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A Research of VoIP Jitter on Packet Loss in GSM Voice over IP Systems

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Abstract: Jitter in the next generation network Voice over Internet Protocol Systems is a fundamental network problem in call quality measurement research. We classify Jitter according to call quality requirement in overall quality of service of a reliable information and communication technology infrastructure especially as convergence of voice, video and data increases in Internet telephony systems. Internet telephony mode of communication systems is delivered via Voice over IP media and signaling techniques adopted for internet transmission in achieving VOIP system. Our goal in this paper is to investigate the impact of VoIP Jitter by studying the optimal packet Call Flow Routing (CFR) model in Real Voice Optimization to reduce Jitter in VoIP systems. The results of the simulations shows that as the time in arrival of consecutive voice packets increase, the optimal packet Call Flow Programmatic Routing (CFR) reduces with time varying demands in the total packetized call transmission across all network links. We observed that the set of call Flow Routing constraints investigated imposes the packet flow load balance at end-to-end gateway router and shows a reduction in packet loss and at a point, records no loss. The value at which the time slots for the network caller party demand increases, more Caller party serves the subsets of packet flows by simulating the behavior of the network per each slot independently of each slots. Hence, the caller router and called party router flows are positive by resolving the individual routing problem per each slot and integrating the models.

Keywords: Jitter, VoIP, Voice, Optimize CFR, Packet Flow,

I. INTRODUCTION

There is a recent strong trend towards a "IP in everything and everything over IP "world. This lure of a common platform for digitized voice communications and the expected costs savings are proving to be very strong motivators for the proliferation of GSM Voice over Internet Protocol technology. Also the volume of voice traffic on all-IP networks continues to increase at 80% rate in 2017 with the use of Skype, facetime, Viber, Google voice and other voice applications. For example, VoIP services alone account for about 70% of increased volume of local calls to the internet with IP Telephony in 2017 and an estimation of 90% by 2030. Around 40 percent of all local lines are used primarily for internet access while international traffic growth is around 70% as explained in (Stylianos Karapantazis, Fotini-Niovi Pavlidou, 2009).

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The opportunities to deliver enhanced next generation services with smart phone which combine voice, data and video have contributed to the paradigm shift to GSM VoIP technology. VoIP GSM technology combine GSM technology and VoIP networks. Connecting the VoIP GSM gateways requires a separate line of the Internet, at a rate of no less than 32 Kb/s for a single channel. GSM has assisted in the lead to convergence between the internet, fixed and mobile services with the Second Generation cellular based telecommunication system based on Global system for mobile (GSM) standards that delivered data rate at 9.6kbps but with higher delay and jitter. However, Lack of synchronization in comparison to TDM (Time-division Multiplexing) brings concerns about variable conditions on network which causes packet loss and fluctuations in jitter. Jitter causes excess packet loss on receiving VoIP buffers depending on the buffer size and delay variance. (Claude-Joachim Hamann, Steffen Zschaler, 2015)

Based on practical experience of voice perception and human conversational model, partial delays occur at different stages of communication path. The delay distribution on IP networks can be successfully modeled and described by optimal packet Flow Routing (PFR) model which give better results compared to stochastic process.

The goal of the research is to investigate to what value the jitter metric and packet delay variation can reduce Jitter in GSM VoIP system. We measure jitter and packet loss using various subjective and objective constraints such as optimal packet Flow Routing (PFR) model in real voice optimization adapted from the mixed integer linear programming (MILP) model. In achieving this aim, our objectives include: 1. Designing two scenarios using OMNET++ modeler; 2. simulating the scenarios to assess their performance on packet loss and 3. Analyzing the simulated results

II. LITERATURE REVIEW

2.1. Background

End-to-End delay is defined by three characteristics: network delay, encoding and decoding delays, and compression and decompression delays. Varying any of these characteristics changes the distributive property of the packet flow and as such can be used to represent digitized packet flow. When arrival time of consecutive packets is altered from end to end of the network, it is termed network delay. Changes and modification that occur in the packet flow header while the packets arrive to the buffers in limited size with some packets being lost or arriving out of order over a period of time is called encapsulation or encoding.

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The variation in time from de-encapsulation of the protocol header from data and the packet header is called decoding delays. Compression is the technique of reducing the size (bytes or kilobytes). Of some packets prior to arrival at the buffers.

Since the buffers can accommodate limited size of packet flow modifications like 10kbyte per slot. This technique is carried out to avoid packet flow loss or arrival out of order. The 'Packet end-to-end delay' is measured by calculating the delay in the arrival of packetized voice from the speaker or caller to the receiver or callee including the compression and decompression delays along the media path and signal path.

According to (Mahdi H & Suhail A. molvi, 2014), the total voice packet delay is calculated as $D_{e2e.}$

$$D_{e2e.} = D_n + D_e + D_d + D_c + D_{de} \quad (1)$$

Where D_n, D_e, D_d, D_c, and D_{de} represent the network, encoding, decoding, compression and decompression end-to-end delay respectively. The packet delay variation is a quality of service parametric requirements that can be measured using various subjective and objective measures such as packet delay variation (PDV). Packet delay variation (PDV) is a pertinent factor in network performance degradation. The packet delay variation (PDV) is the variance of the packet delay calculated as follows:

$$\text{PDV} = \left\{\sum_{i-1}^{N} ([t'(n-t(n)]-\cup)2\right\}/ n\} \quad (2)$$

Where; $\cup$ is the average delay of n selected packet flow.

The higher the packet delay variation, the higher the tendency of backhaul congestion on the network causing more network overheads and equipment suspension in GSM VoIP and performance degradation caused by jitter.

In most VoIP communication systems, end-to-end packet delay is random variables because voice frame packet arrivals and departures are stochastic processes.

End-to-end packet delay Denoted by En the time instant of the nth packet arrival (the last bit or the most significant bit of the packet has been received) at a network and denote by Vn the time instant of the nth packet departure.

An end-to-end delay process $\{Dn,\ n \geq 1\}$ is an ordered sequence of packet end-to-end delays from the 1st packet; and it is a stochastic process. Consider a realisation

$\{d1, d2, d3, \cdot\cdot\cdot, dN\}$ of an end-to-end delay process with N samples.

In large scale VoIP systems where clusters and nodes are distributed, for example, modeling the end-to-end delays of the first 100 packets from Internet VoIP traffic, basic probability concepts are used to alleviate these experiences. The Probability Density Function of the random variable $D\infty$, denoted by $fD\infty(x)$ is the function that gives the likelihood of $D\infty$ taking the value x, for any real number x:

$$fD_\infty(x) = \frac{d}{dx} \Pr\{D\infty \leq x\} \quad (3)$$

in use cases that the random variable is greater than a real number x, the Complementary Cumulative Density Function (CCDF) of a random variable $D\infty$ is the function that gives the probability of $D\infty$ being greater than a real number x. By definition, its value can be computed as the integral of the PDF from x to ∞:

$$\Pr\{D_\infty > x\} = Z \infty\, xf\int_{x}^{\infty} D\infty(t)dt \quad (4)$$

In GSM Voice over IP Systems, Let $\{d1, d2, d3, \cdot\cdot\cdot, dN\}$ be a realization

of an end-to-end delay process with N samples. The empirical CCDF of end-to-end delay is a function of x, which equals the decimal fraction of the observations that are greater than x:

$$P\hat{}r\{D_\infty > x\} = \frac{\text{number of elements in the realisation} \leq x}{N}$$

$$P\hat{}r\{D_\infty > x\} = \frac{1}{N}\sum_{I=1}^{N} 1\{di \leq x\}$$

where $1\{di \leq x\}$ is the indicator of event $\{di \leq x\}$, defined as

$$1\{di \leq x\} = \begin{cases} 1 \text{ if } di \leq x \\ 0 \text{ if } di > x \end{cases}$$

Let us denote X (t) a two-dimensional stochastic process with set of states (M)

$$M = \left\{ \begin{array}{llll} & j = 0;..; & R; & m = 0 \\ (js) & j = L;..; & R; & m = 1 \\ & j = H+1 & R-1 & m = 2 \end{array} \right\}$$

and its subsets $M_i = \{(j; m) \epsilon M | m = i\}$, i = 0; 1; 2,

where j is the number of GSM VoIP customers in the system and m indicates system operating mode for a is a 2-dimentional signaling scheme.

Take the service completion epochs to be 0 < t1 <

t2 <: : : ; where tn is the instant of the nth customer departure. Then the discrete-time process embedded at customer departure epochs X(tn+0) emerges a Markov chain with set of states

$$M = \left\{ \begin{array}{llll} & j = A = \pi r^2\ 0;..; & R-2; & m = 0 \\ (js) & j = L;..; & R-2; & m = 1 \\ & j = H+1 & R-1 & m = 2 \end{array} \right\}$$

$Mi = \{(j; s)\ \epsilon M|m = i\}i = 0; 1; 2:.$

Similarly, (Mahdi H. & Suhail A. Molvi, 2014), argue that jitter buffers in VoIP systems can both over-fill and under-fill, thereby packets are discarded. This is similar to a stochastic process and modeling in (M.Vonznak, A.Kovac and M.Halas, (2012). This is to receive the stationary queue length distribution of the corresponding stochastic process from the stationary queue length distribution of the embedded Markov chain.

A hybrid mechanism that is flexible with protocol modularity that operates semantically with the aid of session initiation protocol (SIP) is the VoIP systems.

VoIP uses a SIP, media gateways for the end-to-end media transfer.

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VoIP is a 2-dimentional signaling scheme in which the media transfer path and signal path operate independently in two sequences consisting of invitation SIP and acknowledged SIP. There is a pair of media real time protocol (RTP) that carries voice frames and normally a GSM FR codec signaling.

All these are used in translating content (e.g audio) unto media signal and between the formats that are supported by the networks (Angeles D.Keromytis, (2012).

2.2. Jitter

Jitter is the variation in arrival time of consecutive packets. Jitter is calculated by computing the difference in delay of packets over a period of time (Un-Ku Moon, Karti Mayaram, and John T. Stonick, (2002).

Jitter, σD is the value that represents the average deviation of delays from the average delay: σD

$$\sigma D = \sqrt{\frac{1}{N-1}\frac{1}{N}\sum_{I=1}^{N}\{di - \mu D\}}^{\;2} \qquad (1)$$

Under time-related network parameters we understand packet transmission delay, mean inter arrival time difference – jitter and secondarily packet loss, which can be understood as infinite delay of packet delivery (Claude-Joachim Hamann, Steffen Zschaler , (2005). According to (H. Toral-Cruz, A. Khan Pathan and J. C Pacheco, 2012), the packet inter-arrival time (IAT) on the receiver side of the gateway router is not constant even if the packet inter-departure time (IDT) on the speaker transmitting side is constant. We represent the jitter measurement between the sending packets and the receiving packets mathematically. Let S_k be the RTP timestamp and R_k be the arrival time in RTP timestamp units for packets K. Then for two packets K and K-1, the OWD difference between two successive packets K and K-1 is given by equation (2)

$$J(K)\text{-} (R_k - S_k) – (R_{k-1} - S_{k-1}) = (R_k - R_{k-1}) (S_k - S_{k-1}) =$$

$$IAT(K) – IDT(K)$$

$$IAK (K) = J(K)+ IDT(K) \qquad (2)$$

Where IDt (K) = (S_k - S_{k-1}) is the inter-departure time and IAT(K) =(R_k - R_{k-1}) is the inter-arrival time or arrival jitter for packets K and K-1 which is referred to as Jitter.

In this paper, our inter-departure time IDT is 5Ms, 10ms, 20ms and 40ms.

If packet K-1 is lost, equation 3 shows the relationship between the jitter and packet loss .

$$IAT(K) = J(K) + (2)*IDT(K) \qquad (3)$$

If n consecutive packets are lost,

$$IAT(K) = J(K) + (n+1)*IDT(K) \qquad (4)$$

Equation 4 describes the VoIP jitter. In VoIP, real-time transmission could be either by transport control protocol (TCP) or user datagram protocol (UDP).UDP experiences packet loss problems.

In this work we study the 2-state and 4 state Markov chains to observe the packet loss behavior. (M.Vonznak, A.Kovac and M.Halas, (2012)

Let S = S_1......S… S_M be the states of an m-state Markov chain and let Pn be the probability of the chain to pass from state S_1 to state S_j. The transition of states can be represented in the matrix P.

$$P = \{ \begin{matrix} S1 & S2; \ldots & Sm \\ S1 & S2; \ddots & Sm\} \\ S1 & S2 \ldots & Sm \end{matrix}$$

Where S1+S2+………..+Sm = 1

In a four steady-state probabilities of the chain,

S1and S3 represent packet lost, S2and S4 packets are found while six parameters (P21, P12, P43, P34, P23, P32 ϵ (0, 1). The four steady states probability of this chain are as follows:

$$S1 = \frac{1}{1+\frac{P12}{P21}+\frac{P12P23}{P21P32}+\frac{P12P23P34}{P21P32P43}}$$

$$S2 = \frac{1}{1+\frac{P21}{P12}+\frac{P23}{P32}+\frac{P23P34}{P32P43}}$$

$$S3 = \frac{1}{1+\frac{P34}{P43}+\frac{P32}{P23}+\frac{P21P32}{P12P23}}$$

$$S4 = \frac{1}{1+\frac{P43}{P34}+\frac{P32P43}{P23P34}+\frac{P21P32P43}{P12P23P34}}$$

Where the probability of the chain to be either S1 or S3 is when r = S1+S3

2.3. Related Work

Voice over Internet protocol (VoIP) is an Internet Telephony technology that is initiated on the Internet with VoIP applications such as Skype calls or Google Voice. The calls are initiated on the Internet usually end at a point that is not on the Internet. Most of the recipients of telephone calls receive the calls with a landline phone or with a smart phone. The route may begin on the Internet but will end at a point that was reached by routing the call from the public switched telephone network (PSTN). We study the GSM VoIP technology from the perspective of modeling the GSM technology. As explained in (O. Ali Abdullah and A. M. Jassim Al-Hindawi, 2014) GSM/EDGE system analysis is based on channel coding, modulation type, multipath channel effect, channel estimation. The results reveal a decrease in signal to Noise ratio (SNR) in the presence of fading channel with SNR values of 40dB and 60dB when the system is operating in Additive White Gaussian Noise (AWGN) channel and Rayleigh fading channel. Their result informs us of the impact of phase noise on symbol error rate in GSM networks. (B. Xi, H.Chen) shows that the validation of models for multiplexed process for VoIP traffic is based on the empirical model and mathematical model.

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The idea of using two models to compare network traffic is a closer work to our investigation in this paper. The first model utilized VoIP traffic generated from a service provider and the analysis of the second model is a parametric statistical model of the IP-inbound traffic to an IP network.

The strength and weaknesses of the two models were compared but the accuracy of models could not check the Quality of Service.

Our work proposes to bridge this gap by using mathematical models like stochastic process and optimal packet Flow Routing (PFR) model for queuing simulations in test runs and compare the delay and jitter results.

In (H. Toral-Cruz, A. Khan Pathan and J. C Pacheco, 2012) the accuracy of the VoIP traffic Qos model using markov models was analyzed. The analysis explains the jitter and packet loss behavior of VoIP traffic which is similar to the self similar process in Hurst parameter. According to Hurst parameter, the time series of VoIP traffic , the continuous self-similarity with the exponent 0<H<1 is a real-valued continuous time series {X(t), t∈ ß} for any a > 0, the finite dimensional distributions of {X(at),t∈ ß are identical to the finite dimensional distributions {a^HX (t), t∈ ß} i.e {X(at), t∈ ß= {a^H X (t), t∈ ß where d denotes equality in distribution. According to Hurst parameter, the discrete self-similarity

X_t = (X_t ; t∈ N) denotes a discrete time series with mean μ_x, Variance $2x$, autocorrelation function r(k) and auto covariance function (ACV) γ(k), k>0; where X_1 can be interpreted as the jitter , at time instance t.

When considering discrete time series , in the context of jitter, the definition of self-similarity is given in terms of the aggregated process as follows:

$$X^{(m)} = (X^{(m)} ; k \in N) \qquad (1)$$

Where m represents the aggregation level and $X_k^{(m)}$ is obtained by average original series X_1 over non-overlapping blocks of size m, and each term $X_k^{(m)}$ is given by:

$$X_k^{(m)} = \frac{1}{m\, i=} \sum_{i(k-i)m+1}^{km} \quad K = 1,2,3 \ldots \ldots \qquad (2)$$

Then it is said that X_k is self-similar (H-ss) with self-similarity parameter (0<H<1) If:

$$X_k^{(m)d} = m^{H-1} X_1 \qquad (3)$$

As explained in (U. Peter Daniel, N. chikazo Agbanusi & K. Danjuma, 2014) their work is towards Optimization techniques to VoIP application and services presents better jitter result rather than modeling the media protocols. They utilized the technique of compression with aggregations, reductions and silence suppression of the voice packet header to improve bandwidth utilization and the quality of service in VoIP networks. The result of the delta-multiplexing technique using the header and payload reduction scheme produced codec relativities.

Our work bridged the identified gap of codec relativities. We investigate the jitter values and propose the SDN-controller architecture in gateway routers to yield faster bit rates, reduce delays for the compressed voice and packet header.

In evaluating the SIP signaling in a SIP based VoIP network in this paper, (G.Vennila & MSK Marikandan, 2016), a scalable detection technique for real-time transport protocol(RTP) flooding attack in VoIP network, the packet inspection technique using statistical methodology proposed in this paper was to detect RTP flooding attacks in VoIP network. The RTP succeptibility to flooding attacks degrades QoS in servers. An innovative approach to design in this paper is a proof that detection rate of the algorithim has over 98% helligner distance. The helligner distatnce propose PIS technique with higher true positive rate over false positive percentatge.

(Colin Perkins, O. Hodson and V. Hardman, 1998) explore the need for packet loss recovery techniques and identified in IP multicast channel by studying the delay characteristics. Their work focuses on transport in video conference systems. The use of simple repair scheme such as repetition with fading gives good quality in such systems.

In (I. Oghogho, D. Odikayor, A-Alli Adebayi and S. Wara). VoIP replaced the plain old telephone systems (POTS) or leaseline just like Internet technology is already competing with GSM technology. However, the growth in VoIP is with challenges such as poor quality of services and reliability, cross border issues, geo-location deficiency etc. Our work investigates the jitter and end-to-end delay as part of the proof of reliability in VoIP Quality of service.

Similarly, (Miroslav Voznak, A. Kovac and M.Halas,) objective methodology research reduced the effect of packet loss caused by jitter on receiving buffers in VoIP. The proposed methods of numerical approximation of general jitter buffer behavior using a modified ITU-T E-model considered the jitter characteristics over 16 packet samples for CODEC with 20ms audio per packet. The proposed modifications gave more accurate Mean opinion score (MOS) with real-time network jitter conditions. The transient response of the real-time network jitter conditions generated by the real-time call flow routing model in this paper is similar to the real-time E-model with fast recovery rate. The analysis of the jitter and packet loss behavior of voice over internet protocol traffic with Hurst parameter (H) investigated the packet loss rate (PLR) for voice traffic measurements. The power law function modeled the relationship between Hurst parameter (H) and packet loss rates (PLR) characterize three parameters. One of the parameter investigated is the focus of our work. In this paper, we utilized synthetic jitter traces generated from VoIP jitter to test, research and evaluate the performance analysis of jitter buffer of VoIP system. The analysis and research method in this paper is similar to the concept in

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(Mahdi H. Miraz, M.Ali, M.A Gannie and A.H Hussien, 2014) but the result of their research work is on multiplatform and integrated systems like WIFI-UMTS networks which is contrary to GSM VoIP which is the focus of our work. It shows that both MOS and packet end-to-end delay were lower because the VoIP jitter was observed to decrease.

In addition, the performance of VoIP services using a varying parameter like compression and decompression (CODEC) scheme in (J.Cao and M. Gregory, 2006) shows that the changes in the number of voice packets and CODEC affects the end-to-end delay and jitter over a UMTS Network.

For example two CODEC; GSM-FR and G.729A gave a result of lower delay of 150ms when 20ms and 30ms frame sizes was used while a combination of the three CODEC; GSM-FR, G.711 and G.729A produced a 158ms delay with voice frame size 4ms and 10ms.

III. RESEARCH METHOD

3.1. CALL FLOW ROUTING MODEL (CFR)

In this section, our methodology is to simulate call measurement for corresponding jitter values by varying time demands and constraints at stationary, uniform and Call flow Load-Balance without sending multiple voice packets down one or multiple

Table 1: Summary of Notation in Optimization Model

	Parameters
R	Set of callee party
S	Set of caller party
G	Set of gateway routers
0	Set of Call flows
T	Set of Time Flows
D	Traffic demand generated by Receiver R for flow f
c	b_{rc} Link Capacity between Gateway G and receivers
b_{pr}	b_{pr} Link Capacity between Speaker S and Gateway G
Bx_{1x2}	B_{x1x2} Link Capacity between Gateway x1 and Gateway x2
a_{of}	a_{of} 0-1 parameter to indicate if speaker s is forwarding flow f
X	Maximum number of flows programmed in each table
Y	Y A large flow
L	Maximum number of VoIP nodes displayed
m	Maximum memory that VoIP node can use for programming.
M	Total memory shared by all VoIP nodes for programming

Given the definitions and assumptions in Table 1 and Table 2, we formulate the optimal call Flow Routing model (CFR) with time varying demands as follows:

This section discusses the proposed Optimization models that we use to study the performance of the SDN and GSM VoIP paradigms. We presents the call flow Programmable model (FR) with time-varying demands. We modify the FR model to represent applicable characteristics of SDN and GSM VoIP, respectively.

A. Flow Programmatic Model with Time-Varying Demands

In this subsection, we explain our proposed call Flow Programmatic model (CFR). This model describes a named data network without caching functionality. In the following subsections we will further extend CFR model with cognizance of the attributes of the SDN and GSM VoIP architectures. As summarized in Table 1, three extensions of the CFR model will be presented below:

1) Call Flow Allocation and Programmatic (CFAP), a model bespoke for SDN that finds the optimal solution on the overall time horizon.

2) FAP- Single Time Slot Heuristic (FAP-TS) bespoke for SDN, which chooses the optimal solution for single time slots;

3) FAP-Single Relocation Time VoIP Heutristic (FAP-TR-VoIP),

We model the network as an undirected graph G (M, N) where M is the set of counters and N the set of tables. The set of counters M is partitioned into three disjoint sub-sets: callee (denoted by R), caller (denoted by S) and open flow GSM Gateway router (denoted by G).

Furthermore, we denote with T the set of time slots, where 0 represents the set of call flows that can b retrieved from the network, also known as index.

Callee gateway routers are counters that express demands for flows in each time slot. We denote with d_{tof} the demand of Callee gateway routers for flows at time. The demand is expressed in voice packet frame size; we assume that traffic demands are inelastic meaning they are fixed per time slots.

In order to meet the Service Level Agreement load, caller party can serve the subsets of flows they own. For each caller and flows, we denote with afp the caller –flow allocation

Where: afp if flow f is available at caller party otherwise:

Table 2: Summary of Notation in Optimization Model Decision Variables

	Decision Variables
x_{to}	variable indicating if Gateway g is programming flow at time t
yto_{r1r2}	Voice Packet flow f from router x1 to the neighbor gateway x2, during time slot t
yto	Voice Packet flow f from caller s x1 to gateway x2, during time slot t

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y_{torc}	btoxc voice Packet flow f related to flow f from gateway x to receiver during time slot t during time slot t
lr	0-1 variable indicating if a GSM VoIP slot is installed in router r
zo;t	0-1 variable indicating if GSM VoIP slot node at route caches voice packet frame o during time slot t

All he GSM gateway routers in the network perform control and forwarding. Considering the carrier GSM gateway-router pairs, we denote the the link capacity with $b_{p,r}$. Similarly, we define the router-Caller party bandwidth while $b_{r1\ r2}$ represents the GSM gateway router-router capacity.

We denote with $X_{p,r}$ the Caller–GSM gateway router time-dependent flow variable . It represents the amount of voice frame that caller party sends to router for flow in time.

We describe the router-caller party flow variable õ with y_{to} Table 1 summarizes the notation used in optimizing the models. This model solves the network traffic forwarding flow routing by a programmatic functionality.

The objective of the model comprises of the following:

1. Network call flow at slot s ($NCFR_s$) comprises of calls at end-to-end gateway router of all component of the total packetized call transmission across all network links

$$TCFR = \sum_{\forall \in R} NNRP_m \quad (1)$$

2. Total voice packet time in arrival of consecutive packets ($TCNRP$)

$$TCNRP = \sum_{s \in N} CNRP_s \quad (2)$$

3. Total packetized call transmission across all network links. It is calculated as follow:

$$JITTER = \sum_{\forall \in 0} Time\ t_s \cdot (CFAP_s + CFR_s) \quad (3)$$

We formulate the optimal call flow routing model (CFR) with time- changing demands of as follows:

The model is defined as follows:

Objective: Minimize $\sum_{\substack{\forall O \in \Theta \\ \forall O \in \Theta T}} (\sum^{o,t}_{\substack{\forall O \in \Theta \\ \forall O \in \Theta}} Yrir2 + \sum^{o,t}_{\substack{\forall O \in \Theta p \\ \forall O \in \Theta R}} Yrir2 + \sum^{o,t}_{\substack{\forall O \in \Theta R \\ \forall O \in \Theta C}} Yrc$ (4)

Subject to $\sum^{o,t}_{\substack{\forall O \in \Theta \\ \forall O \in \Theta}} Yrir2 + \sum^{o,t}_{\substack{\forall O \in \Theta p \\ \forall O \in \Theta R}} Yrir2 + \sum^{o,t}_{\substack{\forall O \in \Theta R \\ \forall O \in \Theta C}} Yrc \sum^{o,t}_{\substack{\forall O \in \Theta R \\ \forall O \in \Theta C}} Yrc$

$$\sum_{j \in N: i \neq j} L^{sd}_{ji} = \begin{cases} L^{sd} & i = s \\ -L^{sd} & i = d \\ 0\ otherwise & \end{cases} \quad \forall\, s, d, i \in$$

Call Flow Routing Constraints

$$\forall (r1, 0, t) \in R \times \theta \times T \quad (5)$$

$$\sum_{\substack{\forall O \in \Theta \\ \forall O \in \Theta t}} Y^{o,t}_{ijr1r2} \leq b_{r1,r2} \quad \forall\, (r1, r2, t) \in R \times R \times T \quad (6)$$

$$\sum_{\substack{\forall O \in \Theta \\ \forall O \in \Theta T}} Y^{o,t}_{ijr1r2} \leq b_{p,r} \ \forall\, (r1, r2, t) \in S \times R \times T \quad (7)$$

$$\sum_{\substack{\forall O \in \Theta \\ \forall O \in \Theta T}} Y^{o,t}_{ijr1r2} \leq b_{r,c} \quad \forall\, (r1, r2, t) \in R \times R \times T$$

(8)

$$\sum_{\substack{\forall O \in \Theta \\ \forall O \in \Theta T}} Y^{o,t}_{ijr1r2} = d_c t, o\ \forall (r1, r2, t) \in R \times 0 \times T$$

(9)

$$Y^{o,t}_{ijr1r2} \leq b_{p,r.apo} \quad \forall\, (r1, r2, t) \in S \times R \times O \times T \quad (10)$$

$$Y^{o,t}_{r1r2} \leq 0 \quad \forall\, (r1, r2, t) \in R \times R \times O \times T \quad (11)$$

$$Y^{o,t}_{pr} \leq 0 \quad \forall\, (r1, r2, t) \in S \times R \times O \times T \quad (12)$$

$$Y^{o,t}_{rc} \leq 0 \quad \forall\, (r1, r2, t) \in R \times R \times O \times T \quad (13)$$

The objective function of (4) minimizes the total packetized call transmission across all network links.

The set of constraints (5) imposes the call flow load balance at end-to-end gateway router. Constraints (6), (7) and (8) is the total link used on router-router, caller party router and called party router links respectively. The set of constraints (9) confirms that the time slots for the network caller party demand. Caller party serves the subsets of call flows they possess and this is expressed by the set of constraints conditions (10). Constraints (11), (12) and (13) impose that router- router, caller router and called party router flows are positive.

The CFR model, the simulated behavior of the network per each slot is independent of each slot. We compute the final objective value by resolving the individual routing problem per each slot and integrating the models.

IV. SIMULATION RESULTS AND DISCUSSION

The first use case utilized two GSM VoIP subnets in Oye-Ekiti Campus and Ikole Ekiti Campus GSM substations. These were configured with a SIP server connected through an IP cloud. The base station configurations are GSM with Oye campus interface display in mobile while Ikole was displayed with GSM workstation. Three work stations are used in this research work.

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Table 3: GSM VoIP subnets in Oye-Ekiti Campus and Ikole Ekiti Campus GSM substations

Subnet Name	Use Case	Base station configuration	Work station Type	Number of work station
Oye-Campus	GSM VoIP	GSM	Mobile	3
Ikole-Campus	GSM VoIP	GSM	GSM workstation	3

The analysis is initiated by running multiple VoIP applications (Skype, Google voice and Viber) in five systems running on Windows and Linux kernel in a closed Campus Wi-Fi LAN environment. The call was established between client and Asterisk server and the traces of the applications were captured in Wireshark for a day, followed by a week and finally for a four month and feature extraction was done with a total 1500 calls.

Out of different statistical features such as payload type, sequence number, timestamp, and SSRC, we concentrated on request message rate, payload and size of the message. Request message rate gives the information about the signaling (INVITE, BYE and ACK) and media packets for a given time. Payload type is a value indicating which codec is used to encode the audio payload and the size of the message indicates the value of message in bits/sec.

Table 4: SIP Based VoIP Applications

SIP Server connect Time Out	Durations Second (ms)	Caller	Callee	Number of Calls in500sec
	5	Node 1	Node 24	115
	10	Node 22	Node 3	60
	20	Node 5	Node 20	30
	40	Node 15	Node 19	15
Voice Codec PCM (G.711.64kbps) GSM (GSM, 13kbps) IPT (G.729 A. 8Kbps)				Total VoIP Call in 500 seconds = 220 calls

Table 5: Call Flow Model (Constraints)

Optimal Call Flow Model (Constraints)	Voice Codec	Total Successful Calls out of 2oo calls	% Successful VoIP Calls	% Rejected VoIP Calls
	PCM	20	10	4
Stationary	GSM	30	20	2
	IP-T	40	35	1
Uniform	PCM	30	20	20
	GSM	40	30	10
	IP-T	50	60	15
Call flow Load-Balance	PCM	40	70	2
	GSM	50	60	3
	IP-T	60	90	2

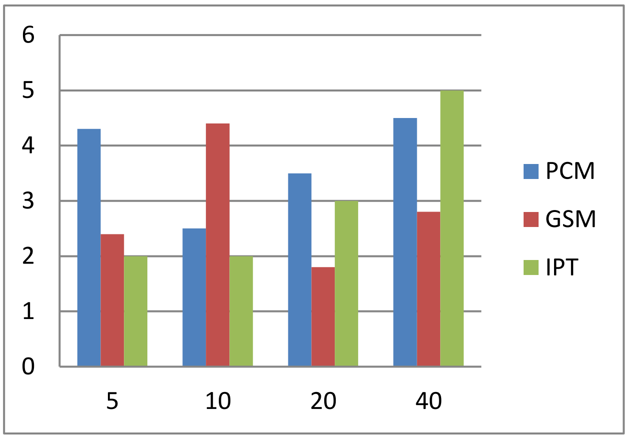

Figure 2: Durations of Voice Codec performance in Second (ms)

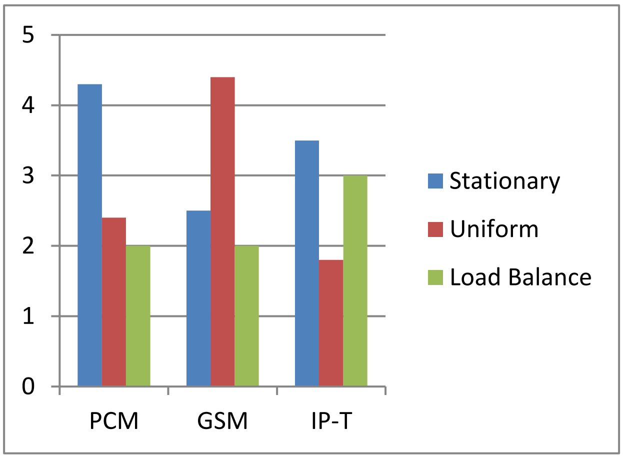

FIGURE 3: Optimal Call Flow Model (Constraints) for the three Voice Codecs.

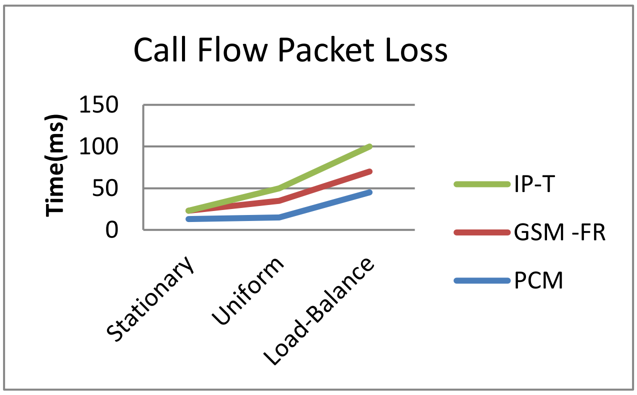

Figure 4: Time in arrival of consecutive voice packets increases and the optimal packet Call Flow Routing (CFR) reduces with time.

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Evaluation: We observed that the set of call Flow Routing constraints investigated imposes the packet flow load balance at end-to-end gateway router and shows a reduction in packet loss and at a point, records no loss. The value at which the time slots for the network caller party demand increases, more Caller party serves the subsets of packet flows by simulating the behavior of the network per each slot independently of each slots. Hence, the caller router and called party router flows are positive in Figure 1.0 by resolving the individual routing problem per each slot and integrating the models. The results of the simulations shows that as the time in arrival of consecutive voice packets increase in figure 4 for IP-T Codec with delay of 150ms under load-balanced constraints, the optimal packet Call Flow Programmatic Routing (CFR) reduces in figure 4 for GSM-Fr and PCM Codec respectively with time varying demands of 40 ms in the total packetized call transmission across all network links.

V.RECOMMENDATION

To achieve higher data on the GSM carriers with reduced jitter, the evolution of 3GPP started from GSM, GPRS, EDGE, UMTS, HSDPA, HSUPA, HSPA, LTE and LTE advanced with the 4G features of release 8 that has reduced delays for both connection establishment and transmission delay. The increased user data throughput, increased cell-edge bit rate, reduced cost per bit implies improved spectral efficiency, simplified network architecture, seamless mobility between different radio access technologies and reasonable power consumption for the mobile devices.

According to (J.Caop,M.Gregory, 2006), the IETF (internet engineering task force) SIP was adopted by 3GPP (Third generation Partnership Project) for the call setup session of VoIP and other IP-based multimedia communication and the current 4th Generation system (Long Term Evolution) with the application of packet switching on radio interface to fully support real-time services such as GSM/EDGE VoIP.

VI. CONCLUSION

This VoIP research impacts on how to mitigate jitter in GSM VoIP systems with the optimal packet Call Flow Routing (CFR) model in Real Voice Optimization. The value at which the time slots for the network caller party demand increases, more Caller party serves the subsets of packet flows. We simulated the behavior of the network per each slot independently of each slot. Hence, the caller router and called party router flows are positive by resolving the individual routing problem per each slot and integrating the models. Eliminating the effect of packet loss by sending multiple voice packets down one or multiple connections of highly complex systems-of –systems has worst performance index and indicate issues in future research work in large-scale VoIP systems. Other direction is the effect of jitter in a 4G LTE advanced WAN technologies environments

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