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Traffic Capacity of UMTS System in Reverse Link

Jahangir Dadkhah Chimeh · Mohammad Hakkak ·
Paeiz Azmi · Hamidreza Bakhshi

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Abstract UMTS can be connected to data networks. So Internet traffic services such as WWW browsing, email, ftp, SMTP, etc 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 traffic service there is some unused (OFF) times between their transmitted packets which we may use them for other services to increase the traffic capacity of the network. In this paper we propose a new method on the Internet traffic evaluation in an interference limited UMTS-WCDMA system. In this method first we calculate ON and OFF time durations of the above traffic services based on the model presented in 3GPP then their activity factors. Secondly we introduce new capacity curves such as The Number of Web browsing users/Email users/ftp users/Telnet users and Fax users versus voice users. These are applicable in traffic planning for wireless systems.

Keywords UMTS · Internet traffic · Multiuser system

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

J. Dadkhah Chimeh (✉)
Science and Research Branch, Islamic Azad University, Tehran, Iran
e-mail: dadkhah@sr.iau.ac.ir

M. Hakkak · P. Azmi
Tarbiat Modares University, Tehran, Iran
e-mail: M.hakkak@modares.ac.ir

P. Azmi
e-mail: Pazmi@modares.ac.ir

H. Bakhshi
Shahed University, Tehran, Iran
e-mail: bakhshi@shahed.ac.ir

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 services. The model presented in [2] like the 3GPP model presented in [3] uses a multi-layer model for describing sources and in the lowest layer uses Weibull and Pareto distributions for ON and OFF durations of Internet traffic respectively. 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 [6]. The traffic generated by data applications is inherently bursty and asymmetric, with higher data rates in the downlink than in the uplink [7]. Tripathi and Capone in [8] and [9] also verified the performance of different types of services in IS-2000 system. Besides Soldani in [4,10] 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 that bursts. Activity factor gives a criterion for this burstiness, besides it depends on traffic type and model. In this paper we use the burstiness property of Internet traffic based on the models presented in 3GPP. Then we calculate data traffic activity factor and using data rates and Eb/No QoS metrics evaluate the number of users for each of the traffic services. 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. In the Sect. 3 based on the presented models and their characteristics we pay attention to activity factor calculations of some data traffic services. Besides, we know WCDMA as a multi-user technique. So UMTS is an interference limited system and we can define soft capacity in that. In Sect. 4 we pay attention to system capacity calculation for voice and different data users [11]. Here we use the activity factor to calculate the number of users. Simulation 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 [12–14]. 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 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

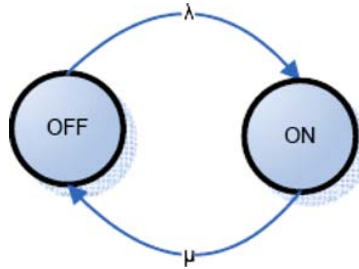


Fig. 1 ON-OFF models

silent periods being generated by an exponential distribution with mean values for active and silence periods both equal to 3 s and independent on the up and downlink [3].

In the real time systems we don't need to 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 and that the ON duration is effectively characterized by heavy-tailed models. The OFF duration is determined by the user's think time which is also modeled by the heavy tailed model [15]. 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.

For circuit switched data services [16], the traffic model should be a constant bit rate model, with 100% of activity [2,3]. 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. A connection may consist of one or more sessions. Figure 2 depicts a typical non-real time 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. The time between two consecutive packet call/packet is called packet call/packet interarrival time. For example in a WWW browsing session a packet call corresponds to the downloading of a WWW document.

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 a data call 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.

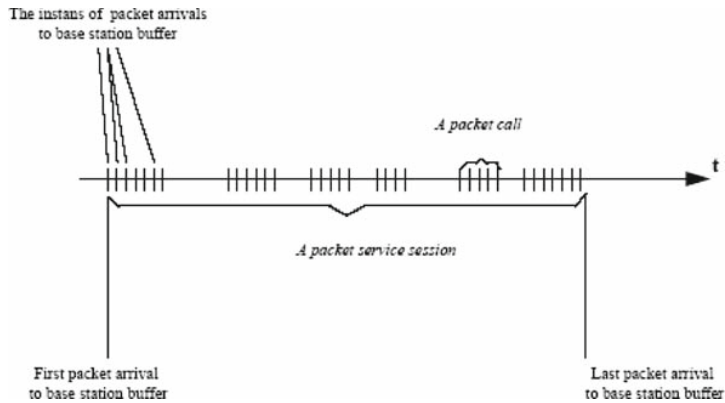


Fig. 2 Packet traffic model [3]

- 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.,

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

- 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.,

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

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

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

- 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.,

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

- Size of a datagram, S_d .

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

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

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

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

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

$$\mu_{N_d} \cdot \mu_x = 25 \times 480 = 12 \text{ kB.}$$

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

3 Activity Factor

A traffic source usually alternates active and silent periods. Indeed activity factor represents the fraction of time that the source is generating traffic. In the OFF (Idle) time the source doesn't generate any packet.

The random characteristic of traffic activity is assumed to be represented by the mean of traffic activity, called the traffic activity factor. The activity factor of voice or data traffic, α , is defined as the probability that the state is ON and can be given as

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

We now calculate activity factor α for traffic types Telnet, WWW, ftp, e-mail and fax based on the Table 2 for a 384 Kbps data traffic. The results are in Table 3. In this calculation we use third column of the Table 2 and show this calculation for the Web browsing traffic as follows:

According to that table we have five packet calls per a session, inter arrival time of 120 s between packet calls, 25 packets per packet call, the average packet size of 480 and inter arrival time of 0.067 s between packets. We first calculate the whole OFF time in WWW session. The average time of a packet is $480 \times 8/384000 = 0.01$ and the OFF time between two consecutive packets is $0.67 - 0.01 = 0.057$ and OFF time in a packet call is $(25 - 1) \times 0.057 = 1.37$. The whole OFF time between packet calls is $5 \times 1.37 = 6.84$. So the whole OFF time in a session is $4 \times 120 + 6.84 = 486.84$ s.

Now we calculate ON time of a WWW session in a packet call as $25 \times 0.01 = 0.25$ s. So the whole ON time in a WWW session is $5 \times 0.25 = 1.25$ s.

Finally the activity factor α is calculated as $\alpha = 1.25/(1.25 + 486.84) = 0.00256$. We have also calculated this parameter for other services which are listed in Table 3.

Table 1 Characteristics of datagram information types [3]

Packet based information types	Average number of packet calls within a session	Average reading time between packet calls [s]	Average amount of packets within a packet call []	Average interarrival time between packets [s]	Parameters for packet size distribution
WWW surfing	5	412	25	0.5	$K=81.5$, $\alpha=1.1$
UDD 8 kbit/s	5	412	25	0.25	$K=81.5$, $\alpha=1.1$
WWW surfing	5	412	25	0.0625	$K=81.5$, $\alpha=1.1$
UDD 32 kbit/s	5	412	25	0.0277	$K=81.5$, $\alpha=1.1$
WWW surfing	5	412	25	0.01040	$K=81.5$, $\alpha=1.1$
UDD 64 kbit/s	5	412	25	0.00195	$K=81.5$, $\alpha=1.1$
WWW surfing	5	412	25		$K=81.5$, $\alpha=1.1$
UDD 144 kbit/s	5	412	25		$K=81.5$, $\alpha=1.1$
WWW surfing	5	412	25		$K=81.5$, $\alpha=1.1$
UDD 384 kbit/s	5	412	25		$K=81.5$, $\alpha=1.1$
WWW surfing	5	412	25		$K=81.5$, $\alpha=1.1$
UDD 2048 kbit/s	5	412	25		$K=81.5$, $\alpha=1.1$

Table 2 Traffic models and their characteristics [8]

Model parameter	Telnet	WWW	ftp	E-mail	Fax
No. of packet calls per session	Geometric (mean of 114)	Geometric (mean of 5)	1	Geometric (mean of 2)	Geometric (mean of 3)
Inter arrival time or reading time between packet calls (s)	1	Geometric (mean of 120)	–	Pareto (mean of 90)	Weibull (mean of 30)
No. of packets per packet call	1	Pareto (mean of 25)	Pareto (mean of 62)	Weibull distributed packet cal size (mean of 15)	Pareto distributed packet cal size (mean of 15)
Packet size(bytes)	Geometric (mean of 90)	480	480	480	480
Inter-arrival time between packets(s)	Geometric (mean of 1 sec)	Geometric (mean of 0.067)	Geometric (mean of 0.067)	Geometric (mean of 0.067)	Geometric (mean of 0.27)

Table 3 Activity Factors based on the Table 2

	ON duration (s)	OFF duration (s)	Activity factor
Telnet	0.216	0.998	0.0019
WWW	1.25	486.84	0.00256
ftp	0.61	3.477	0.149
E-mail	0.3	90.787	0.0033
Fax	1.11	88.1	0.012

4 Capacity

For i th user in a multi-user flat fading channel we express E_b/N_o as signal to interference and noise ratio according to:

$$\left(\frac{E_b}{N_o}\right)_i = \frac{W s_i}{R_i n} \quad (11)$$

where W is the total bandwidth, R_i the bit rate of the i th user, s_i is its power at the receiver and n is total noise and interference at the receiver [18].

We assume there are N user groups in reverse link. A user group consists of some users that have the same QoS parameters. One group is for voice service, and the other groups are for various data services for examples ftp and web browsing and etc. We define the power received by the BS as $S_{v,i}$ for the i th voice user in the voice user group and $S_{d,j,n}$ for the n th user in the data user group j ($j = 1, 2, \dots, N - 1$), and define the information data rates as R_v for the voice user group and $R_{d,j}$ for the data user group j . For the i th voice user, the received E_b/N_o is represented as follows [17].

$$\left(\frac{E_b}{N_0}\right)_{v,i} \cong \frac{W}{R_v} \cdot \frac{S_v}{\sum_{k=1, k \neq i}^{N_v} \alpha_v S_{v,k} + \sum_{j=1}^{N-1} \sum_{n=1}^{N_{d,j}} \alpha_{d_j} S_{d_{j,n}} + I + \eta_k W} \quad (12)$$

where W is the bandwidth; N_v and N_{d_j} represent the number of users in the voice user group and the data user group j in a sector, respectively; α_v is the voice activity factor; α_{d_j} is the data user group j activity factor; I is the other cell interference; and η_0 is the level of the background noise power spectral density. For the simplicity of the analysis, we assume first that BS uses three ideal directional antennas, and perfect power control. Besides, path loss is proportional to $10^{\zeta/10} r^{-4}$ (ζ is considered to be a Gaussian random variable and r is user distance from BS) and fast fading not to affect the power control.

According to the perfect power control, we have $S_{v,k} = S_v$ and $S_{d_j,n} = S_{d_j}$ for all k and n . From the fact that the background noise η_0 can be negligible compared to the user interference, (12) is approximately modified to

$$\left(\frac{E_b}{N_0}\right)_v \cong \frac{W}{R_v} \cdot \frac{S_v}{\alpha_v (N_v - 1) S_v + \sum_{j=1}^{N-1} N_{d_j} \alpha_{d_j} S_{d_j} + I} \quad (13)$$

Similarly, the received E_b/N_0 for i th data user in the data user group j is

$$\left(\frac{E_b}{N_0}\right)_{d_j,i} \cong \frac{W}{R_{d_j}} \cdot \frac{S_{d_j,i}}{\sum_{k=1}^{N_v} \alpha_v S_{v,k} + \sum_{j=1, j \neq i}^{N-1} \sum_{n=1}^{N_{d_j}} \alpha_{d_j} S_{d_{j,n}} + I + \eta_k W} \quad (14)$$

for $i = 1, 2, \dots, N - 1$

According to the perfect power control, we have $S_{v,k} = S_v$, $S_{d_j,i} = S_{d_j}$ and $S_{d_j,n} = S_{d_j}$ for all k and n . From the fact that the background noise η_0 can be negligible compared to the user interference, (14) is approximately modified to

$$\left(\frac{E_b}{N_0}\right)_{d_j} \cong \frac{W}{R_{d_j}} \cdot \frac{S_{d_j}}{\alpha_v N_v S_v + (N_{d_i} - 1) \alpha_{d_i} S_{d_i} + \sum_{j=1, i \neq j}^{N-1} N_{d_j} \alpha_{d_j} S_{d_j} + I} \quad (15)$$

After some manipulations on (13) and (15) we have

$$\{(SIR)_v^{-1} + \alpha_v\} S_v = \{(SIR)_{d_i}^{-1} + \alpha_{d_i}\} S_{d_i} \quad (16)$$

where

$$(SIR)_v = \frac{R_v}{W} \cdot \left(\frac{E_b}{N_0}\right)_v \quad \text{and} \quad (SIR)_{d_j} = \frac{R_{d_j}}{W} \cdot \left(\frac{E_b}{N_0}\right)_{d_j} \quad (17)$$

To satisfy BER requirement which is one of the QoS factors of various services for all the user groups, the received E_b/N_0 should be greater than the required E_b/N_0 i.e.

$$\left(\frac{E_b}{N_0}\right)_v \geq \left(\frac{E_b}{N_0}\right)_{v,req} \quad \text{and} \quad \left(\frac{E_b}{N_0}\right)_{d_j} \geq \left(\frac{E_b}{N_0}\right)_{d_j,req} \quad (18)$$

Besides, to satisfy the information data rate requirement for all user groups, the following relations should be satisfied:

$$R_v \geq R_{v,req}, R_{d_j} \geq R_{d_i,req} \quad (19)$$

After replacing (13) and (15) in (18) and (19) and some manipulation we have

$$\alpha_v(N_v - 1)S_v + \sum_{j=1}^{N-1} N_{d_j} \alpha_{d_j} S_{d_j} + I \leq \frac{W}{R_{v_{req}}} \left(\frac{E_b}{N_0} \right)^{-1}_{v_{req}} S_v = (SIR)^{-1}_{v_{req}} S_v \quad (20)$$

where $N_v, N_{d_1}, N_{d_2}, \dots, N_{d_{N-1}}$ are number of users.

We can reduce Eq. 20 to

$$\alpha_v \frac{N_v}{(SIR)^{-1}_{v_{req}} + \alpha_v} + \sum_{j=1}^{N-1} \frac{N_{d_j}}{(SIR)^{-1}_{d_{j_{req}}} + \alpha_{d_j}} \leq 1 - z \quad (21)$$

where

$$z = \frac{I}{S_v} \frac{1}{(SIR)^{-1}_{v_{req}} + \alpha_v} = \frac{I}{S_{d_j}} \frac{\alpha_{d_j}}{(SIR)^{-1}_{d_{j_{req}}} + \alpha_{d_j}} \quad (22)$$

For a single cell system, the other cell interference has no effect on the capacity, and the term z of (21) is set to zero. Therefore, (21) is simplified to the following equation for a single cell case:

$$\gamma_v N_v + \sum_{j=1}^{N-1} \gamma_{d_j} N_{d_j} \leq 1 \quad (23)$$

where

$$\gamma_v = \frac{\alpha_v}{(SIR)^{-1}_{v_{req}} + \alpha_v}, \quad \gamma_{d_j} = \frac{\alpha_{d_j}}{(SIR)^{-1}_{d_{j_{req}}} + \alpha_{d_j}} \quad (24)$$

Equation 23 specifies a capacity plane in the N dimensional space.

All points $N_v, N_{d_1}, N_{d_2}, \dots, N_{d_{N-1}}$ under the hyper plane represent possible numbers of supportable users in voice and data user groups in a sector. In (23) the total resource amount of the system, the resource amount used by one voice user, and the resource amount used by one data user in the group j correspond to 1, γ_v and γ_{d_j} respectively. Equation 23 also means that the resources used by users should not exceed total system resource.

5 Simulation

We consider a UMTS system with the parameters shown in the Table 4. We also consider the data rate 384 kbps for all the data service types (this corresponds to a pedestrian mobile

Table 4 Parameters of UMTS system

Item	Symbol	Value
Bandwidth	W	5 Mhz
Voice activity factor	α	≤ 0.5
Information data rate for the voice group	R_v	9.6 Kbps
Information data rate for the data group	R_{d_i}	64, 144, 384, 2048 Kbps
Quality requirement for the voice group	$\left(\frac{E_b}{N_0} \right)^{-1}_{v_{req}}$	5(7) dB
Quality requirement for the data group	$\left(\frac{E_b}{N_0} \right)^{-1}_{d_{i,req}}$	12, 10, 5 dB

user) and 9.6kbps for voice service and use activity factors from Table 3. We also consider six traffic types Telnet, www, e-mail, ftp, fax and voice and UMTS system parameters as is shown in Table 4.

After simulation by MATLAB we find Figs. 3, 4 and 5. In Fig. 3 we found the number of the data users respective to the voice users for each of the traffic types Telnet, www, email,

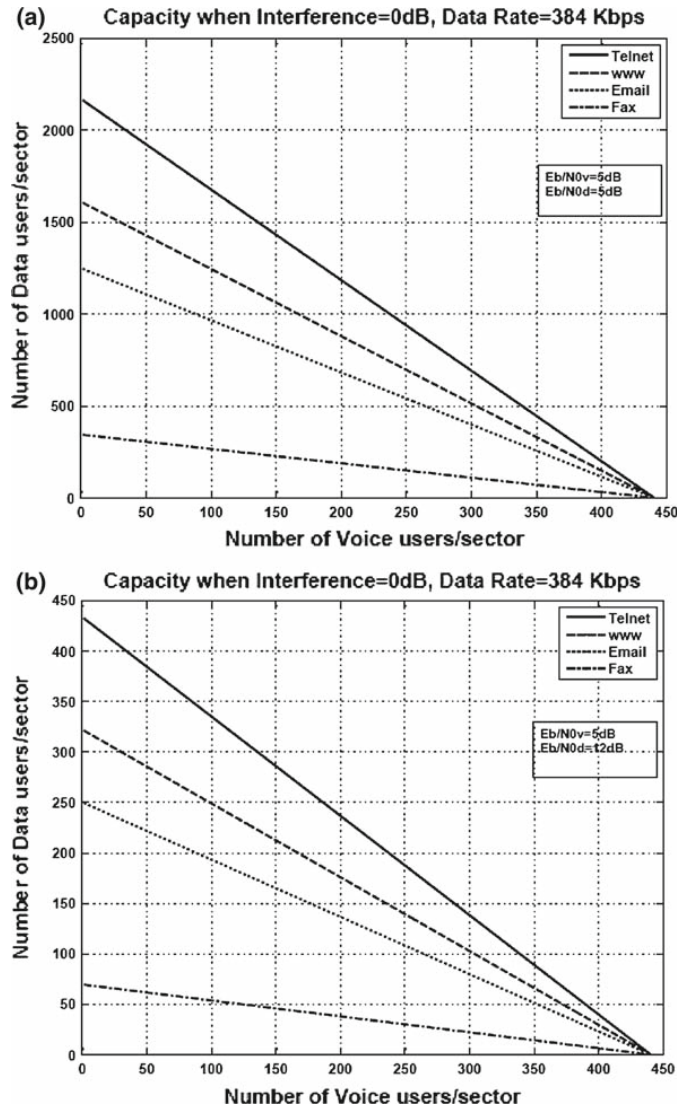


Fig. 3 Capacity for voice and one groups of data with $R_v = 9.6\text{kbps}$ in a single cell case and voice activity factor = 0.375

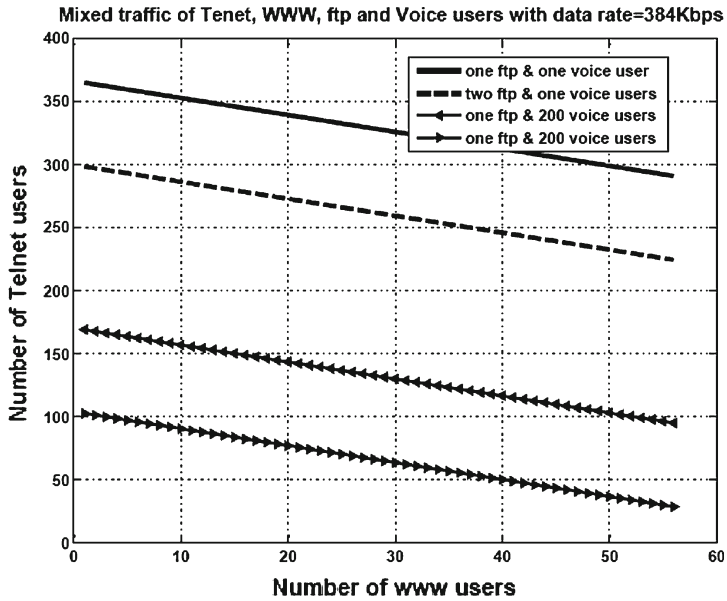


Fig. 4 Capacity for one voice group of $R_v = 9.6$ kbps together with three data user groups in a single cell case with $E_b/N_0 = 12$ dB for data and voice

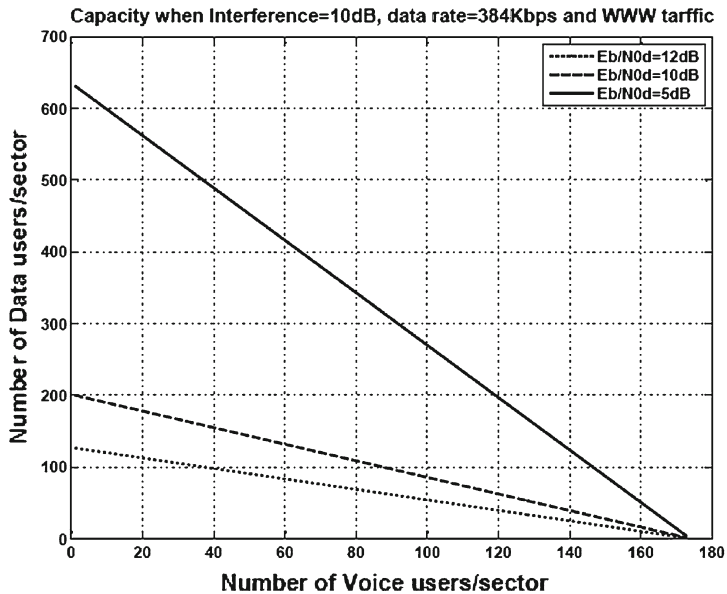


Fig. 5 Capacity for voice user when $R_v = 9.6$ kbps and one data users when $R_{d_i} = 384$ kbps in multiple cell case and voice activity factor=0.375

fax which each one is applied distinctly along side with voice in a single cell scenario. As we see in the Fig. 3 the smaller the activity factors, the larger the number of the users. In addition to that, we see the slope of the Telnet traffic line is greater than the www traffic line and the slope of the e-mail traffic line is greater than the fax traffic line, so for a specific voice users, Telnet users can be increased more than the www users and e-mail users can be increased more than fax users. Besides if for example we select the Fig. 3b and assume there are 200 voice users in the system, we find that we can have at most 240 Telnet users, 180 www users, 140 e-mail users and 40 fax users in the system.

Then we found the number of www users with three kinds of $E_b/N_o = 5, 10$ and 12 dB versus the number of voice users in a multi-cell scenario and other cell interference = 10 dB as are shown in Fig. 5. Figure 5 shows the capacity reduction as a result of the other cell interference.

Finally we applied all the Telnet, www, ftp and voice traffic service types in the network and plotted the number of the Telnet users versus the number of the www users in four cases; one ftp user and one voice user, two ftp users and one voice user, one ftp user and 200 voice users, two ftp users and 200 voice users as are shown in Fig. 4. Figure 4 depicts the resource distribution among the service users' types.

6 Conclusion

In this paper by involving activity factors in SNIR formula we introduced new curves and calculated the effects of the different data traffic types on the users' numbers in a slow fading multi-user channel. Besides, we showed the smaller the activity factors the larger the number of the users, because we can use the spaces between the packets for another traffic type to increase the number of the users in the packet traffic mode. We also showed that for a specific voice users Telnet users can be increased more than www users and e-mail users can be increased more than fax users.

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Author Biographies



Jahangir Dadkhah Chimeh received his B.Sc. from Sharif University of Technology in 1988 and his M.S. from K.N.T. University of Technology both in Tehran/Iran in 1994. He is now a Telecommunication Engineering Ph.D. student in Islamic Azad University Science and Research branch, Tehran, Iran. Besides, he is a faculty member of Iran Telecommunication Research Center (ITRC), Tehran. He was project manager of 3G mobile pilot project in 2005 and has some papers and a book in the mobile field.



Mohammad Hakkak obtained the M.S. and Ph.D. degrees in Electrical Engineering from the University of Illinois, Champaign, Urbana, USA in 1971. Since then, he served as a faculty member and Head of Electrical Engineering Department at Baghdad University, Senior Researcher and Head of Transmission Department at the Iran Telecom Research Center and a faculty member of Tarbiat Modares University in Tehran where he is now a Professor of Electrical Engineering. He has taught and researched microwave antennas, radiowave propagation, passive microwave components, satellite communication, small satellite technology, earth station technology and mobile communications. Prof. Hakkak has published more than 140 journal and conference papers, authored and co-authored six books. He also organized many national and international conferences, particularly the series of the International Symposium on Telecommunications (IST) in Iran. He is now a Fellow of IEE (UK), Senior Member of IEEE (USA) and Senior Member of IAEEE (Iran).



Paciz Azmi was born in Tehran—Iran, on April 17, 1974. He received the B.Sc., M.Sc., and Ph.D. degrees in Electrical Engineering from Sharif University of Technology (SUT), Tehran—Iran, in 1996, 1998, and 2002, respectively. Since September 2002, he has been with the Electrical and Computer Engineering Department of Tarbiat Modares University, Tehran—Iran, where he became an associate professor on January 2006. From 1999 to 2001, he was with the Advanced Communication Science Research Laboratory, Iran Telecommunication Research Center (ITRC), Tehran, Iran. From 2002 to 2005, he was with the Signal Processing Research Group at ITRC. His current research interests include modulation and coding techniques, digital signal processing, and wireless and optical CDMA communications.



Hamidreza Bakhshi was born in Tehran, Iran on April 25, 1971. He received the B.Sc. Degree in Electrical Engineering from Tehran University, Iran in 1992, the M.Sc. and Ph.D. Degree in Electrical Engineering from Tarbiat Modarres University, Iran in 1995 and 2001 respectively. Since 2001, he has been as an Assistant Professor of Electrical Engineering at Shahed University, Tehran, Iran. His research interests include wireless communications, multiuser detection, and smart antennas.