Scientific Journal of Impact Factor (SJIF): 4.72
e-ISSN (O): 2348-4470
p-ISSN (P): 2348-6406
International Journal of Advance Engineering and Research Development
Volume 4, Issue 8, August -2017

Application of Erlang B Model for Estimation of Performance Parameters in Modern IP Network Design

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ABSTRACT: *One of the most commonly used techniques for communications network design and evaluation is the analytic one. Unfortunately, it becomes inapplicable, due to the size and complexity of models. Analytical network design techniques are usually specific to circuit-switched networks. Network design techniques specific to packet-switched networks usually rely on network simulations. One class of problems for which simulation is rather unsuitable is those involving the estimation of probabilities of rare events. Such events as packet loss probabilities are typically designed to have very low values to guarantee a good quality of service. This implies simulating the system for a long time without the event occurring even once. Fortunately, recently, the Erlang B formula has been modified for IP networks. Therefore, in this paper, the Erlang B model is used for estimation of packet loss probability and bandwidth utilization for the core transmission links in the process of a large-scale IP backbone network design. Specifically, the paper describes the performance analysis of the IP backbone network topology designed for Benue State in Nigeria within the framework of a doctoral thesis. The entire scope of the paper includes performance analysis and comparison of the results between the Erlang B and simulation models of the IP backbone network. The results of the analysis of the two models compare favourably and further confirms the issue of rare packet loss probability associated with simulation among other interesting outcomes.*

Keywords – *Erlang B Model, Simulation Model, Rare Event Probability, Packet Loss Probability, Bandwidth Utilization, Packet-switched network, Queuing theory*

I. INTRODUCTION

Packet loss and delay and their impact on the performance of communication systems are known as Quality of Service (QoS) parameters. A major issue in the design of modern packet-switched telecommunication systems based on Internet Protocol (IP) is the prediction of the QoS that the network's user will get; or, conversely, how to engineer the network such that the service offered will satisfy the user's quality requirements while maintaining efficient usage of network resources [1]. In order to mathematically investigate the packet loss and delay properties of a telecommunication network or a component of such a network, a model of the system must be formulated in which such phenomena are expressed. This can be done using a branch of mathematics known as queueing theory [1]. In queueing theory, models are studied of systems in which customers randomly arrive at a service station in order to be served; since there may be other customers ahead of them, they may need to wait in a buffer or queue [1]. Queueing models are characterized by the probability distribution of the time between arrivals, the probability distribution of the time needed to serve a customer, size of the buffer space, queueing policy (e.g., first come first served), etc [1].

A queueing model of an IP communication network typically includes one or more sources which send packets into the network; these packets are then transmitted over a network link if the link is free, or stored temporarily in a buffer memory if the link is busy transmitting another packet. Given such a queueing model, the question is how to evaluate performance parameters of interest, such as the buffer overflow (or packet loss) probability and link utilization [1]. Ideally, one would like to do this using only analytical means, such as probability theory and calculus, in order to arrive at closed-form expressions of the performance measure in terms of the parameters of the model. Indeed, for simple models this is often possible. Such models typically contain only one queue, and/or the distribution of the interarrival and service times have nice properties (e.g., memoryless distribution), and/or the performance parameter is simple (e.g., the average waiting time). In slightly more complicated models, an explicit closed form expression may not be obtainable, but results may still be found by a numerical evaluation. Furthermore, in cases where an exact solution is not possible, approximations may be used to simplify the calculations. If analytical or numerical calculation is not possible and appropriate approximations are not available, simulation can often be useful [1].

One of the original motivating applications for the development of queueing theory was in fact the circuit-switched telephone network at the beginning of the 20th century: the waiting time until a call could be handled by an operator, and the probability of a call blocking due to the unavailability of free lines were among the first results by A. K. Erlang [1], [2]. The Danish mathematician focused exactly on problem of traffic load and its relationship to available capacity of trunk lines. The result of his effort are mainly two models for call loss or call waiting probabilities calculations also known as Erlang B and Erlang C models or formulas [2]. These equations put together offered traffic load, number of available telecommunication lines and probability of call not being processed immediately on its arrival. Especially the first Erlang's equation covers the problem of dimensioning of sufficient trunk lines capacity. Erlang B model is the basic model which does not contain the waiting queue [2]. Incoming calls are assigned to the idle server/line directly if there is any available, otherwise they are considered blocked or lost [2]. This implies the Erlang B model is widely used to dimension the trunk capacity between contact centre and communication networks. Today, the Voice over IP (VoIP) technology is more and more important, but the basic capacity problem is only slightly modified to available data throughput of the connection. Thus Erlang B model can be used in this case as well [2].

One class of problems for which simulation is rather unsuitable is those involving the estimation of probabilities of rare events, i.e., events which have a very low probability of occurrence [1]. Such events as packet loss probabilities are of much interest in queueing models of telecommunications systems, since these are typically designed to have very low values to guarantee a good quality of service [2]. These low probabilities imply that the system can be simulated for a long time without the event occurring even once [2]. As noted above, an event needs to be observed many times during a simulation run for the estimate of its probability to be accurate; therefore, the estimation of rare event probabilities requires impractically long simulation runs if no specialized techniques are used [2]. Hence the modified Erlang B model for IP networks serves to offer a useful alternative in this situation. Therefore, in this paper, the Erlang B model is used for estimation of packet loss probability and bandwidth utilization for the core transmission links in the process of a large-scale IP backbone network design. Specifically, the paper describes the performance analysis of the IP backbone network topology designed for Benue State in Nigeria within the framework of a doctoral thesis. The entire scope of the paper includes performance analysis and comparison of the results between the Erlang B and simulation models of the IP backbone network.

The rest of the paper is organized as follows: Section 2 discusses the design of telecommunications networks. This is followed by a discussion in section 3 about the modified Erlang B model for IP backbone networks. Section 4 is about experimental data and analysis. The comparative performance analysis of Erlang B and simulation models is presented in section 5. Lastly in section 6 is the conclusion.

II. DESIGN OF TELECOMMUNICATIONS NETWORKS

For the analysis of a telecommunication system, a model must be set up to describe the whole (or parts) of the system [3], [4]. This modelling process is fundamental especially for new applications of the traffic theory; it requires knowledge of both the technical system as well as the mathematical tools and the implementation of the model on a computer [4]. Such a model contains three main elements as depicted in Figure 1. They are system structure, the operational strategy, and the statistical properties of the traffic [4]. The system structure is technically determined and it is in principle possible to obtain any level of details in the description at component level. Reliability aspects are stochastic and will be considered as traffic with a high priority. The system structure is given by the physical or logical system which normally is presented in manuals [4]. As part of the operational strategy, a given physical system can be used in different ways in order to adapt the traffic system to the demand. In a telecommunication system, strategies are applied in order to give priority to call attempts and in order to route the traffic to the destination [4]. In Stored Program Control (SPC) or packet switching exchanges, the tasks assigned to the central processor are divided into classes with different priorities [4]. The classical telephone systems used wired logic in order to introduce strategies while in modern systems it is done by software, enabling more flexible and adaptive strategies.

Performance and dependability evaluation of modern systems becomes a challenging problem due to the complexity involved [5]. Several solution techniques are available in the literature. One of the most commonly used techniques is the analytic one which produces accurate results [5]. Unfortunately, it becomes inapplicable quickly, due to the size and complexity of models or due to non- Markovian nature of the problem involved [5]. In such cases, approximation methods are applied [5]. Even these approximation methods may become inefficient in most cases, and then simulation becomes inevitable [5].

2.1. Analytical Models

User demands are modelled by statistical properties of the traffic [4]. Only by measurements on real systems it is possible to validate that the theoretical modelling is in agreement with reality [4]. This process must necessarily be of an iterative nature. A mathematical model is built up from a solid knowledge of the traffic. Properties are then derived from the model and compared to measured data [4]. If they are not in satisfactory accordance with each other, a new iteration of the process must take place. It appears natural to split the description of the traffic properties into stochastic processes for arrival of call attempts and processes describing call holding times [4]. These two processes are normally assumed to be mutually independent meaning that the duration of a call is independent of the time the call arrived [4]. Models also exists for describing users experiencing blocking, i.e. they are refused service and may make a new call attempt a little later (repeated call attempts) [4]. One of such models is represented by Erlang B formula. Analytical network design techniques are usually specific to circuit-switched networks, where traffic between two nodes follows a well-defined path, and where end nodes keep traffic statistics allowing for the computation of the traffic matrix [6]. In a packet switched network, traffic between any two nodes may follow multiple paths, and each packet in the flow is routed independently. Consequently, network design techniques specific to packet-switched networks usually rely on network simulations and require accurate modelling and forecasting of the incoming traffic, as well as knowledge of the network traffic matrix [1].

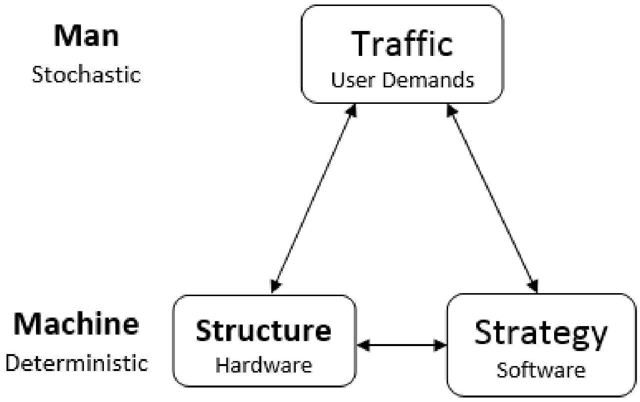

Figure 1. The elements of a telecommunication system model [3]

2.2. Simulation Models

An alternative to a mathematical model is a simulation model or a physical model (prototype) [4]. In a computer simulation model, it is common to use either collected data directly or to use statistical distributions [3], [4]. Unlike analytical models, which often require many assumptions and are too restrictive for most real-world systems, simulation modelling places few restrictions on the classes of systems under study [3]. The success of a simulation study hinges on identifying appropriate performance metrics and then devising a strategy for exploring the ensuing performance response surface [3], [7], [8].

Communications networks can be seen as users who generate demands for network resources, and protocols (distributed algorithms) that control the allocation of network resources to satisfy the demands [3]. The generation of user demands and their satisfaction are encapsulated in simulation events, which are ordered by their time of occurrence [3]. The action of the protocols depends on the state of the network at the time the demand was issued. A simple routing algorithm, for instance, may send packets to the output link with the shortest buffer. The simulation program based on these events is called Discrete Event Simulation (DES) [3]. Most simulation tools for telecommunications are based on DES [3]. Every DES has an initialization mechanism to establish the initial system state, statistical collection routines to obtain measurements, a post-processor to transform the collected statistics into the desired performance estimates, and a coordinating program to control the event list, post-processor, and initiation and termination of the simulation. Network entities acted upon by event routines include calls, messages, packets, and cells [3]. These entities are represented internally by data structures, which are often closely related to the massage/packet format defined by the protocol. For example, source and destination addresses as well as control information and data may be organized into standard packet formats. A data structure, representing a packet containing these elements in addition to simulating specific information, for instance, could be used for statistics collection [3]. The data field might contain a length indication or a pointer to another data structure that represents a network layer packet. Such encapsulation of data structures is a common feature of communications networks and also supported by object-oriented programming languages. For communication networks, developing a simulation program requires: modelling random user demands for network resources; characterizing network resources needed for processing those demands; and estimating system performance based on output data generated by the simulation [3].

III. THE MODIFIED ERLANG B MODEL FOR IP NETWORKS

One of the earliest and widely used mathematical models are the Erlang's formulas A, B and C. Recently, the Erlang B formula has been modified for IP networks [2] with the voice over IP (VoIP) technology becoming more and more important. Erlang B model is widely used to dimension the trunk capacity between contact centre and communication networks. Erlang B model is the basic model which does not contain the waiting queue. Incoming calls are assigned to the idle server/line directly if there is any available, otherwise they are considered blocked or lost [2].

The Erlang B formula uses four basic parameters:

A – the traffic load in Erlangs,

N – number of lines / trunks (requested simultaneous connections),

P_B – probability of call blocking.

k – number of users in the system

The original form of the equation allows us to find the blocking probability, P_B if *A* and *N* values are known. It is given as:

$$P_B(N,A) = \frac{\frac{A^N}{N!}}{\sum_{k=0}^{N}\frac{A^k}{k!}} \qquad (1)$$

If the rate of calls per unit time, λ is known and the average number of served requests per the same time unit μ then the traffic load can be easily evaluated as [2].

$$A = \frac{\lambda}{\mu} \tag{2}$$

If A is substituted in equation (5) the following form is obtained:

$$P_B(N, \lambda, \mu) = \frac{\left(\frac{\lambda}{\mu}\right)^N}{N!} \frac{1}{\sum_{k=0}^{N}\left[\left(\frac{\lambda}{\mu}\right)\frac{1}{k!}\right]} \tag{3}$$

It can be seen that it is the same formula as obtained for M/M/m/m queuing system [2].

In a VoIP environment, the same link between two neighbouring nodes of data network is shared among multiple data streams. Thus the basic characteristic of connection is not the number of lines but the throughput of the link, i.e. amount of data transferred per unit time [2]. Data are transferred in form of packets of various lengths for various applications. Quality of service depends on the codec used for voice encoding/decoding, packet loss during transmission, total cumulative delay during processing and transfer and several other factors [2]. Each codec defines the set of rules for voice packetization/depacketization, sample size, packet size, number of voice samples in one packet, packetization interval, etc. [2].

Based on the codec parameters, the required bandwidth per one VoIP connection can be derived as [2]:

$$w = \frac{d}{\tau} \quad [bit/s] \tag{4}$$

Where d is the packet size and τ is the packetization interval.

If VoIP data channel (link between nodes) is characterized by its capacity (link speed) W, then the theoretical link capacity expressed in terms of number of parallel connections the link can carry through can be calculated as [2]:

$$N' = \frac{W}{w} \tag{5}$$

The traffic load A' remains the same as for the classic telecommunication networks based on the average call arrival rate from sources and average line occupation time (average call duration), thus [2]

$$A' = \frac{\lambda}{\mu} \quad [erlang] \tag{6}$$

Hence, in the case of VoIP data network, packet loss probability (otherwise known as packet loss ratio is given by Erlang B formula modified as [2]:

$$P'_B(N', A') = \frac{\frac{A'^{N'}}{N'!}}{\sum_{k=0}^{N'} \frac{A'^k}{k!}} \tag{7}$$

IV. EXPERIMENTAL DATA AND ANALYSIS

The estimated average number of call arrivals and corresponding traffic intensities used for the design and simulation of the Benue State IP backbone network topology as reported in [9], [10] and [11] were used to carry out this study. Table 1 shows the aggregate traffics used for design and simulation of the Benue State IP backbone network topology [10].

4.1. Estimation of Packet Loss Probability

For estimation of packet loss probability and bandwidth utilization using the modified Erlang B model, it is necessary to estimate the equivalent number of channels, N' the core transmission links can carry. Towards this end, the data from previous study [9] were used to compute the aggregate average call arrival rate across the core transmission links. The number of channels, N' were estimated based on VoIP codec G.711 using Equations (4) and (5). Then the packet loss probability was estimated based on the estimated number of channels for each core transmission link using Erlang B traffic table. The results of the channels and packet loss probability estimations are presented in Table 2. The benchmark for packet loss probability is ≤1%.

Table 1. Aggregate traffic table for simulation of the core transmission links of Benue State IP backbone network [10]

Core Transmission Links	*Traffic Intensities in Erlangs*									
Gboko – Makurdi	3,679	7358	11038	14718	18398	22078	25758	29438	33098	36798
KatsinaAla – Gboko	1092	2180	3269	4358	5447	6536	7625	8714	9803	10892
Otukpo – Makurdi	2800	5603	8406	11209	14012	16815	19618	22421	25224	28027
Oju – Otukpo	909	1820	2731	3642	4553	5464	6375	7286	8197	9108
Okpoga – Otukpo	417	833	1249	1665	2081	2497	2913	3329	3745	4161
Vandeikya – Gboko	609	1219	1829	2439	3049	3659	4269	4879	5489	6099

Table 2. Estimated number of channels and packet loss probability using modified Erlang B model for IP networks

Core Transmission Link	*Number of Channels, N'*	*Traffic (erlangs)*	*Packet Loss (%)*
Gboko – Makurdi	18000	18092 19988 35998	1 10 50
Otukpo – Makurdi	31000	31227 34434 61998	1 10 50
Katsina-Ala – Gboko	58000	58493* 64434 115998	0 10 50
Oju – Otukpo	89000	89803* 98878 177998	0 10 50
Okpoga – Otukpo	105000	105964* 116656 209998	0 10 50
Vandeikya – Gboko	64000	64552* 71101 127998	0 10 50

*Values of traffic for packet loss probability of 1% are well outside the range specified for the respective transmission links (see Table 1). So the packet loss is taken as $\ll$ 1% or 0.

4.2. Estimation of Bandwidth Utilization

From Equation (5), the bandwidth of each transmission link is given by

$$W = N'w \quad (8)$$

Where N' represents the number of channels and w represents the bandwidth per one VoIP connection.

Substituting for the value of w from Equation (4) in Equation (8), gives

$$W = \frac{N'd}{\tau} \quad (9)$$

Where d and τ have their usual meanings.

From Equation (9), the number of channels is given by:

$$N' = \frac{W\tau}{dc} \quad (10)$$

Where c is the same number of call arrivals as used earlier for the estimation of packet loss probability.

Now, bandwidth utilization factor is given by:

$$\rho = \frac{A'}{N'} \quad (11)$$

Where A' and N' are the traffic intensity and number of channels respectively.

From Equation (11), we have:

$$N' = \frac{A'}{\rho} \qquad (12)$$

From Equations (10) and (12), we have:

$$\frac{W\tau}{dc} = \frac{A'}{\rho} \qquad (13)$$

From Equation (13), the bandwidth of each channel is obtained as:

$$W = \frac{A'dc}{\rho\tau} \qquad (14)$$

Or $$\rho = \frac{A'dc}{W\tau} \qquad (15)$$

But, based on the VoIP codec G.711, d (packet size) = 1744 bits, and τ (packetization interval) = 20 seconds. Substituting these values in Equation (15) gives:

$$\rho = \frac{87A'c}{W} \qquad (16)$$

The values of bandwidth utilization estimated using Equation (16) are as shown in Table 3. The values of bandwidth utilization were estimated for ten separate values of call arrivals varied in accordance with the network traffic changes.

V. COMPARATIVE PERFORMANCE ANALYSIS OF ERLANG B AND SIMULATION MODELS

The packet loss and bandwidth utilization performances of the Erlang B model and simulation model of the Benue State IP backbone network were compared to verify the issue of rare occurrence of packet loss probability associated with simulation and for validation of the simulation model. The detailed development of the simulation model and the results of its performance analyses are reported in [10], [11].

It should be noted that the Erlang B model has yielded the values of packet loss probability of $\leq$ 1% up to the fifth value of traffic intensity for the Gboko-Makurdi core transmission link. The fifth value of traffic intensity corresponds to 100% of the estimated aggregate traffic along that link. For the Otukpo-Makurdi core transmission link, the value of traffic intensity corresponding to 1% packet loss probability falls just outside the range specified for the link. The last traffic in this range is 28027 erlangs while the traffic corresponding to 1% packet loss probability is 31227 erlangs. This implies that the packet loss probability for link is less than 1% for the entire traffic range. These results are in contrast to the results of the simulation model which gave the value of packet loss probability as 0 for all the six core transmission links thereby confirming the issue of rare packet loss probability occasioned by simulation. However, the results of the Erlang B model for the performance metric is still 0 for all the other four core transmission links showing a favourable comparison as indicated in Table 2.

It is instructive to note that the performance implications of the packet loss calculations by the Erlang B formula for the six core transmission links compare favourably with that revealed by bandwidth utilization obtained from the simulation process. For instance, it can be seen in Table 3 that the performance of Otukpo-Makurdi core transmission link is still a bit better than the Gboko-Makurdi link with the Erlang B model just as it was with the simulation model. Also all the other four core transmission links have no serious performance implications as revealed by the packet loss and bandwidth utilization performances for both the Erlang B and simulation models (see Table 3). However, it can be noticed that the results of the Erlang B model generally give a picture of a less stable network at high values of traffic and a much more stable network at low values of traffic.

Table 3 represents the packet loss probability and bandwidth utilization performances for the Benue State IP backbone network. The simulation model results of the network are compared with the results obtained with Erlang B model. It is noteworthy that the results compare favourably which serve to demonstrate the validity of the simulation method. The benchmarks for packet loss probability and bandwidth utilization are 1% and 100% respectively.

Table 3. Comparison of bandwidth utilization and packet loss performance for IP backbone network

Core Transmission Link	*Traffic Intensity (erlangs)*	*Simulation Model*		*Erlang B Model*	
		Bandwidth Utilization (%)	*Packet Loss Probability (%)*	*Bandwidth Utilization (%)*	*Packet Loss Probability (%)*
Gboko-Makurdi	3,679	14	0	4	<1
	7358	28	0	16	<1
	11038	43	0	36	<1
	14718	56	0	65	<1
	18398	70	0	98	1
	22078	99	0	143	>10
	25758	100	0	198	>10
	29438	100	0	256	<50
	33098	100	0	318	<50
	36798	100	0	401	>50
Otukpo-Makurdi	2800	12	0	2	<1
	5603	24	0	8	<1
	8406	37	0	17	<1
	11209	49	0	29	<1
	14012	61	0	46	<1
	16815	75	0	67	<1
	19618	100	0	89	<1
	22421	100	0	106	<1
	25224	100	0	149	<1
	28027	100	0	180	<1
Oju-Otukpo	909	7	0	0.2	0
	1820	14	0	1	0
	2731	22	0	2	0
	3642	28	0	3	0
	4553	36	0	5	0
	5464	43	0	7	0
	6375	50	0	10	0
	7286	54	0	13	0
	8197	58	0	17	0
	9108	60	0	21	0
Okpoga-Otukpo	417	7	0	0.1	0
	833	14	0	0.3	0
	1249	22	0	1	0
	1665	28	0	1	0
	2081	36	0	2	0
	2497	43	0	3	0
	2913	50	0	4	0
	3329	54	0	5	0
	3745	58	0	7	0
	4161	60	0	8	0
Katsina-Ala-Gboko	1092	9	0	0.4	0
	2180	17	0	2	0
	3269	25	0	4	0
	4358	33	0	6	0
	5447	42	0	10	0
	6536	50	0	14	0
	7625	56	0	18	0
	8714	60	0	24	0
	9803	64	0	31	0
	10892	68	0	37	0
Vandeikya-Gboko	609	8	0	0.2	0
	1219	16	0	1	0
	1829	24	0	2	0
	2439	33	0	3	0
	3049	40	0	5	0
	3659	48	0	7	0
	4269	54	0	9	0
	4879	58	0	12	0
	5489	62	0	16	0
	6099	66	0	19	0

Figures 2 - 7 show the graphical comparison of results for bandwidth utilization between the simulation and Erlang B models. Figure 2 shows the graph for the Gboko-Makurdi core transmission link. It can be seen from the graph that the curves for the Erlang B model and simulation model compare favourably within the utilization region corresponding to lower values of traffic intensity up to 22078 erlangs. Beyond this point, the gap between the curves increases progressively until they become widely separated at the highest traffic intensity. While the simulation model shows the normal utilization region up to the sixth increase in traffic intensity, the Erlang B model shows it up to the fifth increase. Both curves show serious congestion of the network starting from the seventh and sixth traffic with utilizations of 100% for the simulation model and 143% for the Erlang B model respectively. It should be noted that the fifth traffic intensity is the maximum estimated traffic for the network. This is an indication of validity of the simulation model. For network management purposes, the graph of the simulation model gives a better indication of network behaviour as it peaks at 100% which is the maximum limit of the normal utilization region. Thus simulation facilitates not only the network design process but also management and operations of IP networks better than analytical methods.

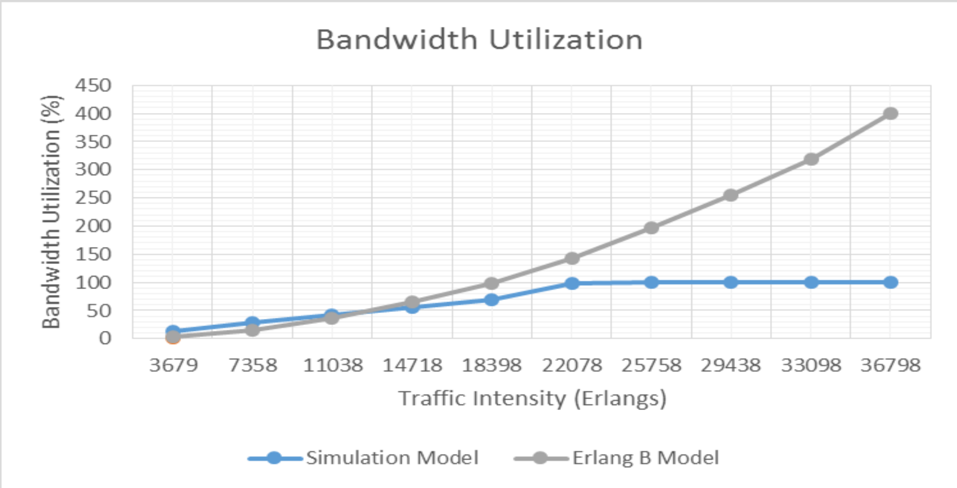

Figure 2. Comparison of bandwidth utilization performance for Gboko-Makurdi core transmission link

The graph showing the comparison of the bandwidth utilization performance for Otukpo-Makurdi core transmission link is shown in Figure 3. It can be seen that the graph indicates better performance than the Gboko-Makurdi link. This is instructive because the Gboko-Makurdi is more highly loaded. It can be seen that the network bandwidth utilization performances of the Erlang B model and simulation model also compare favourably only at the lower values of traffic intensity in the same fashion as the Gboko-Makurdi core transmission link. Both curves show the normal utilization region up to the sixth and seventh increase in traffic intensity with utilizations of 75% and 89% respectively. Both models show a state of congestion for the network from about 20000 erlangs to the last increase in traffic intensity. The Erlang B model displays a trend of utilization which is low at lower traffic loads and increases rapidly at higher loads while the simulation model displays a graceful utilization trend with both low and higher traffics as can be seen between the Gboko-Makurdi and Otukpo-Makurdi transmission links. This is a clear indication that the simulation process is better for designing IP backbone networks optimally.

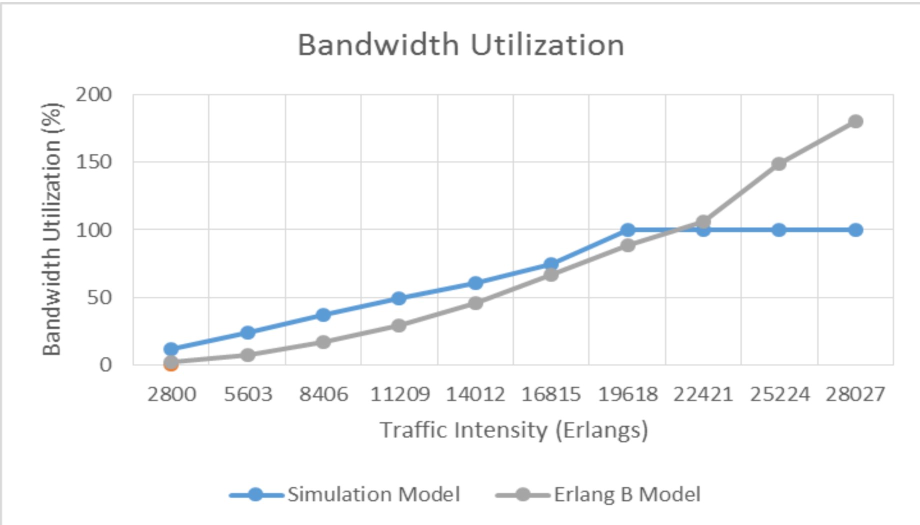

Figure 3. Comparison of bandwidth utilization performance for Otukpo-Makurdi core transmission link

Figure 4 represents the bandwidth utilization performance comparison for the Oju-Otukpo core transmission link. It can be noticed that the Erlang B model curve is closer to the horizontal axis than the simulation model and both curves are more separated for the entire traffic range than did the graphs for the Gboko-Makurdi and Otukpo-Makurdi transmission links. This is a clear indication that the two models compare more favourably at high traffic loads than at low traffic loads because the Oju-Otukpo link is relatively lightly loaded. It can be seen that both models are performing normally with the link utilization increasing with the traffic load; as the traffic increases, the simulation model curve seem to be bending downwards and the Erlang B model curve upwards towards a certain point of intersection. Going by the graphs of the Gboko-Makurdi and Otukpo-Makurdi links, this point is likely to be close to the point at which congestion occurs in the network. Hence it may be concluded that the separation between the two curves at the highest traffic represents a measure of scalability of the network. This is a clear indication in favour of using the simulation and Erlang B methods for cooperating the design process of large-scale IP backbone networks. It can be noticed that Figures 4 through 7 look similar. This is because the Oju-Otukpo, Okpoga-Otukpo, Katsina-Ala – Gboko and Vandeikya-Gboko core transmission links have similar traffic characteristics of all being lightly loaded.

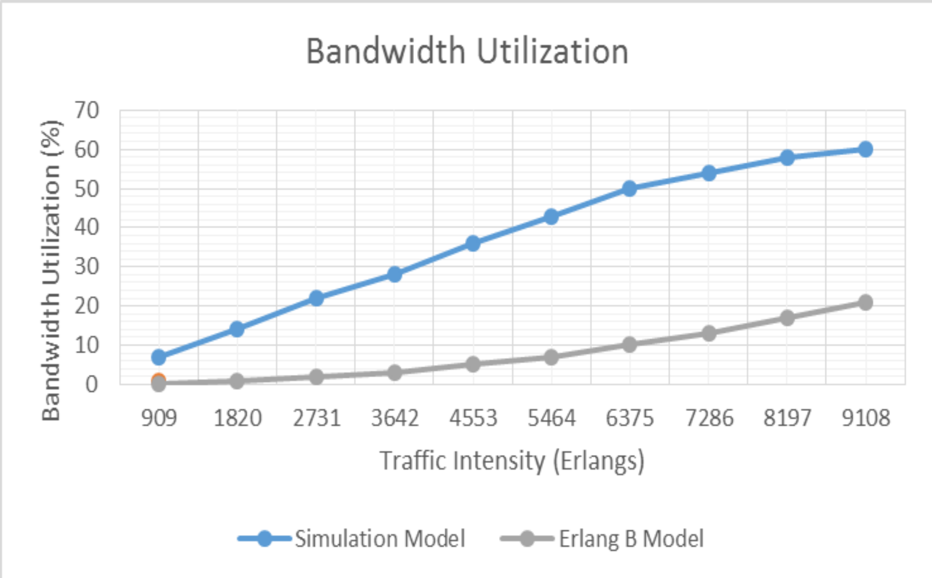

Figure 4. Comparison of bandwidth utilization performance for Oju-Otukpo core transmission link

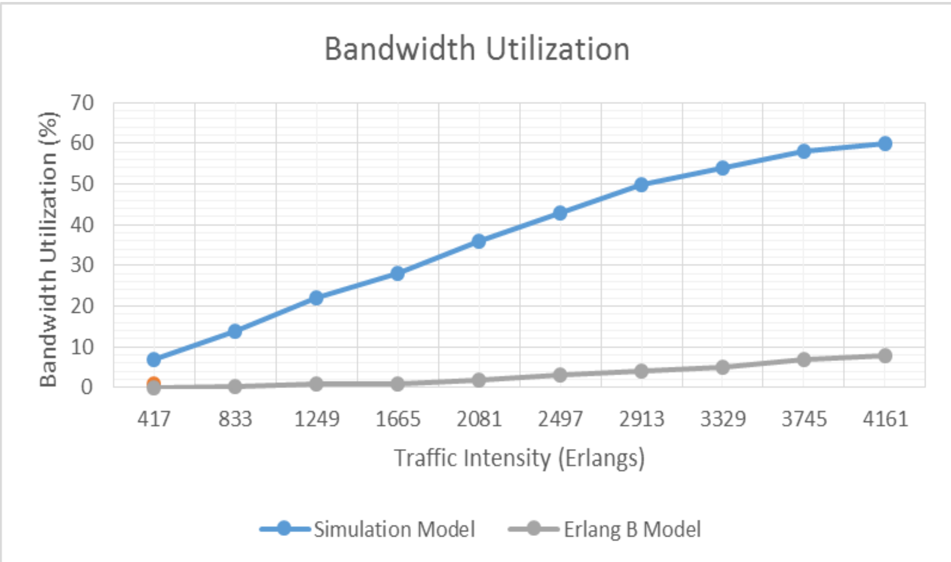

Figure 5. Comparison of bandwidth utilization performance for Okpoga-Otukpo core transmission link

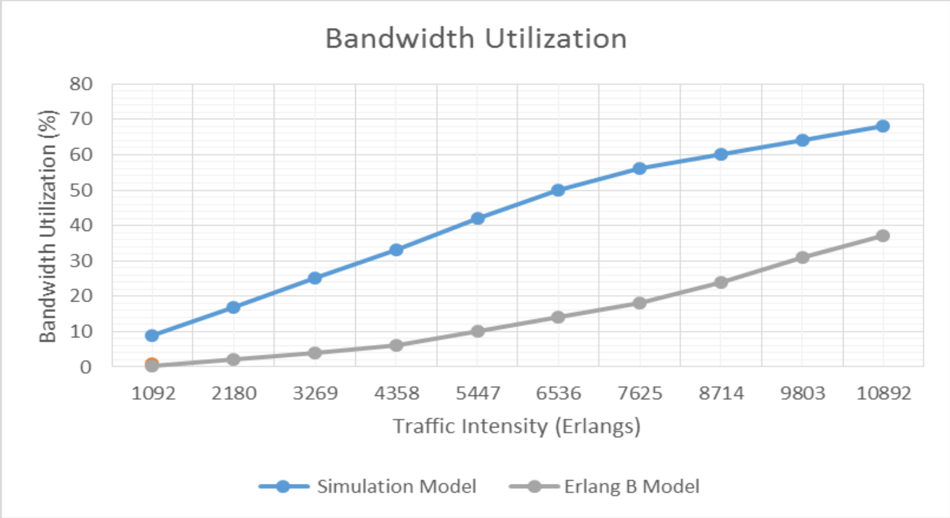

Figure 6. Comparison of bandwidth utilization performance for Katsina-Ala - Gboko core transmission link

Figures 5 and 7 represent the bandwidth utilization performance comparison for the Okpoga – Otukpo and Vandeikya-Gboko core transmission links respectively. It can be noticed that the two curves are more separated from each other because the highest traffic point for the two links are the least, 4161 and 6099 erlangs respectively. There is no doubt that the two links are the most scalable of all the six core transmission links. In the same vein, if examined carefully, the bandwidth utilization characteristics for Oju-Otukpo and Katsina-Ala – Gboko transmission links (see Figures 3 and 6) are closely related to each other and can be used to reach similar conclusions.

From the foregoing analysis, it is clear that the results of Erlang B model compare and complements each other favourably. It can be seen that the four subnetworks, namely Otukpo-oju, Otukpo-Okpoga, Gboko-KatsinaAla, and Gboko-Vandeikya transmission links are all lightly loaded and hence they exhibit similar performance characteristics and their designs are highly scalable. On the other hand, the Gboko-Makurdi and Otukpo-Makurdi links are highly loaded and both show congestion at middle range of traffic load variations.

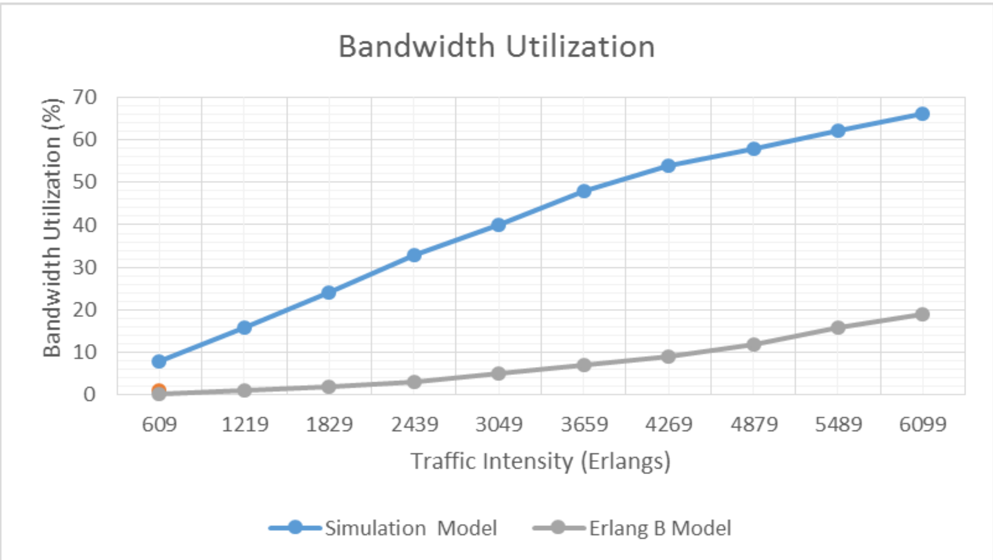

Figure 7. Comparison of bandwidth utilization performance for Vandeikya - Gboko core transmission link

VI. CONCLUSION

The entire scope of this paper includes performance analysis and comparison of the results between the Erlang B and simulation models of IP backbone network. Starting from basic principles of the Erlang B model adaptation for IP networks, it was possible to estimate the equivalent number of channels each of the core transmission links in the IP backbone network can carry. Then as usual, the packet loss probability was estimated from Erlang B traffic table using the number of channels and aggregate traffic along the core transmission links. The bandwidth utilization was estimated using equations derived also from the basic principles of the Erlang B model adaptation. The results of the performance analyses revealed the following interesting situations: packet loss probability is obtainable only when the network traffic is appreciably high; at low values of network traffic, the packet loss probability was observed to be 0 for both the Erlang B and simulation models; the bandwidth utilization performances of the Erlang B and simulation models compare more favourably when the network traffic is high than when it is low; the intersection point of the bandwidth utilization curves for the Erlang B and simulation models falls close to the point of maximum utilization of the network; the separation between the Erlang B and simulation model bandwidth utilization curves at the highest traffic represents a measure of scalability of the network. Another interesting revelation is the confirmation of rare packet loss probability associated with simulation. Apart from the Gboko-Makurdi and Otukpo-Makurdi transmission links which were highly loaded, all the other core transmission links which were lightly loaded produced 0 packet loss probability. On the whole, the revelations from the results of performance analysis of the Erlang B and simulation models leads to the conclusion that combined use of the models provides a worthwhile approach for the design large-scale IP communications networks.

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