Modelling and simulation of an analytical approach to handle real-time traffic in VoIP network

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Abstract: In recent years, internet protocol (IP) has become a good choice over public switched telephone network (PSTN). VoIP implementation uses hard IP phones, soft IP phones and uses Softswitch for call signalling. The Erlang B model is used to determine number of trunks in circuit switched network and found traffic intensity and grade of service (GoS). This paper utilises an extension of Erlang B model for traffic engineering VoIP, i.e., extended Erlang B model. The main purpose for extended Erlang B is that it has better efficiency to handle the percentage of blocked calls by choosing a threshold value (∝). We propose a new measurement scheme based on extended Erlang B model using FreeSWITCH and simulated and analysed VoIP traffic. We compare our version with the original definition of EB model and present further results from simulations. The proposed scheme is also analysed for other QoS parameters, i.e., jitter, end to end delay and MOS.

Keywords: voice over IP; VoIP; session initiation protocol; SIP; Softswitch; call admission control; CAC; traffic load measurement; Erlang B.

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1 Introduction

Voice over IP (VoIP) is a technology that transforms and carries the voice communication over internet protocol (IP) network like internet. Instead of using circuit switch network, voice is transmitting in the form of digital packets by using packet switched network (Yu and Al Ajarmeh, 2008). VoIP call includes the call setup signalling protocol, call admission control (CAC), etc. A large number of factors are involved in making a high-quality VoIP call like codec, packetisation, packet loss, delay, delay variation, etc. to provide QoS (Goode, 2002). In VoIP signalling, when session initiation protocol (SIP) servers have insufficient resources to handle all the SIP messages they receive then overload occurs in SIP networks. A scheme has been proposed that defined the behaviour of SIP servers involved in overload control and also specified a loss-based overload scheme for SIP (Gurbani et al., 2014; Sinam et al., 2014). VoIP is encapsulating digital voice in IP packets. Traditional IP networks are not handling voice traffic that may cause packet delay or loses. Although there has been work that focused on software infrastructure that combines the scalability and flexibility benefits of real-time network simulation with the realism of open-source routing protocol implementations (Li et al., 2009) and route selection methods in mobile wireless networks to a given destination based on the probability (Lee et al., 2006) but for voice and data traffic, traffic engineering is an important aspect. Once the session is initiated, real-time protocol (RTP) packets can be exchanged between the initiator and the receiver end. Real-time control protocol (RTCP) is a protocol that controls end to end information about the session of RTP packets. An introduction to telecommunication system engineering that deals with the study of communication via public switched telephone network (PSTN), VoIP and SIP protocols have also been discussed (Freeman, 2015).

VoIP implementation uses hard IP phones, soft IP phones and does not rely on a traditional PBX. Call servers perform IP phone registration and coordinate call signalling. In VoIP, call servers are also known as Softswitch (Shunyi, 2001). Softswitch is used to divide the call control function from media gateway and comprehend the call control and other functions like access control, interpretation, routing-selection, gateway management, call control, wide-band management, signalling, security, call detail records (CDRs), etc. by software. There are different open-source Softswitches available like FreeSWITCH, YATE, Asterisk, Kamilio, etc. Earlier work on VoIP traffic has been done using Asterisk server, Ethernet Switch, Alpine Linux, Kamailio server (Imran and Qadeer, 2009; Imran et al., 2009) but nobody has worked for capacity evaluation and analysis of VoIP traffic on FreeSWITCH as Softswitch.

Erlang B (EB) model is used for classic telecommunication networks. Authors analysed the potential applicability to IP network and introduced a procedure for determining proper values of input variables for EB model based on the characteristics of network link and codec in use. They proved the model applicability with simulation results by using NS2 software (Mišuth and Baroňák, 2011). We also found that most of the work has determined number of trunks by using EB model (Cisco 'Voice Design and Implementation Guide'), but we have proposed a scheme for capacity evaluation in VoIP based on extended Erlang B (EEB) model for traffic engineering. EEB is used for determining number of channels based on grade of service (GoS) and traffic intensity (I). Our proposed scheme also provides recall system for the percentage of blocked calls by choosing threshold value ($\propto$) that improves the efficiency of attempt calls. Our simulation for VoIP traffic engineering is based on QualNet simulator that simulates end to end conversations. The initiator and the receiver can generate real-time traffic that simulates real life conversation. For VoIP traffic, QualNet uses SIP and H.323 protocols but we have considered SIP protocol as it is better protocol for handling HTTP traffic. We also analysed the output of our proposed scheme by considering different codecs like G.711, G.729A and G.723.1 that measure how many packets can be sent on network for processing.

The rest of this paper is organised as follows. Section 2 presents literature review. Section 3 focuses on traffic engineering and also includes introduction to SIP, CAC, FreeSWITCH and traffic load measurement. Proposed technique is presented in Section 4 and simulation and results are analysed in Section 5. Section 6 concludes the paper.

2 Literature review

This section is focusing on work done in literature on traffic engineering, capacity evaluation, etc. Yu and Al Ajarmeh (2008) used EB model that uses traffic intensity and GoS for finding the number of channels. They conducted experiments to determine the maximum load based on various voices codec schemes. Their results show that call load is more confined by network devices as compared to physical connections. Hole and Tobagi (2004) give an analytical upper bound on the ability for VoIP application in IEEE 802.11b networks, examining a wide range of models that include delay, jitter, and channel conditions. Garg and Kappes (2003) experimentally find the ability of VoIP traffic with fixed packetising interval 10 ms and results of VoIP traffic on UDP data traffic in 802.11b. They determine that VoIP traffic has capacity of six and the bandwidth available is decreases by increasing VoIP connections. Shin et al. (2007) also determined the ability of VoIP traffic in 802.11b test-bed and gave capacity estimation for 64 Kb/s CBR VoIP traffic for 15 calls with packetising interval 20 ms and 34 to 36 calls for VBR VoIP traffic with activity ratio 0.39.

Thompson et al. (2013) presented the design and implementation of a prototype for an IP-based telecommunication system (IPTS) that give many important features for telecommunication with the support for text, data as well as video. Ansari et al. (2013) implemented an interactive voice response system (IVRS) for SIP-based phones. They implement this system using FreeSWITCH as a server and the SIP protocol. Ganiga et al. (2012) used QualNet 4.5.1 simulation tool for measuring the VoIP and CBR traffic and comparing the performance of different routing algorithm over the WiMAX network. Fauzia et al. (2008) defined and classified a process for free VoIP flows in a wireless network. Their main focus was to impose charges to obtain services or block these calls while entering the network. Tariq et al. (2013) examine the action of various VoIP codec schemas with or without effect of RTP jitter buffer. They found under no RTP jitter buffer, with higher voice quality, the VoIP flows have better voice quality as compared with the RTP jitter buffer. Costa et al. (2015) also found the suitability of the Asterisk PBX server that gives capability of VoIP communication with an average mean opinion score (MOS). Their experiment shows that Asterisk server can handle not more than 160 calls with a blocking probability of less than 5% and has an average MOS around 4.

From the literature review, as per our knowledge, nobody has considered EEB model for traffic engineering. EEB has better efficiency to handle percentage of blocked calls. In this paper, we have proposed a scheme based on EEB model for capacity evaluation using FreeSWITCH. Simulation of VoIP traffic is done in QualNet 7.3 network simulator and the scheme is analysed for different traffic measurement parameters.

3 Traffic engineering

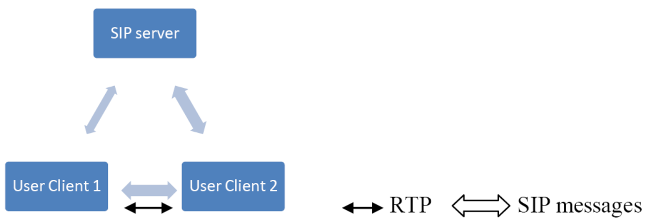
Traffic engineering is a technique for optimising the performance of telecommunication by controlling and analysing the behaviour of data transmitted over that network. In this section, we have presented an overview of SIP, CAC, FreeSWITCH.

3.1 Session initiation protocol

SIP is an application layer protocol that is used to initiate, establish, modify and terminate the session. It is one of the most popular protocols that are designed for establishing the VoIP calls. It combines with other IETF protocols like RTP, RTCP, and gateway control protocol (GCP) to establish the entire multimedia architecture. It is very similar to HTTP, the web protocol, or SMTP. There is no format defined for SIP message, so many kinds of information may be conveyed via SIP. SIP is responsible for finding the location on the basis of the URL of the end point to be used and can be done with the help of DNS server (Rosenberg et al., 2002). SIP-based architectural model was proposed by Rasol et al. (2016) which can be deployed in service provider data centres for maintaining service availability, scalability, and security.

User agent client (UAC) sends requests to SIP server and SIP server sends back responses. The SIP server act as a proxy server and sends requests to some another server on the favour of a client. The SIP server also acts as a registrar and it also checks that UAC is authorised or not as shown in Figure 1.

Figure 1 SIP flow (see online version for colours)



SIP uses basically six types of messages:

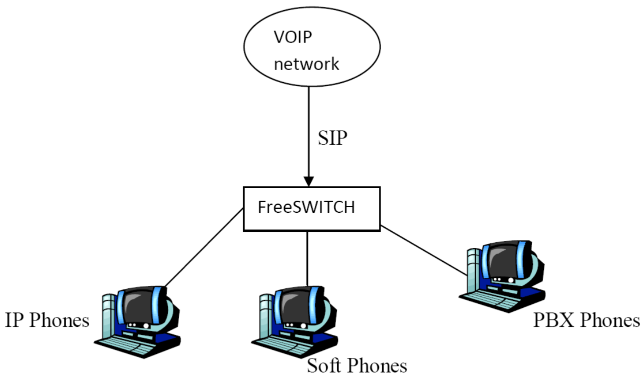
- INVITE – When a UAC wants to establish a call then request is sent.

- ACK – used to acknowledge that client has received the last response to an INVITE request.
- REGISTER – use to register the UAC.
- CANCEL – used to cancel the INVITE request.
- OPTION – used to question the ability of the other user agent.
- BYE – it can be sent by either the caller or the callee to terminate the call.

In SIP, for sending voice traffic, firstly, UAC1 (sender) sends an INVITE request to a server for initiating a session and then server sends 100 Trying responses immediately to UAC1 to stop retransmission of INVITE message. The server searches the address of UAC2 (receiver) in location server. After, it further sends the INVITE request. Thereafter, UAC2 generates 180 ringing and sends back to UAC1. A 200 OK response is generated when UAC2 picks up the phone call. UAC2 receive an ACK from UAC1, once it gets 200 OK, the session gets established and RTP packets can be sent. Any UAC can send BYE request for terminating the session.

CAC is used to determine if network has adequate resource to convey an incoming call. CAC is used for preventing over subscription of VoIP network by queuing, traffic shaping, buffering, and packet dropping. CAC uses resource-based control (RBC) and measurement-based control to reject a VoIP call request if network could not guarantee voice quality. In RBC, resources are vital for VoIP traffic in network bandwidth (Erlang and VoIP bandwidth calculator). The CAC approach that is being used in this paper is RBC.

Figure 2 FreeSWITCH using VoIP network (see online version for colours)



3.2 *Softswitch*

Softswitch is a central device in a telecommunication network that connects calls from one phone line to another with the help of software running on a computer system. In this paper, we have used FreeSWITCH (FreeSwitch, 2013). FreeSWITCH is an open source IP-based platform (Minessale et al., 2016). Its design is based on the central core which is stable and has modules for specific functionality. Interfaces to these modules are provided by core that allows the developer to manage the system and add their own module, if needed. Modules that operate on the central core are independent of each other. Each module can generate events that enter into the core. The most important modules are the endpoint modules that give interfaces to communication protocols. Session is initiated by the connection between FreeSWITCH and a SIP protocol.

4 Proposed technique

In this section, we propose a technique using EEB model for capacity calculation for designing VoIP network.

4.1 *Traffic load measurement*

During voice and data communications, links transmit the calls to servers then calls are sent to group of servers. If any server is available, then call is handled. When all servers are busy, the caller can:

- Receive a busy signal and ask the caller to try later.
- Automatically route the call to another server that is free.
- Wait in a queue until a server is available.
- Wait for some sufficient time interval, then disconnect if no server is available.

When all servers are busy, the disposition of call has great influence on mathematical model for traffic engineering that can be used. The formulas explained below cover those cases in which sources are either blocked or queued, when all servers are busy. For both cases, probability of all servers being busy is calculated. Total probability (P) is 1. The probability of all servers that are served is 1 – P of all servers that are busy. For example, if the probability of all servers that are busy is 10%, then the probability of servers that served = 1 – 0.01 = 0.09, which is equal to 90%. This means 90% of the servers are served and 10% of the servers are busy during connection. The next section gives details of proposed scheme.

4.2 *Proposed algorithm*

EB is used for determining number of trunks. If all trunks are busy then calls forward to next group and never returns (Angus, 2001). In EB, we have to calculate the traffic intensity and GoS as shown in equations (1) and (2) (Parkinson, 2002; Chen, 2009).

$$\begin{aligned} &\textit{Traffic intensity } (I) = \textit{Call rate} \\ &*\textit{Average holding time } (h) \end{aligned} \tag{1}$$

$$GoS = \frac{\frac{I^N}{N!}}{\sum_{X=0}^{N} \frac{I^X}{X!}} \tag{2}$$

where I is traffic intensity and N is number of servers. *GoS* is the probability of incoming call that is being blocked. It is a critical factor for calculating the required number of trunks. If *GoS* is 0, that means no call is blocked, i.e., 100% non-blocking.

EEB is the enhancement of EB in which caller recalls after receiving the busy signal. In EEB, for calculating the probability of blocking calls, we need the offered calls, number of trunks, and percentage of calls being blocked that will be attempted again by choosing a threshold value between 0% to 100% (Erlang and VoIP Bandwidth Calculator, 2014). For simulation purpose, we have considered the threshold value, ∝, as 50% so that load can be balanced on server. The complete working of proposed scheme is shown in form of steps as shown below.

Algorithm: Computes the maximum calls received and processed using EEB

Input: n = number of call attempts from client

B = total bandwidth available

Output: Maxcalls = maximum call received and processed

1) Calculate number of channels using EEB
   - i. Calculate the GoS

$$GoS = \frac{\frac{I^N}{N!}}{\sum_{X=0}^{N} \frac{I^X}{X!}}$$

   - ii. Blocked Erlangs (BE) = Number of trunks (N)*GoS.
   - iii. Overflow Traffic (O) = BE * Recall factor (R).
   - iv. Carried Traffic (CT) = ($N - BE$) + O.
   - v. Calculate sum of Carried Traffic and traffic that never returns, i.e., $CT + O$.
   - vi. Calculate sum of Original traffic and Recall Traffic, i.e., $N + R$.
2) Calculate bandwidth
   - i. VoIP encapsulates digitised voice in IP packets.
   - ii. In G.711 Codec, digitised voice packet is of 160 bytes (64,000 bps digitised voice channel * 20 ms sampling interval).
   - iii. Total packet size (includes Layer-2 header, IP header, UDP header, RTP, 160 bytes) comes out to be 238 bytes.
   - iv. Bandwidth = 95,200 bps ((238*8 bits)/every 20ms interval).
3) If B is available for n calls

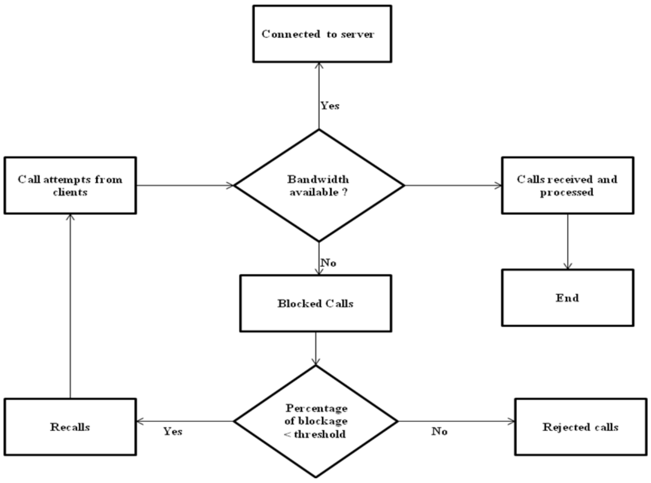
   then Call connected to server

   else Block Call
4) If percentage of blocked calls < threshold (α)

   then Recalls and go to step 3.

   else Reject
5) End

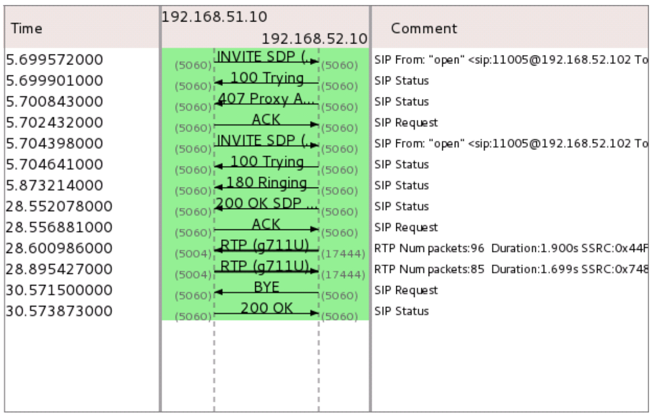
Figure 3 EEB flow chart



5 Simulation and results analysis

For simulation study, we have used QualNet 7.3. Some researchers have proposed real-time discrete event simulator (Wainer, 2016) but QualNet has modular and layered stack design. It has standard APIs for composition of protocols across different layers and has built-in measurements on each layer. QualNet contains GUI tools for system/protocol modelling (http://www.web.scalable-networks.com/content/qualnet).

Figure 4 Wireshark traces (see online version for colours)



Simulation is designed by varying the number of VoIP flows and we have captured packets on network using Wireshark and traces of same are shown in Figure 4.

5.1 *VoIP parameters*

For analysing the performance of VoIP traffic, the important parameters that affect the voice quality are:

1 End to end delay

End to end delay is determined from sender's side to receiver's side. Delay includes encoding/decoding, packetisation, propagation, and transmission delay. To maintain a voice quality, one way delay should be less than 150 ms.

$$\text{Delay (D)} = \sum_{i=1}^{n} \frac{d_i}{n} \tag{3}$$

where d_i is sum of all delays and n is number of delay parameters.

2 Jitter

It is a measure of variation in time of interval for consecutive packets delivery. The variation can be caused by serialisation, contention effect and queuing of IP networks. The packets enter into the jitter buffer at a variable rate and are taken out at a constant rate for decoding.

$$\text{AvgJitter} = \frac{\sum_{i=}^{n} |D_i|}{n} \tag{4}$$

3 MOS

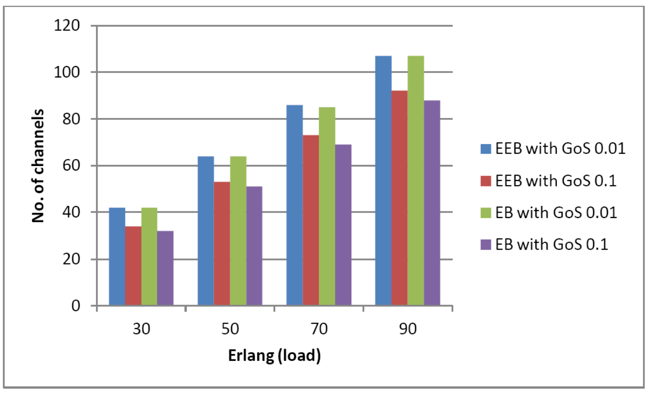
MOS is measurement parameter in which different voices of samples are collected and played back to a group of people who rank the voice quality between 1 to 5. 1 is worst case and 5 is the best case.

5.2 EEB and EB load

Figure 5 shows a graphical representation of calculation of number of channels with EB and EEB model by considering different GoS. We have varied load from 30 to 90 for simulation study. Channel requirement is more in case of GoS 0.1 as in comparison with GoS 0.01 because there are strict constraints on blocking probability. When load is varied from 30 to 90 with GoS 0.01, then there is no major difference in terms of channels requirement. In case with GoS 0.1, there is a variation of 3% to 5% in terms of number of channels with varying load because of blocking probability. Since we are using EB and EEB calculator, it calculates up to 90 Erlang only (Erlang and VoIP Bandwidth Calculator, 2014). When load is 90 Erlang, 107 channels are required to handle load on network. From this, total bandwidth required is calculated as:

$$\begin{aligned} Bandwidth &= number\ of\ channels \\ &\times bandwidth\ per\ channel \end{aligned}$$

Figure 5 Number of channels associated with Erlang (load) (see online version for colours)

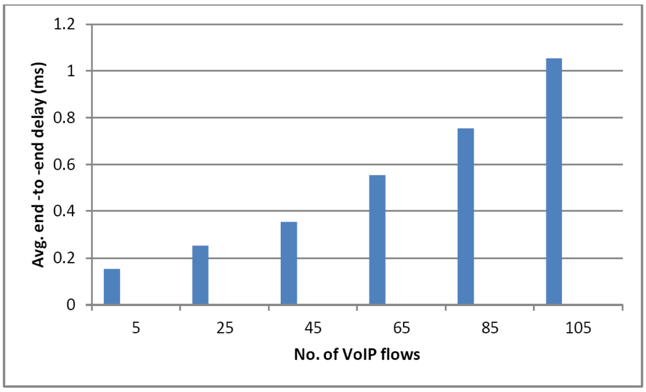


i.e., 107× 95 Kbps (calculated using step 2 of algorithm) that comes out to be approximately 10 Mbps. For simulation, we have considered varying load up to 105 for analysing the effect of average end to end delay, jitter and MOS with bandwidth as 10Mbps.

5.3 Average end to end delay

Figure 6 shows result of average end to end delay over number of VoIP flows. In this simulation, it can be seen that as the number of VoIP flows increases, average end to end delay also increases by up to 10%. The reason for this is as the load on network increases, delay also increases since server needs more time and computational resources for processing large numbers of registration packets, signalling packets, retransmission of INVITEs, etc.

Figure 6 Average end-to-end delay with number of VoIP flows (see online version for colours)



5.4 Average jitter

Figure 7 shows results of average jitter over number of VoIP flows. As the number of VoIP flows increases, average jitter also increases by up to 3%. As load on network increases, variation in delay also increases because of fixed jitter buffer. The server needs more time for processing large number of packets after signalling. As number of users increases, there would be large number of RTP packets and their processing in network can introduce jitter.

5.5 MOS

Figure 8 shows MOS over number of VoIP flows. We can analyse that as the number of VoIP flows increases, MOS increases or decreases by up to 0.9%. Since it is a theoretical measurement of users, as load increases on network, users may experience decrease in voice quality, here maximum MOS comes out to be 3.20789 during simulation.

Figure 7 Average jitter with number of VoIP flows (see online version for colours)

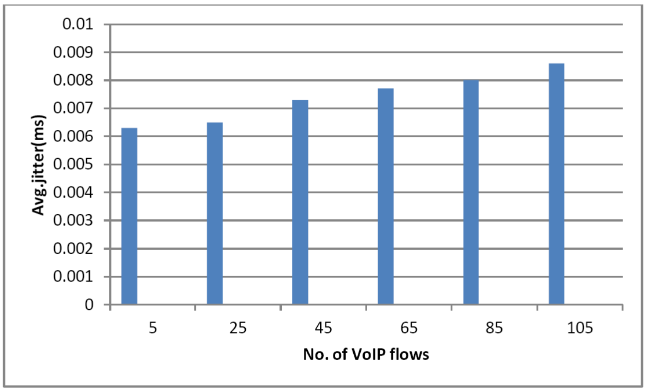
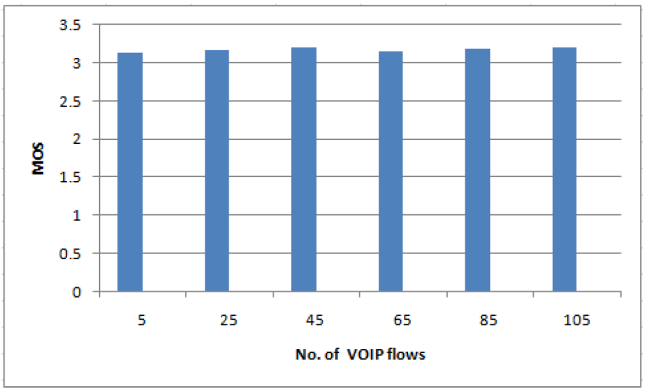


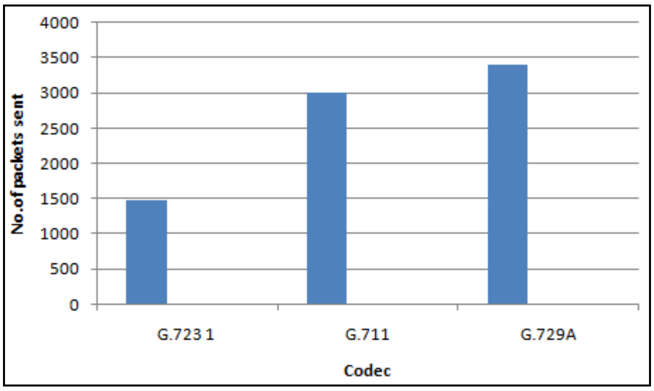
Figure 8 MOS associated with number of VoIP flows (see online version for colours)



5.6 Codec and max call load

Codec is a term used for coder/decoder that converts audio analogue signals into compressed digital signals for transmission and then again converts it into uncompressed audio signals for reception. There are different types of codec that are based on data rate, sampling rate and implemented compression. The most commonly used codecs for VoIP application are G.711, G.723.1 and G.729A. All are varying in terms of voice quality, the bandwidth required and delay.

Figure 9 Number of packets sent with different codec schemes (see online version for colours)



Simulation is also done to study the effect of different codec over number of packets sent as shown in Figure 9. In this simulation, we have considered three codecs, i.e., G.723.1, G.711, and G.729A. In case of G.723.1, since packetising interval is always 30 ms, so it will process packets with data rate 6.3 Kbps and fewer packets are sent on network, i.e., up to 1456. For G.711 and G.729A, packetising interval is 20ms, so data is transmitted with 64 Kbps and 8 Kbps, respectively. In this case, both these codecs are able to send 2,999 and 3,411 packets for processing to FreeSWITCH server using SIP.

6 Conclusions and future scope

In this paper, we have modelled VoIP traffic and proposed a scheme based on EEB model that determines number of trunks. EEB has better efficiency than EB model to manage the percentage of blocked calls using threshold value. QualNet 7.3 network simulator has been used and simulation results show that our technique is reliable and efficient for managing VoIP calls on network in terms of average end to end delay, jitter and MOS. In this paper, we have considered only single SIP-based call manager server for VoIP calls. In future work, we will consider load balancing and clustering that helps to maintain sessions in which requests corresponding to the same session are sent to the same server and if one server in a cluster of servers fails, it can temporarily remove that server from the cluster and divide the load onto the functioning servers. The work will also focus on different real-time applications and call flow scenarios like call forwarding on busy call, call forwarding in case of no answer, and different video calls on connectivity to PSTN and other global communication networks.

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