

Internet Traffic Modeling and Capacity Evaluation in UMTS

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Abstract

UMTS (Universal Mobile Telecommunications System) can be connected to the data networks. So Internet traffic services such as WWW browsing, email, ftp, SMTP (Simple Mail Transfer Protocol) should be handled by UMTS. Each of the traffic services has some specific properties but all of them obey a basic ON/OFF model. For a specific traffic user there is some unused (OFF) times between transmitted packets which we may use them for other services. In this paper based on the offered traffic services, models and parameters presented in 3GPP (3th Generation Partnership Project) we calculate ON and OFF time durations of the above traffic services in some different data rates, then calculate their corresponding activity factors. Finally we survey their impacts on the UMTS capacity and users. Simulation showed that the number of packet traffic users could be increased in comparison with circuit switch voice traffic users because they have lower Erlang traffic figures.

1. Introduction

Voice service is no longer the only service in wireless and mobile system. Multiple traffic systems now include voice and data services, or more accurately, many differentiated services with distinctive QoS requirements. That is even within the data services, there are many types of services which have different characteristics and models [1].

In order to use radio resources optimally we need clear and precise models for different services. Services are divided to real time and non-real time services. Voice and peer to peer video communications are examples of real time traffic. Internet traffic services are subset of non-real traffic services which will be reviewed in this paper. Totally traffic models are based on the statistics characteristics of the services. The Internet traffic model presented in [2] like the 3GPP model presented in [3] uses a multi-layer model for describing sources which in the lowest layer uses Weibull and Pareto distributions for ON and OFF durations. The other model presented in [4], [5] assigns Log-Normal and Pareto distributions to ON and OFF durations of Internet traffic respectively.

In third generation WCDMA systems, data applications are expected to finally dominate the overall traffic volume [17]. The traffic generated by data applications is inherently bursty and asymmetric, with higher data rates in the downlink than in the uplink [18]. Tripathi and Capone in [6] and [7] also verified the performance of different types of services in IS-2000

system. Besides Soldani in [4], [8] worked on QoS of UMTS based on the offered traffic model.

Because Internet traffic is bursty we can transmit traffic information of other users in the empty sections of the bursts. We can use the activity factor as a criterion for this burstiness, besides burstiness depends on the traffic type and model. In this paper we use the burstiness property of Internet traffic based on the models presented in 3GPP. So this paper is organized as follows: traffic model is explained in the next section. Because web browsing traffic is the most common traffic type, in this section we emphasize to it as a general traffic model. Based on the presented models and their characteristics, in section 3 we pay attention to activity factor calculations of some data traffic services.

On the other hand we can assume a channel as a server and since we know the WCDMA system as an interference limited system and define soft capacity in that, we can theoretically hypothesis extremely large channels for that system. So, section 4 gives system capacity based on $M/M/\infty/M$ queuing model [9]. Then we apply the activity factor to it. Sketching, evaluation and conclusion are given in the next two sections.

2. Traffic model

A number of voice and data models are represented as ON/OFF source models. We can use this simple and flexible model for the traffic sources [10], [11], [12]. When a source is ON, it generates packets with a constant inter arrival time. When the source is OFF, it does not generate any packets (Fig. 1).

Regarding to the voice source modeling, the process of a voice call which transits between ON and OFF states can be modeled as a two-state Markov chain. The state transition diagram shown in Fig. 1 depicts how the state transition occurs in such a way that the amount of time spent in each state is exponentially distributed and, given the present state of source traffic, the future is independent of the past (Markov chain). λ is OFF to ON rate and μ is ON to OFF rate and the average lengths of the ON and OFF periods are $1/\mu$ and $1/\lambda$ respectively.

For all real time services, calls should be generated according to a Poisson process assuming a mean call duration of 120 seconds for speech and circuit switched data services [3]. As mentioned above the traffic model of speech should be an ON-OFF model with activity and silent periods being generated by an exponential distribution with mean values for active and silence periods both equal to 3 seconds [3].

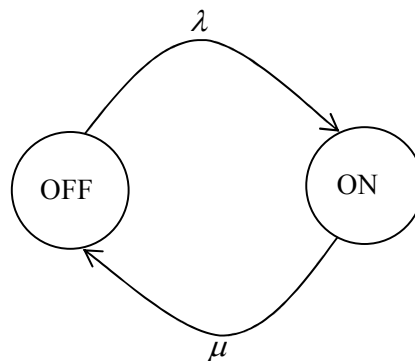


Figure 1. ON-OFF models

Data services which are carried by circuit switched connections are circuit switched data services. A user can communicate with the network after a connection is established. For circuit switched data services, the traffic model should be a constant bit rate model, with

100% of activity [2], [3], [14]. In the real time systems we don't need any queue. Indeed in this case we should use a M/M/m/m Erlang model in which the system capacity (queue length) and the number of servers are equal. For example if there are m servers in a telephony system, m subscribers (channels) can connect to the system to make a call at most. Now if all the channels are busy at a moment not only any new subscriber can not connect to the network but also his turn can not be preserved.

In a non-real time system we can use a queue in the system. Here we can use M/M/m/K/M model which has m servers, a queue of length K and M subscribers.

For data traffic services such as Web traffic it has been shown that the probability of loading large file sizes is not negligible. The ON duration is effectively characterized by heavy-tailed models and the OFF duration is determined by the user's think time which is also modeled by the heavy tailed model [13]. A random variable X can be said to have a heavy-tailed distribution if its complementary cumulative distribution function (CCDF) has $P_r\{X > x\} \approx x^{-\sigma}$ as $x \rightarrow \infty$ where $0 < \sigma < 2$. Roughly speaking, the asymptotic shape of the distribution follows a power law, in contrast to exponentially decay. One of the simplest heavy-tailed distributions is the Pareto distribution.

Fig. 2 depicts a typical non-real time WWW browsing session, which consists of a sequence of packet calls. The user initiates a packet call when requesting an information entity. During a packet call several packets may be generated. It means that the packet calls are constituted of a number of the burst sequence of packets. The 3GPP traffic model has three tiers, namely, session, packet call and packet as illustrated in Fig. 2.

A packet service session contains one or several packet calls depending on the application. For example in a WWW browsing session a packet call corresponds to the downloading of a WWW document.

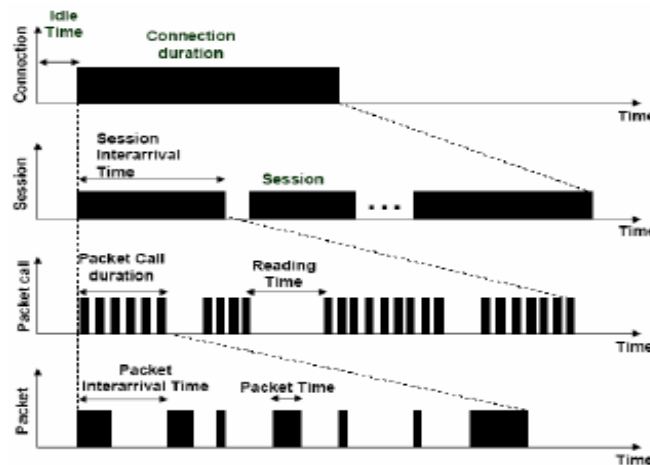


Figure 2. Three tier traffic model representing packet mode traffic

After the document is entirely arrived to the terminal, the user is consuming certain amount of time for studying the information. This time interval is called reading time. It is also possible that the session contains only one packet call. In fact this is the case for a File Transfer Protocol (FTP). As in [3], [4] the following statistical models may be applied to web browsing in order to catch its typical behaviour described in Fig. 2:

- Session arrival process

The arrival of session set-up request to the network is modeled as a Poisson process with λ rate.

- Number of packet calls per session, N_{pc}

This is a geometrically distributed random variable with a mean $\mu_{N_{pc}}$ [packet calls], i.e.,

$$N_{PC} \in Geom(\mu_{N_{PC}}) \quad (1)$$

- Reading time between packet calls, D_{pc}

This is a geometrically distributed random variable with a mean $\mu_{D_{pc}}$ [reading time steps], i.e.,

$$D_{PC} \in Geom(\mu_{D_{PC}}) \quad (2)$$

- Number of datagrams within a packet call, N_d can be geometrically distributed random variable with a mean μ_{N_d} [packet], i.e.,

$$N_d \in Geom(\mu_{N_d}) \quad (3)$$

- Inter arrival time between datagrams (within a packet call) D_d . This is a geometrically distributed random variable with a mean μ_{D_d} [model time steps], i.e.,

$$D_d \in Geom(\mu_{D_d}) \quad (4)$$

- Size of a datagram, S_d .

Pareto distribution with cut-off is used for packet size. Packet size is defined with the following formula:

$$PacketSize = \min(x, m) \quad (5)$$

where x is normal Pareto distributed random variable ($\alpha = 1.1$, $k = 81.5$ bytes) and m is maximum allowed packet size, $m = 66666$ bytes [3]. The PDF of the Packet size becomes:

$$f_x(x) = \begin{cases} \frac{\alpha \cdot k^\alpha}{x^{\alpha+1}}, & k \leq x \leq m \\ \beta, & x > m \end{cases} \quad (6)$$

where β is the probability that $x > m$. It can easily be calculated as:

$$\beta = \int_m^\infty f_x(x) dx = \left(\frac{k}{m}\right)^\alpha \quad \alpha > 1 \quad (7)$$

In (7), the requisite that $\alpha > 1$ is only to ensure that the $E[X]$ of the Pareto distribution exists.

The result (7) is valid for all $\alpha > 1$.

Then mean packet size can be calculated as:

$$\mu_x = \int_{-\infty}^\infty x f_x(x) dx = \int_k^{m^-} x \frac{\alpha \cdot k^\alpha}{x^{\alpha+1}} dx + m \left(\frac{k}{m}\right)^\alpha \quad (8)$$

And after simplifying we have

$$\mu_x = \frac{\alpha k - m \left(\frac{k}{m} \right)^\alpha}{\alpha - 1} \tag{9}$$

With the parameters above the average packet size is $\mu_x = 480$ bytes.

Table 1 gives default mean values for the distributions of some typical www services of different rates. As a consequence, the average size of a packet call is $\mu_{N_d}, \mu_x = 25 \times 480 = 12kbytes$.

We can also calculate the parameters of other traffic types according to traffic models presented in Table 2.

3. Activity factor

Fig. 3 shows an example of time-based ON/OFF trajectory of the traffic activity [15]. The random characteristic of traffic activity is assumed to be represented by the mean of traffic activity, called the traffic activity factor.

Table 1. Characteristics of datagram information types [3]

Packet based information types	Average number of packet calls	Average reading time between packet calls [h]	Average amount of packets within a packet call [l]	Average interarrival time between packets [s]	Parameters for packet size distribution
WWW surfing UTDD 8 kbits	5	412	25	0.5	$K=01.5, \alpha=0.1$
WWW surfing UTDD 32 kbits	5	412	25	0.25	$K=01.5, \alpha=0.1$
WWW surfing UTDD 64 kbits	5	412	25	0.0525	$K=01.5, \alpha=0.1$
WWW surfing UTDD 144 kbits	5	412	25	0.0237	$K=01.5, \alpha=0.1$
WWW surfing UTDD 384 kbits	5	412	25	0.01040	$K=01.5, \alpha=0.1$
WWW surfing UTDD 2048 kbits	5	412	25	0.00485	$K=01.5, \alpha=0.1$

Table 2. Traffic models and their characteristics [6]

A traffic source usually alternates active and silent periods. Indeed activity factor represents the fraction of the time that the source is generating traffic. In the OFF (Idle) time the source doesn't generate any packet. The activity factor of voice or data traffic, α , is defined as the probability that the state is ON and can be given as

$$\alpha = \frac{E[ON \text{ duration}]}{E[ON \text{ duration}] + E[OFF \text{ duration}]} \tag{10}$$

We now calculate the activity factor α for traffic types Telnet, WWW, ftp, E-mail and fax based on the Table 2 and 64, 144 and 384 Kbps data rates. The results are in Table 3. Here, we only calculate the web browsing traffic activity factor regarding to third column of the Table 2.

According to that Table we have 5 packet calls per a session, inter arrival time of 120s between packet calls, 25 packets per packet call, average packet size of 480 bytes and inter arrival time of 0.067s between packets. We first calculate the whole OFF time in WWW session. The average time of a packet is $480 \times 8 / 64000 = 0.06s$ and the OFF time between two consecutive packets is $0.067 - 0.06 = 0.007s$ and OFF time in a packet call is $(25 - 1) \times 0.007 = 0.168s$. The whole OFF time between packet calls is $5 \times 0.168 = 0.84s$. So the whole OFF time in a session is $4 \times 120 + 0.84 = 480.84s$.

Now we calculate ON time of a WWW session in a packet call as $25 \times 0.06 = 1.5s$. So the whole ON time in a WWW session is $5 \times 1.5 = 7.5s$.

Finally the activity factor α is calculated as $\alpha = 7.5 / (7.5 + 480.84) = 0.015$. We have also calculated this parameter for other services and data rates which are listed in Table 3.



Figure 3. An illustration of time-based ON/OFF trajectory of traffic activity.

Table 3. Activity Factors based on the Table 2 and data rate of 64, 144 and 384kbps

	ON duration(s)	OFF duration(s)	Activity Factor	ON duration(s)	OFF duration(s)	Activity Factor	ON duration(s)	OFF duration(s)	Activity Factor
Telnet	1.28	111.73	0.0113	0.57	112.43	0.005	0.217	112.79	0.0019
WWW	7.5	480.84	0.015	3.34	484.84	0.0068	1.25	486.84	0.00256
ftp	3.66	0.427	0.89	1.63	2.44	0.4	0.61	3.477	0.149
E-mail	1.8	90.1	0.019	0.43	90.56	0.0094	0.3	90.8	0.0033
fax	6.66	82.68	0.075	2.96	86.28	0.033	1.11	88.1	0.012
	Data rate 64kbps			Data rate 144kbps			Data rate 384kbps		

4. Capacity

Let us consider an area where a cellular system is deployed and there are M subscribers in the area generating calls (sessions). Subscribers are either in the system or outside the system and in some sense 'idle'. Then, when a subscriber is in the 'idle' state then the time it takes him to arrive in the system is a random variable with exponential distribution whose mean is $1/\lambda$ sec. Let us also assume that call duration is exponentially distributed with an average value of $1/\mu$ sec. Then, a continuous time birth-death Markov chain can be built, where the states are given by the number of users having a call in progress, usually denoted as active users (Fig. 4).

When there are N users with a call in progress, the birth rate λ_N and death rate μ_N in the state N are given by [9, 16]:

$$\lambda_N = \begin{cases} (M - N)\lambda & 0 \leq N \leq M \\ 0 & otherwise \end{cases} \quad (11)$$

$$\mu_N = N\mu \quad N = 1, 2, \dots$$

Under equilibrium conditions it can be obtained that the probability of having N users with a call in progress is given by:

$$\begin{aligned}
 p_N &= p_0 \prod_{i=0}^{N-1} \frac{\lambda(M-i)}{(i+1)\mu} \\
 &= p_0 \left(\frac{\lambda}{\mu}\right)^N \binom{M}{N} \quad 0 \leq N \leq M
 \end{aligned}
 \tag{12}$$

From the above equation and $\sum_{N=0}^M p_N = 1$ we find

$$p_0 = \frac{1}{(1 + \lambda/\mu)^M}
 \tag{13}$$

Thus

$$p_N = \frac{\binom{M}{N} \left(\frac{\lambda}{\mu}\right)^N}{\left(1 + \frac{\lambda}{\mu}\right)^M} = \binom{M}{N} \frac{\rho^N}{(1 + \rho)^M}
 \tag{14}$$

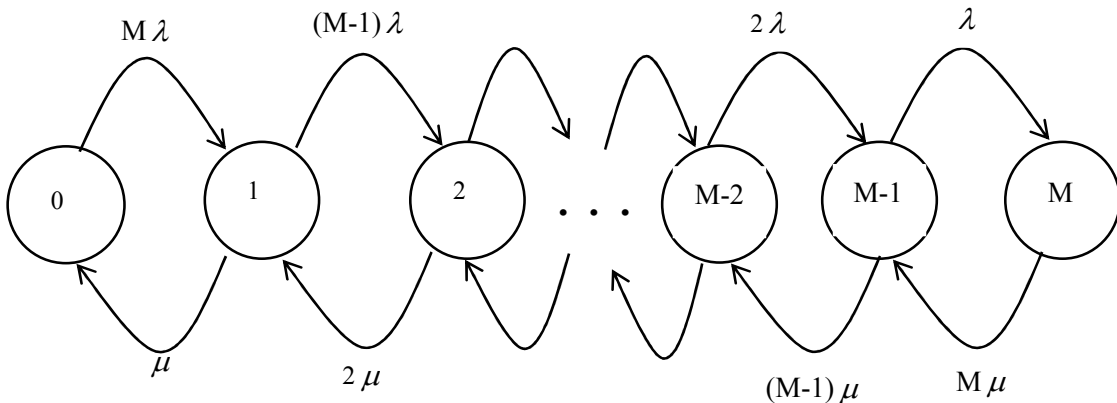


Figure 4. State-transition-rate diagram for infinite server finite subscriber system M/M/∞//M

where it has been considered that $\rho = \lambda/\mu$. ρ is arrival/outgoing ratio.

Then provided that there are N users with a call in progress, the conditional probability of n simultaneous users occupying the radio interface is given by:

$$p_{n|N} = \binom{N}{n} \alpha^n (1 - \alpha)^{N-n}
 \tag{15}$$

Therefore, the probability of n users simultaneously transmitting can be computed as:

$$p_n = \sum_{N=n}^M p_{n|N} p_N = \binom{M}{n} \frac{(\alpha\rho)^n (1 + (1 - \alpha)\rho)^{M-n}}{(1 + \rho)^M}
 \tag{16}$$

Where in (16), M , N and n are the number of users camping on a cell, the number of users with a call in progress and the number of users transmitting in a given moment.

5. Sketching and evaluation

From (14) the cumulative distribution function (CDF) of the number of users with a call in progress, N , for the case of M camping users is:

$$F_N(k) = \text{Pr ob}(N \leq k) = \sum_{j=0}^k \binom{M}{j} \frac{\rho^j}{(1+\rho)^M} \quad k = 0, \dots, M-1 \quad (17)$$

and the cumulative distribution function (CDF) of the number of users with a call in progress simultaneously, n , for the case of M camping users is:

$$F_n(k) = \text{Pr ob}(n \leq k) = \sum_{j=0}^k \binom{M}{j} \frac{(\alpha\rho)^j (1+(1-\alpha)\rho)^{M-j}}{(1+\rho)^M} \quad k = 0, \dots, M-1 \quad (18)$$

As we see from the above formula, in an interference-limited scenario like WCDMA, n is a very important parameter and it depends on α and M and ρ .

After sketching the CDF users by the above formulas and Table 3 using MATLAB we find the Figs. 5 through 9. We used the case in which the population is $M=100$ camping users and $\rho = 0.5$ which means that incoming users are equal to outgoing users. Activity factors α with values given in the Table 3 are as parameters in Figs. 6 through 9.

It can be seen from Fig. 5 that the probability of having 38 active data users at the same time is around 10%. Similarly the probability of having less than 27 active users at the same time is also around 10%. From Fig. 6 we see that there is around 8% probability of having more than 1 simultaneous user for www user with the data rate 64Kbps. Besides this probability is around 3% and 1% for the www users with 144Kbps and 384Kbps data rates respectively. Comparing Figs. 6 & 7 we find they are very similar. This is because their average activity factors are very close. Figs. 8 & 9 show that for Email and Fax users the probability of having simultaneous active users is more than the cases www and Telnet users.

6. Conclusion

we find that Telnet and Ftp have the least and the greatest activity factors respectively. So Telnet and Ftp can occupy resources the least and the most respectively. In other words simultaneous Telnet users might be increased more than the other Internet users and simultaneous Ftp users can be increased less than the other Internet users. Besides the more the data rates are, the less the probability of the simultaneous users are. On the other hand we find that we can use a Radio Resource Management module to enhance the QoS metrics (increase bit rate and decrease BER and delay). We can increase the number of packet traffic users in comparison with circuit switch voice traffic users because they have lower Erlang traffic figures.

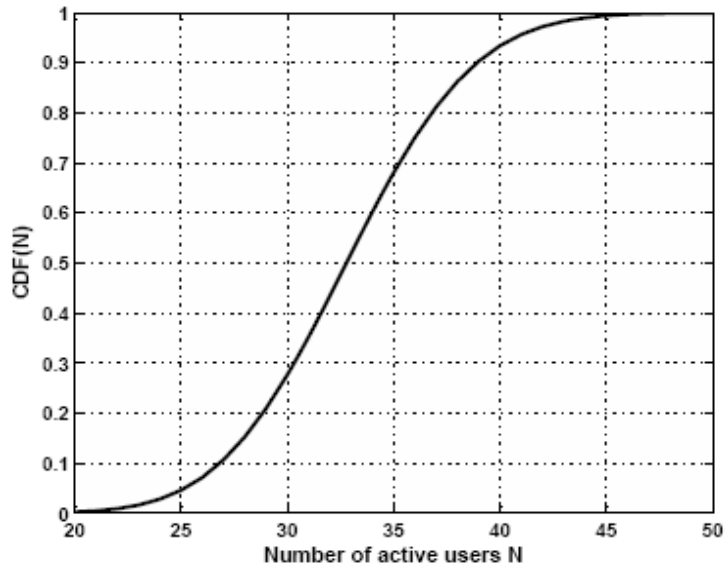


Figure 5. Distribution of active data users for $M=100$, $\rho = 0.5$

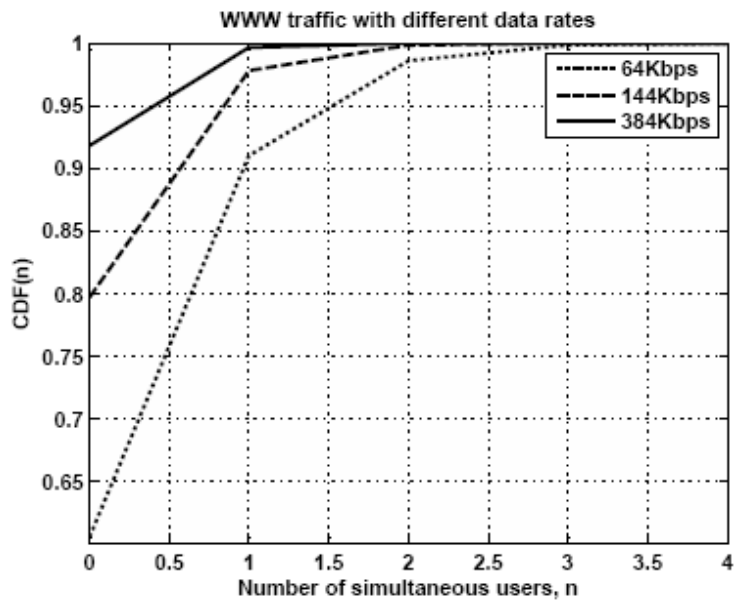


Figure 6. Distribution of the number of simultaneous data users for $M=100$

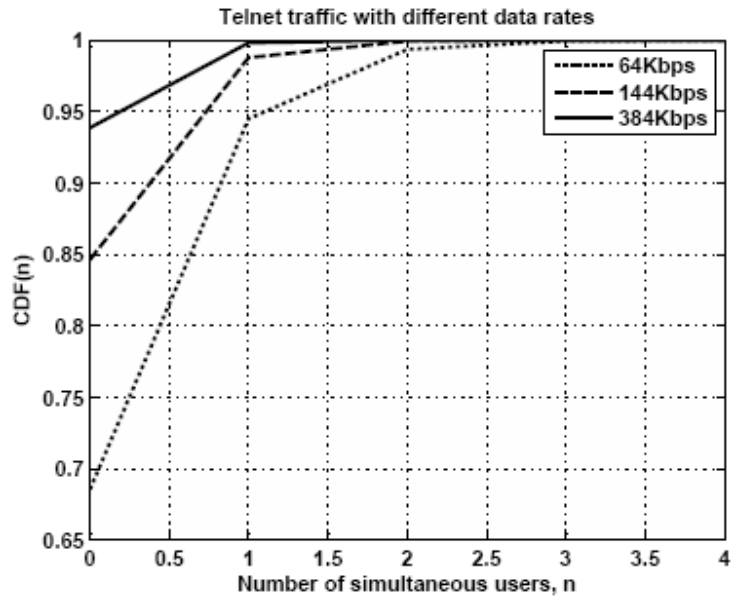


Figure 7. Distribution of the number of simultaneous data users for M=100

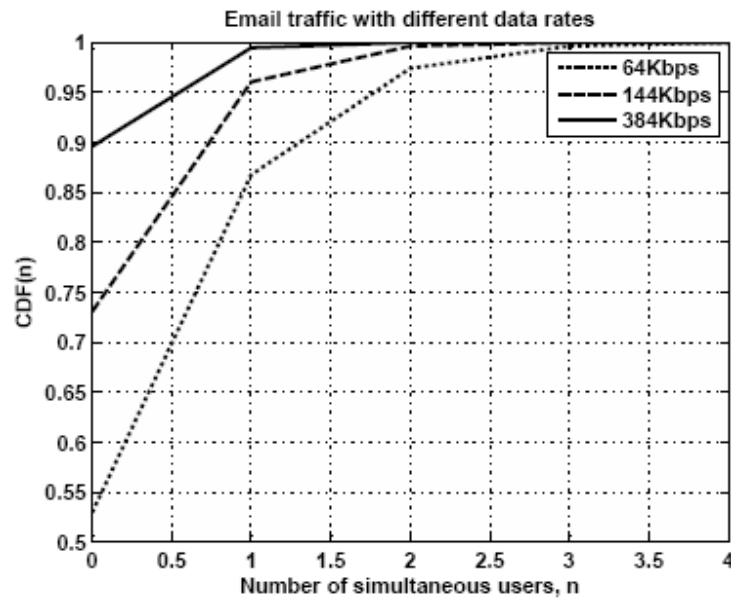


Figure 8. Distribution of the number of simultaneous data users for M=100

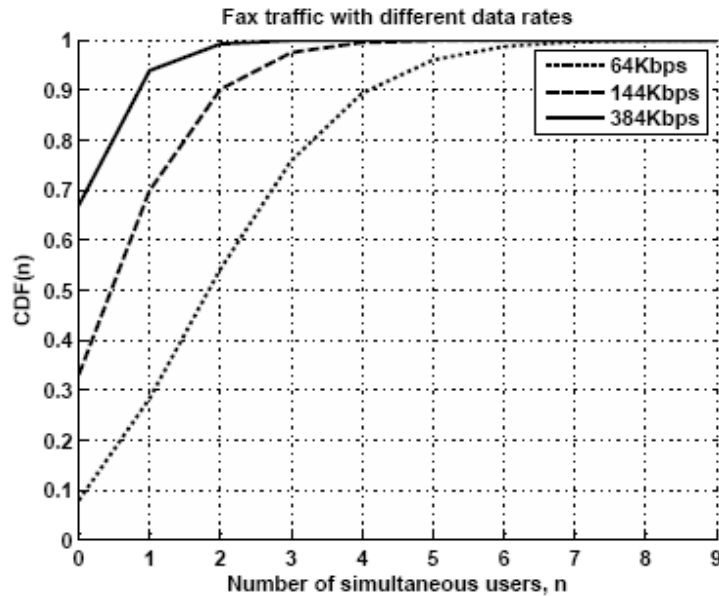


Figure 9. Distribution of the number of simultaneous data users for M=100

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