over IP (VoIP) applications over the years attracts the attention of a number of telecommunication industries. VoIP is targeting to overtake the existing traditional Public Switch Telephone Network (PSTN), with the packet based network, that can support multimedia services globally. In this work we proposed a new standalone Internet Telephony Transport Protocol (ITTP) architecture that unite the functionality User Datagram Protocol (UDP) and Real-time Transmission Protocol (RTP) to convey media data aim at providing better performance in terms of bandwidth, delay and throughput based on the proposed model equations as compared to the existing inefficient protocol. The protocol provides VoIP developers with options to augment new services and technique for other use cases and future work toward implementing the architecture is also discussed.

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INTRODUCTION

The evolving auspicious Fifth Generation (5G) mobile services is believed to have mutual basic requirements such as high reliability and extreme data rates (Muhammad, Yaro, Ya'u, & Abubakar, 2019; Muhammad, Yaro, Ya'u, & Salawudeen, 2019). Despite its miscellaneous application settings, meeting up with this compelling demands originate a rethink in the overall 5G network architecture and its infrastructure with respect to emerging wireless techniques (Song et al., 2017). For seamless and reliable Quality of Experience (QoE), 5G technology is aim at interconnecting all wireless heterogeneous network. In such a way that, every single network operator will be in control of its User Equipment (UE) mobility and the diverse network providers at the network terminal finally decides on how to manage the operations in the network. The transport protocol conceptualization in this scenario is different. Since,

losses in wireless network might be cause by high error rate at the radio interface whilst is assumed that, network congestion in Transmission Control Protocol (TCP) is tight to lost segment for wired network. This open door for researchers to view on a modified TCP that can retransmit impaired or lost TCP segment (Habib et al., 2016; Song et al., 2017).

The key behind 5G applications and requirements is to trace and recognize those requirements and have a good understanding governing their limitations. Nowadays, 5G networks is the leading challenger in 5G technology. The trend now is to remodel the network architecture as in Figure 1, that can handle dynamic nature of varying traffic pattern and support modern technology such as Network Function virtualization (NFV), Software Define Network (SDN), Network Slicing, edge computing, Internet of Things (IoT) etc. (Ding & Janssen, 2018)

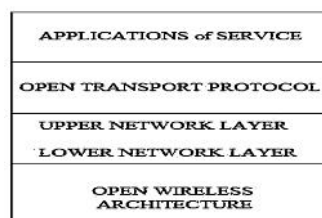


Figure 1: 5G Network protocol stack ("5G network architecture | 5g protocol stack," 2017)

5G will empower and diversify a wide range of industries from simple smart homes, sensitive and important healthcare services, up to the vital marketing and education sector. Other related fields includes Industrial IoT to smart farming, transportation to entertainment. The technology will provide much more interesting feature and this will provide some area of applications with regard 5G and introduces some basic concept and Applications of HetNets (Ding & Janssen, 2018). The recent performance improvement in the network equipment experienced numerous key transformation in the communication industries. The proliferation of VoIP applications in the past few years attract the attention of telecommunication sector, it is assumed that, by the year 2017 the amount of VoIP packets running on top of internet will surpass 158

petabytes monthly (Abu-Alhaj, Kolhar, Chandra, Abouabdalla, & Manasrah, 2010). Additionally, the CISCO global mobile data traffic forecast through Visual Networking Index (VNI) also provides another broad global mobile data prediction stating that, the amount of mobile data packets over internet will exceed 75 EB by 2022 as shown in Figure 2 that is about sevenfold increase over 2017 data traffic. Moreover, the Compound Annual Growth Rate (CAGR) is expected to grow at the rate of 46% from 2017 to 2022. This can be evidently seen from the notion that VoIP calls and services can be achieved at free and affordable rates through portable devices such as mobile phones, tablet and PC's (Cisco public Cisco Visual Networking Index: Global Mobile Data Traffic The Cisco® Visual Networking Index (VNI) Global Mobile Data, 2019).

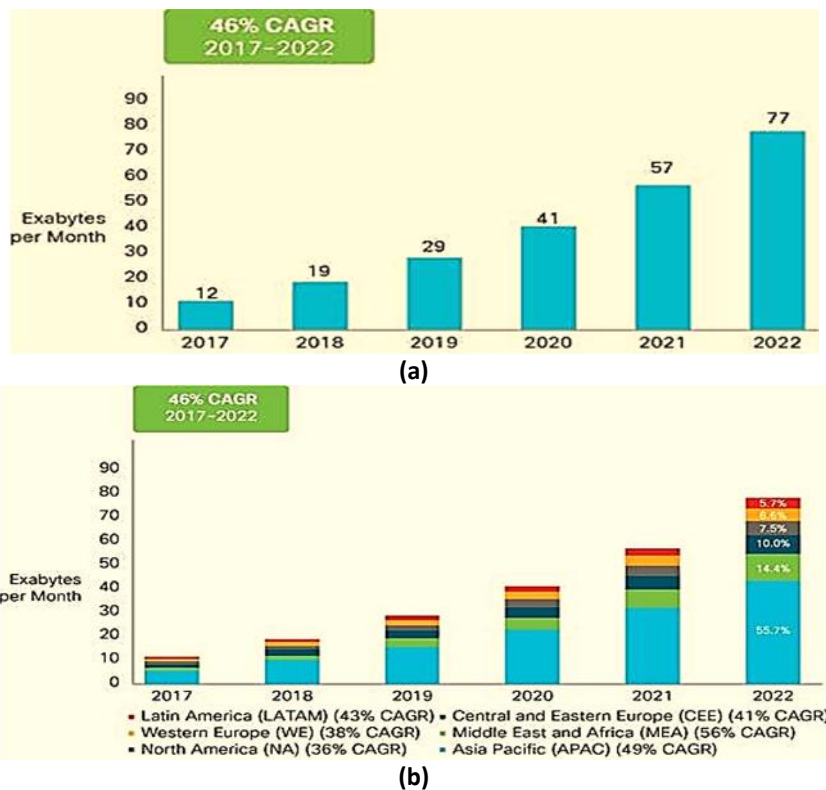


Figure 2: (a) Global Mobile Data Traffic, 2016 to 2021. (b) Global Mobile Data Traffic Forecast by Region (Cisco public Cisco Visual Networking Index: Global Mobile Data Traffic The Cisco® Visual Networking Index (VNI) Global Mobile Data, 2019).

VoIP is aim to overtake the existing Public Switch Telephone Network (PSTN), by replacing primitive circuit switched telephone network with the packet switched network, which can handle abundant multimedia services globally. VoIP protocol can be achieve through two major classification as follows:

- (1) Here calls between two or more end points is initiated and managed by the signaling protocols. Various standard and protocol were proposed for signaling protocol but only H.323 and Session Initiation Protocol (SIP) were standardized. H.323 was first

introduced by International Communication Union (ITU) in 1996 to harmonize the transmission of videoconferencing over IP networks, which was gradually dominated and overtook by SIP. SIP is a standard introduced by Internet Engineering Task Force (IETF) to carry out all kind of real time communications in 1999. Inter Asterisk exchange protocol (IAX) is another signaling that yet to be standardized (Abu-Alhaj, et al, 2012).

- (2) The second category is based on media transfer protocol that runs on session layer to facilitate the movement of all form of real-time media data among end points. One of the common protocol specialized in this area is Real-Time Transport Protocol (RTP) (Abu-Alhaj, et-al, 2012).

The telecommunication industries begin to integrate VoIP technology in their network hoping that it will entirely replaces PSTN in the near future as found in fourth generation (4G) access technology. VoIP offers various opportunities as compared to its PSTN counterparts. Amongst a few are: (1) the extensive growth of VoIP as an open system with free and open architecture, receive attention from considerable number of researchers and open doors to numerous business opportunities. (2) Ability to make free and subsidized rate calls globally. (3) Improved users QoE with abundant multimedia service such as voice mail three calling live video streaming, and live chart. (4) The VoIP works around software capability (In such a way that, damaged network links and out-of-order network hops will not affect its operation) making it more resilient and consistent. Figure 3 presents summary of VoIP benefits. (Ding & Janssen, 2018).



Figure 3: Benefits of VoIP (“7 Benefits of Hosted VoIP Service for Startups,” 2016)

The rapid development of VoIP over the years encounter several challenges, particularly in the area transport protocol. Majority VoIP data were transported over protocols that also run on another protocol which degrade its performance. Considering the viability of the VoIP technology lots of effort have been made to produce VoIP applications with optimized functionalities that can compete in the global market. In view of that a new transport protocol capable of conveying VoIP packet by its self is introduced. This protocol is referred to as Internet Telephony Transport Protocol (ITTP) (Abu-Alhaj, et-al, 2012; Edeline, et-al, 2016). This paper presents the architecture of standalone ITTP protocol, capable of conveying VoIP packet with low latency, enhanced throughput and high bandwidth for 5G mobile applications by exploiting multipath characteristics of Heterogeneous Networks (HetNets).

The remaining sections of the paper is organized as follows. Section II provide an overview

of 5G applications and features. A review of related works and some challenges with the existing protocol is discuss in section III, section IV proposed a new model protocol for VoIP with its new features and applications and finally a feature work and a conclusion is presented in section V and VI respectively.

5G APPLICATIONS

This section discuss the current trend of applications and features of 5G and HetNet:

High-speed mobile network

5G is expected to provide ultrahigh data rates of about 10 to 20 Gbps of download speed with concurrent efficient transmission of voice and high-speed data. Low latency in the rate of milliseconds vital for autonomous driving and other sensitive applications, is among the key features of 5G using a novel radio millimeter wave transfer that can support

very huge bandwidth (“Applications of 5G technology - RF Page,” 2018; Ding & Janssen, 2018). Mobile services will be more exciting with secured cloud storage accessibility, and open a new door for business approach especially to device manufactures and software developers. Which in turn creates more jobs and markets opportunities through the

emergence of VoIP and smart devices. Wi-Fi transfer and Device-to-Device (D2D) communication are also a part of the technology targeted to improve and sustain the access and mobile networks limitations, this is seen in femtocell, model for high data, low power and network cloud accessibility (Rajiv, 2018).

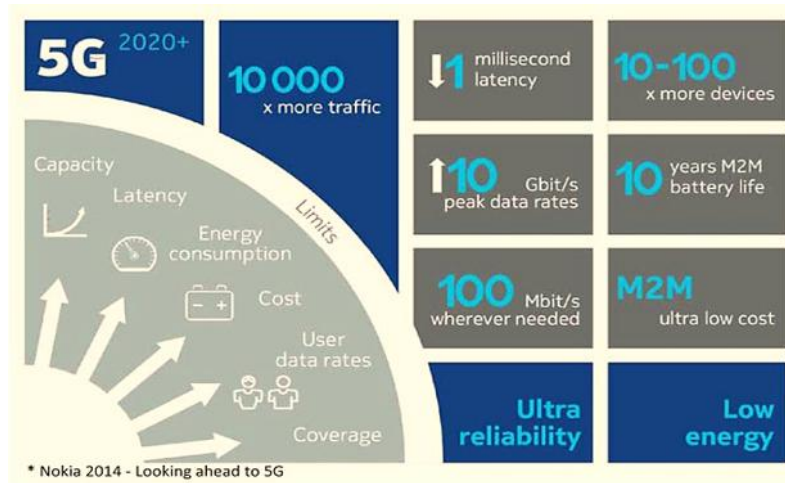


Figure 4: High-speed mobile network opportunities (“Applications of 5G technology - RF Page,” 2018)

Entertainment and multimedia

Multimedia services consumed about 55 % of worldwide internet mobile traffic particularly on video transfer in 2015. With the advent High Definition (HD), Video on Demand, down to Ultra HD or 4K video technology that are expected to feature in 5G. The 5G technology is believe to provide ultrahigh speed data streaming with extra crystal audio clarity at few seconds for live stream video for

business and entertainment industries. Another sector that is also attract the attention of researchers is Augment Reality and Virtual Reality (AR/VR) requirements that targeted Ultra low latency and high quality resolution video technology. In addition to that is HD virtual reality gaming that is getting much attention from investors (Ding & Janssen, 2018; Rajiv, 2018).

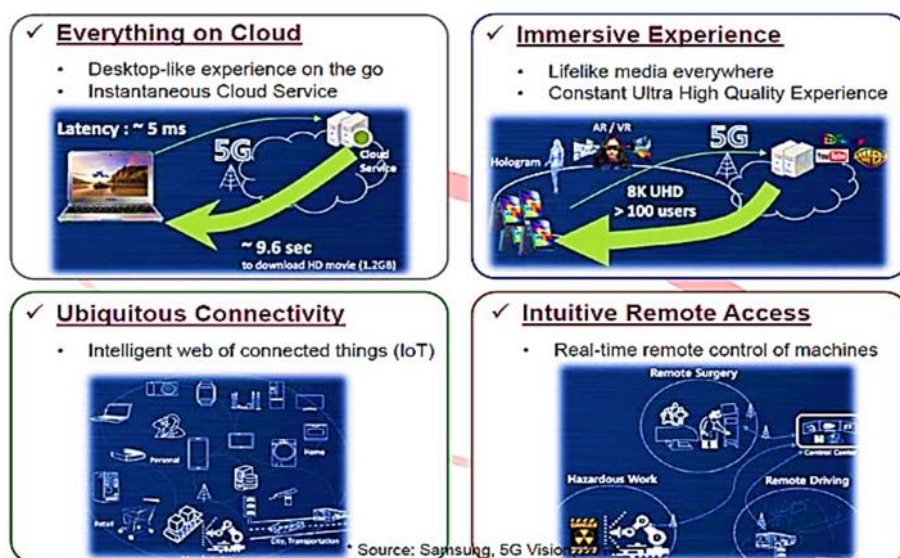


Figure 5: Entertainment and multimedia (“Applications of 5G technology - RF Page,” 2018)



into account bandwidth and delay asymmetry. The performance analysis model is developed through average throughput evaluation by dividing the TCP flow into a series of transmission round. The results highlight one major problem that has to do with packet reordering, and can lead to TCP throughput degradation. Two key performance improvement over heterogeneous multipath presented. The first one is by designing a novel scheduling algorithm to tackle the occurrence of packet reordering, and secondly, is to improve the existing TCP that can handle packet reordering by instructing congestion control to stop responding to packet reordering the same way to congestions (Song et al., 2018).

In another perspective, the possibility of UDP as a potential feature internet transport protocol based on UDP encapsulation conceptualization. Google's Quick UDP Internet Connections (QUIC) and Web Real-Time Communication (WebRTC) are two transport protocol that were used to evaluate performance. To investigate UDP blocking on access networks, RIPE atlas is used to measure UDP connectivity and first packet latency. Secondly, to ascertain the congestion traffic with UDP headers a copycat approach is deployed as one of the measuring tool to generate TCP traffic based on UDP wire's image in a wider range of test networks. The results shows that UDP can provide a room for the new transport protocol, on the basis of coexisting UDP with other alternates protocols such as QUIC. One major drawback in this work is access network link which leads to UDP impairment (Edeline, et al., 2016). Potentiality of VoIP start to evolve as Sze et al. Combine RTP/UDP protocols in such a way that VoIP packets that belongs to the same destination are coordinated into one UDP/IP header. The RTP header undergo compression before delivering the VoIP packet to the receiver endpoints. In the process each unique RTP header is maintain in every packet.

A decompression is perform at the receiver to retrieve the original message and the result shows bandwidth utilization improvement to about 72%. In another development Abualhaj et al (2010), presents an ITTP-multiplexing technique that coordinate packets in the same destination into a single IP header, and allow ITTP header to deliver the packets without any alteration. The simulation results record a bandwidth improvement to about 29.1% as compared to conventional ITTP protocol (Abualhaj, Kolhar, Halaiyqah, Abouabdalla, & Sureswaran, 2011; Abualhaj, Al-Khatib, & Baklizi, 2018). Furthermore, Mosleh et al. proposed delta-multiplexing technique

over ITTP to overcome the issue of wasted bandwidth from ITTP/IP packet header. In their work packet that belongs to the same destination are compile in single IP header in a process called ITTP-multiplexing and a compression mechanism is deployed on the header which are then forwarded to the destination, and a header decompression is perform at the receiving endpoints to restore the original packet. The results shows about 46.4% to 72.2% bandwidth utilization efficiency relative to header and packet size respectively (Abu-Alhaj, et al., 2010).

Although RTP and IAX can transmit media data on real time and can provide VoIP calls with much better efficiency, but suffered from inability to convey data independently by themselves. As such, they operate on top of another transport layer such as UDP to deliver their data (Abu-Alhaj, et-al, 2012). Despite IAX mini-frame precedence to VoIP data with lesser overhead as compared to RTP/UDP, the protocol still need a transport layer protocol (such as UDP) to convey their data. Secondly, IAX mini-frame is not yet recognized by RFC or any other standardization like middle boxes which limits its worldwide interoperability (Kayed, Abualhaj, & Alharibat, 2013).

Middle boxes implementations limits the acceptance and introduction of quite a number of new protocols such as Stream Control Transmission Protocol (SCTP). The universal deployment of middle boxes use cases such as Network Address Translator (NAT), firewalls, Intrusion Detection Systems (IDS), and load balancers etc. that only accept few numbers of protocols complicate the whole process of introducing new protocols (Carlucci, De Cicco, & Mascolo, 2015; Gratzner, Gallenmüller, & Scheitle, 2016). There is no doubt that majority of the internet applications runs on top of TCP or are TCP based, built on a number of session layer protocol (such as hypertext transfer protocol version 1, 2 and 3 HTTP/HTTP/2/HTTP/3 etc.). The ability and robustness of TCP in treating difficult reliability cases and congestion control is the driven factor towards explosive growth of internet (Jimenez Mateo, 2017; Song et al., 2017). The current TCP provides reliable single stream services which hinders the performance of many applications. Lots of effort has been made to address the issue of multipath TCP that can support middle boxes detection and avoidance (as found in MPTCP and TCP-based Concurrent Multipath Transfer (CMT), and their variant). But, it is believe that heterogeneous transmission path may result in packet reordering, in addition to other limitations

such as managing data streams in concurrent transmission and multipath fairness to other traffic that affects multipath transport protocol which will in turn degrade TCP throughput and congest the network immensely (Dolson & Snellman, 2017; Mayoof et al., 2015; Shailendra, Aniruddh, Panigrahi, & Simha, 2017). To overcome the issue of TCP, two transport protocol trend were deployed that operates based on the concept UDP encapsulation (i.e. using UDP as an outer transport protocol). The first one is QUIC deployed in 2013 and drafted in 2015 by Google (Langley et al., 2017), and an open source WebRTC initially release in 2011 and a stable release is produced in 2017 (Edeline et al., 2016; Xiao, 2018). The elements of transport protocol such as congestion control, connection establishment, transmission scheduling and reliability are manage by the inner protocol in both scenario. QUIC is a stand-alone transport protocol unlike its WebRTC counterparts which is SCTP running on Datagram Transport Layer Security (DTLS) (Edeline et al., 2016). Furthermore, firewall rules prevented UDP at some point with few restricted access network, which is commonly found in business oriented establishment.

Other related issues with UDP coexist in the header as found in fast NAT timeouts or throttling (Langley et al., 2017; Wilk, Iyengar, Swett, & Hamilton, 2015). Despite its allurements, VoIP experienced some major challenges in terms voice quality and bandwidth wastage as compared to the conventional PSTN (Abualhaj, et al., 2018).

INTERNET TELEPHONY TRANSPORT PROTOCOL (ITTP).

ITTP unite the functionality of two existing protocol UDP and RTP to convey media data in the transport layer, UDP functionality serves as a data carrier whereas RTP provides the voice media data delivery. ITTP handle VoIP applications with the fundamental data necessary to operate with minimal packet and processing overhead. Any other service(s) that can be added and transported by the protocol itself. The simplicity of the protocol provide VoIP developers with options to augment new service(s) and technique for other use cases. Thus, for worldwide interoperability ITTP is aim at extending its functionality beyond any specific signaling protocol (Abu-Alhaj, et-al, 2012).

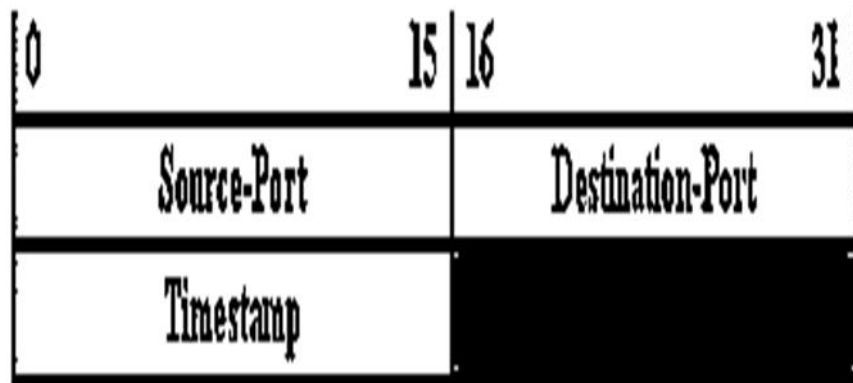


Figure 7: ITTP header format(M. Abu-Alhaj & Manjur, SK, R Sureswaran Wan, T Imad, JM Ahmed, 2012).

Figure 7, presents ITTP header format consisting of three main attributes namely: 2-bytes source port field, 2-byte destination source field and 2-byte timestamp field.

4.1 Source port number: The main purpose of source port number in transport layer is to recognize application at the receiver endpoint, and is also responsible for acknowledgement at the receiver endpoint. Source port is required by ITTP for other additional purposes such network address translator (NAT), as

there is no provision for acknowledgment from the sender side in VoIP applications.

4.2 Destination port: The role of destination port is to identify the delivered data at the receiver side, where individual port number within a given IP device are used to identify particular applications process.

4.3 Timestamp: This is the vital ITTP field that define the amount of time (in milliseconds) spent from the initiation of

the first packet data transmitted per call. Timestamp tends to increase by a value equivalent to the time length of packet payload until it reaches its maximum value and then wraparound from the packet payload predetermine set. Timestamp is at the same time guarantees smooth transmission of packets in real-time.

4.4 Why ITTP: The motivations behind the ITTP transport layer protocol design that can convey VoIP applications are as follows:

- a. **Inability to address VoIP requirement by the existing protocol:** Several transport layer protocol have one specific design and purpose that relies on type of network and applications requirements. As such, a free, reliable and efficient VoIP transport protocol with improved performance is needed.
- b. **RTP/UDP Voice quality degradation:** RTP and UDP can handle all types of real-time media data as such, RTP is attached on top of UDP to deliver VoIP data which in turn complicated the network. Similarly, the added functionality and information resulted

in packet loss which at the same time increases the processing time.

- c. **RTP/UDP bandwidth wastage (Inefficient use of bandwidth in RTP/UDP):** RTP/UDP increases header size of VoIP packet payload as overhead (the ratio of the header size to that of payload size ranges from 67% to about 200%) which degrade internet bandwidth (i.e. VoIP header is between 10-20 bytes then plus 12 byte RTP and 8 bytes UDP: VoIP+RTP+UDP = 30+12+8 = 50 bytes). In addition to 40 bytes IP header to the overall overhead.

ITTP MODEL EQUATIONS FOR PERFORMANCE ANALYSIS.

The model equations can be adapted in performance analysis of IAX/UDP, and RTP/UDP in terms of bandwidth, delay and throughput which is aim of this work in the near future. The recent development in VoIP applications increases the overall internet traffic as they constitute a large portion of packets running on top of internet which affects the end to end link performance.

Per-cell Bandwidth (C_b): This is the amount of bandwidth (BW) consumed per VoIP call, and is a function of size of frame, and size of packet header.

$$C_b = P_k \times P \quad (1)$$

Where: P_k = Packets size in bytes, and
 P = Number of packets per second.

$$P_k = P_h + F \quad (2)$$

Where: P_h = Packets header size in byte, and
 $F = (F/8)$ = Frame size in bytes.

$$P = \frac{C_b}{F/8} \quad (3)$$

Where: C_b = Code bit rate in seconds.

With regard to ITTP this can be reconfigured as follows:

$$I_1 C_b = P_k \times P \quad (4)$$

By substituting (2) and (3) in (4) we have:

$$I_1 C_b = (P_h + F) \times \left(\frac{C_b}{F/8} \right) \quad (5)$$

ITTP in kilobits per second (Kbps) is then given as:

$$I_1 C_b = \frac{[(P_h + F) \times \left(\frac{C_b}{F/8} \right)] \times 8}{1} \quad (6)$$

Similarly, the ITTP packets header is given as:

$$P_h = I_h + I_{I_h} \quad (7)$$

Where: I_h = ITTP protocol header, and

I_{I_h} = IP header.

Substituting (7) in (6) $I_1 C_b$ becomes:

$$I_1 C_b = \frac{[(I_h + I_{I_h} + F) \times \left(\frac{C_b}{F/8} \right)] \times 8}{1} \quad (8)$$



The aforementioned model equations provides a guide towards performance analysis of ITTP in terms of bandwidth, and latency.

The Goodput in (Kbps) is also given by the relation for further analysis as follows:

$$G = \frac{\sum(R_p \times F_k \times 8)}{1} \quad (9)$$

Where: R_p = Received packets in bytes.

FUTURE WORK

The future is aim at adopting an overlay ITTP protocol and optimized the design using one of the heuristic algorithm for 5G mobile services (i.e. HetNet with multipath characteristics). That can also provide reliable, secured and efficient connection at high data rate with the intention to support available use cases in line with Request for Comment (RFC) and Internet Engineering Task Force (IETF) standards. Secondly, comparative analysis will be carried out with respect to ITTP using two other protocol as follows: (1) InterAsterik exchange protocol/UDP (IAX/UDP), and (2) Real time transport protocol/UDP (RTP/UDP). The Key Performance Index (KPI) to be considered is going to be based on Bandwidth, delay and throughput, and validation of the results is expected to be carried out through one of the available network simulation software's (Network Simulator 2 and 3 (NS2/3)).

CONCLUSION

The proposed paper targeted multipath behavior in HetNet for 5G applications using ITTP protocol. The protocol is a standalone protocol that can support other session layer protocol to deliver their data. 5G technological features, applications and challenges with regard to real world has also been discussed. The paper try to look into challenges with existing protocol and how it affected VoIP services and proposed a profound solution that can be used in the feature, with ability to support other protocol to convey their message. Lastly, a future work towards real implementation of this work is also presented.

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