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Mitigating multimedia call failure in CDMA networks

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ABSTRACT: *It is very challenging to effectively mitigate call failures while maintaining the quality of service (qos) in the mobile networks. Interestingly, more efforts have been channeled towards voice calls management. In this paper, an efficient scheme to mitigate multimedia call failure in a network is proposed. To achieve this, the incoming calls (multimedia) are divided into real time and non-real time calls. Priority is given to the real time multimedia calls. Statistical approach using the continuous-time single dimensional birth–death process (Markov chain) was adopted to model the system. Consequently, a mathematical model known as success probability model of multimedia calls (SPMMC) was developed and simulation environment built to assess the performance of the proposed scheme.*

Keywords –*Multimedia calls, Call Failure, Resource management, Real time Call, Non-Real time Call*

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I. INTRODUCTION

The challenges for achieving optimum spectral efficiency and high data rate in wireless cellular communication networks is increased by the wireless communication environment which is characterized by dynamic channels, high influence of interference, bandwidth shortage and strong demand for quality of service support [1]. In order to support various integrated services like multimedia, with certain quality of service requirements in these wireless networks, the study of Radio Resource Management (RRM), Radio Resource Provisioning (RRP), and Mobility Management (MM) is useful [2]. The radio spectrum is characterised with scarce resources, hence, there is need for effective radio resource management (RRM) for the management and allocation of these resources. Multimedia is a term that describes multiple forms of information, including audio, video, graphics, animation, images, text, etc. The best examples are continuous media such as animation, audio and video that are time-based, that is., each audio sample or video frame has a timestamp associated with it, representing its presentation time. Multimedia data has to be presented in a continuous fashion, in accordance with their associated timestamp. Multimedia is usually considered to be real time or non-real time. The real time is usually delay sensitive while the non-real time is delay tolerable. The performance of a system with given physical resource (for example., given bandwidth of radio spectrum) depends heavily on the resource management schemes including the multiple access techniques, the call admission control policies, and the congestion control schemes. In other to carry multimedia traffic, high capacity of system is a basic necessity [4]. The wide band code-division multiple-access (WD-CDMA) technique is widely used for the third-generation (3G) mobile communications systems because of its advantage in capacity. In addition, CDMA has other merits, for instance, a simple frequency planning. On the other hand, a remarkable characteristic in wireless multimedia communications is the traffic (load) asymmetry between uplink (from mobile to base) and downlink (from base to mobile). To cope with this situation, some systems assign the total available bandwidth (i.e., the radio resources) in a cell to the uplink and the downlink asymmetrically. An example is the CDMA systems with time-division duplex mode (TDD systems) that adopts asymmetric time slot allocation between uplink and downlink. Note that TDD is one of the strongest candidates for the radio transmission technology (RTT) of the International Mobile Telecommunications-2000 (IMT-2000) systems. Wireless multimedia services should be efficient in the environments with multiple classes of traffic, where the traffic asymmetry pattern of a class differs from those of other classes. CDMA systems are interference-limited systems. Thus, the capacity of a cell (that is, the total resource in the cell) varies with the loading of the home and neighboring cells because the co-channel interference changes according to the loading. On the other hand, for guaranteeing adequate call quality, the signal to interference ratio (SIR) of a call should be maintained to be higher than a predefined value. To accomplish this objective, a call request is admitted only when even though it is accepted, the SIR of an ongoing call is expected to be not smaller than a threshold value [1]. With the increased demand for wireless communication systems, a guaranteed quality of service (QoS) is required in a satisfactory manner, to manage the incoming new calls and handoff calls more efficiently. QoS provisioning in wireless networks is a challenging problem due to the scarcity of wireless resources, i.e. radio channels, and the mobility of users. Call admission control (CAC) is a fundamental mechanism used for QoS provisioning in a network. It restricts the access to the network based on resource availability in order to prevent network congestion and service degradation for already supported users. A new call request is accepted if there are enough idle resources to meet the QoS requirements of the new call without violating the QoS for already accepted calls. Admitting too many users usually results in a situation where the mutual interference between the connections degrades the QoS for the new user as well for the ongoing connections. Therefore, admission control plays a very important role in providing the user with the requested QoS as well as making an efficient use of the available capacity and preventing the system from an outage situation due to overloading [5].

It has been pointed out that the rate of growth in the mobile users' population most times exceeds the rate at which the operator of the network can deploy additional communication facilities to meet the demand. As the results of this, network subscribers oftentimes have their calls (voice or multimedia) prematurely terminated while others find it difficult to initiate a call. This is very frustrating especially for real time multimedia calls.

II. THE REVIEW OF RELATED LITERATURE

2.1 MULTIMEDIA CALLS

Multimedia is the integration of information that may be presented by several media types such as audio, text, video and still images. Multimedia is a term that describes multiple forms of information, including audio, video, graphics, animation, images, text, etc. The best examples are continuous media such as animation, audio and video that are time-based, for example, each audio sample or video frame has a timestamp associated with it, representing its presentation time. Multimedia data has to be presented in a continuous fashion, in accordance with their associated timestamp. For example, video is typically rendered at 30 frames per second to give the viewers the illusion of smooth motion. As a result, multimedia applications typically have the real-time constraint, for example, media data has to be delivered and rendered in real time [12]

2.1.1 Non-Real Time Multimedia

The non-real time multimedia is characterized by the fact that they can tolerate delays. This implies that the non-real time multimedia is time independent. An example of a non-real time multimedia is the multimedia massage (MMS). This class is similar to real time, except that jitter and delay are not so important but packet loss is much more important [13].

2.1.2 Real Time Multimedia

Real time multimedia refers to applications in which multimedia data has to be delivered and rendered in real time. Unlike the non-real time multimedia, the real time multimedia is time dependent. It is delay sensitive. In real time multimedia application, the information must be delivered immediately.

Real-time multimedia can be broadly classified into interactive multimedia and streaming media. Interactive multimedia applications include Internet telephony, Internet video-conferencing, Internet collaboration, Internet gaming, etc. In interactive multimedia applications, the delay constraint is very stringent in order to achieve interactivity. For example, in Internet telephony, human beings can only tolerate a latency of about 250 milliseconds. This imposes an extremely challenging problem for interactive multimedia applications over the Internet that provides only the best effort service. Over the years, great efforts have been made to facilitate the development of interactive multimedia applications over the Internet. For example, Microsoft Research's Conference XP Research Platform 1 supports the development of real-time collaboration and videoconferencing applications by delivering high-quality, low-latency audio and video over broadband connections. The second class of networked multimedia technology is streaming media. Streaming media technology enables the real time or on demand distribution of audio, video and multimedia on the Internet. Streaming media is the simultaneous transfer of digital media so that it is received as a continuous real-time stream. Streamed data is transmitted by a server application and received and rendered in real-time by client applications. These client applications can start playing back audio and video as soon as enough data has been received and stored in the receiver's buffer. There could be up to a few seconds of startup delay, i.e., the delay between when the server starts streaming the data and when the client starts the playback. Some of the popular streaming media products are Microsoft's Windows Media Player and RealNetworks's RealPlayer for Internet streaming, and Packet Video's embedded media player for wireless streaming to embedded devices such as the next generation multimedia applications. In this class of applications, throughput, delay, jitter and packet loss are considerable. The throughput is the effective number of data units transported per unit time (e.g., bits/second), this parameter is usually specified as a “bandwidth guarantee”. The delay is the time interval between the departures of data from the source to the arrival at the destination and connection delay could be included in it [13]. The jitter is usually referred to as “delay variation”. The loss is the percentage of data units that did not make it to the destination in a specific time interval. It is usually represented as a probability of loss.

2.2 Resource Allocation

Call admission control scheme serves to get high quality and maximum resource utilisation in cellular communication. Different generations of cellular system successfully working by the support of efficient call admission control scheme. In cellular communication system the modem is designed to utilize the spectrum in efficient manner. The cellular communication system contains base stations. That base stations are interconnected through communication network. The geographical area has divided into small regions. Those regions are known as cells. Each cell contains a base station. Using that base station mobile user can communicate [4]. The mobile user can start communication by making connection with base station. The base station provides channel with required signal strength to the mobile user. Using channel with required signal strength; the mobile user can perform communication. The new call and handoff calls will share and utilize resources of base station to provide service to the mobile users. The resources are bandwidth, time and power. The resource type varies based on the access technique. The call will be blocked if the required resources are not available. The user may release the resources which they have used by handoff or by call termination. The quality of service of a particular cell is declared based on the new call blocking probability and handoff call dropping probability. The higher priority has given to handoff calls to avoid call dropping during communication session.

2.3 Review of Some Existing CAC Schemes

In [17], an algorithm for assigning of the channels is proposed. This assigns the channels adaptively. At the event that forced termination probability exceeds a threshold preset by the system, the Guard channel number is increased. This results in decrease in the forced termination probability.

Another method in determining the number of guard channel is by the use of neighboring base stations. The number of mobile stations in pre-handover zone (PHZ) was determined by each base station periodically. The base stations inform each other with information related that PHZ. When the base station gets the information about the number of mobile stations in PHZ, it then reserves that amount of Guard channels for handover calls. This scheme accepts a new call only if no handoff calls are queued in the queue where handoff calls are kept and the total number of free channels is greater than the number of Guard channels. A static reservation typically results in poor resource utilization.

The cutoff priority scheme is another way of handling the Guard channel as stated IN [8]. In this method, a portion of the channel is reserved for handoff calls. Whenever a channel is released, it is returned to the common poll of channels.

[5] proposed an analytical approach to model and analyze the performance of Global System for Mobile Phones by developing a new Handoff Scheme (M+G+Q Scheme). This scheme integrates the handoff queue into the M+G scheme. This helps to further minimize handoff failure. This helps to further minimize handoff failure. Using simulation in MatLab, the Scheme was evaluated in terms of handoff failure probability. This approach was applied in the analysis of mobility management and connection performance with emphasis on the prioritized. The arrival rates of originating and handoff calls were assumed to be Poisson while time variables such as call holding time, cell residence time, channel holding time, registration area (RA) residence time, and inter-service time were assumed to be exponentially distributed.

M+G+Q Algorithm

The notable feature of this scheme is the integration of buffer (queue) for handoff request in the M+G scheme. The scheme also considers signal strength and channel availability and for handoff decision. At the arrival of the request, the scheme performs a check on the signal power of the request. This is denoted by the factor α. The value for this factor ranges from zero to 1 ($0 < \alpha \leq 1$). As the value of α increases, it implies that the signal strength is stronger. For the request to be processed initially, this factor must be at acceptable level (0.7). If the signal power of the request is greater than or equals to 0.7 (threshold), the request is admitted for further checks. After the signal power has been established, the scheme then determines if there are free channels in the shared channel. It assigns any channel available to the request on the basis of first in first out irrespective of the type of request. This is because both handoff and originating calls are treated equally in these shared channels. If there are no channels in the shared channels, the next action is to determine the type of request. The originating call request is refused when there is no free channel in the shared channels. For the handoff request, the scheme checks for a free channel in the reserved channels and assigns it. If the reserved channels are congested, the handoff requests are put in the FIFO handoff queue. Any released channel in the poll is assigned to the first handoff request in the queue. If the queue waiting time exceed the maximum allowable waiting time, the handoff request is denied or reattempted.

III. SYSTEM MODEL PARAMETERS AND ASSUMPTIONS

The following assumptions are adopted in this system model.

1. Both the new real time and non-real time multimedia calls rates in the cell form a Poisson process (Markovian) with mean values of λ_{RT} and λ_{NRT} respectively. Therefore, total call rate is $\lambda = \lambda_{RT} + \lambda_{NRT}$
2. Real time and non-real time multimedia calls completion time are exponentially distributed with mean rates of μ_{RT} and μ_{NRT} respectively. Therefore, the effective service rate is $\mu = \mu_{RT} + \mu_{NRT}$

The system cell is made of a total of C channels. Priority is given to the hot spot (real time) multimedia. This is because mobile users are more sensitive to the hot spot (real time) multimedia calls than non-hot spot (non-real time) multimedia calls. The given priority will be implemented using the Guard channel method. Out of the C channels of the call, R channels are reserved exclusively for real time while the remaining R = C-M channels are shared by real time and non-real time multimedia calls.

The system model is in fig 1.

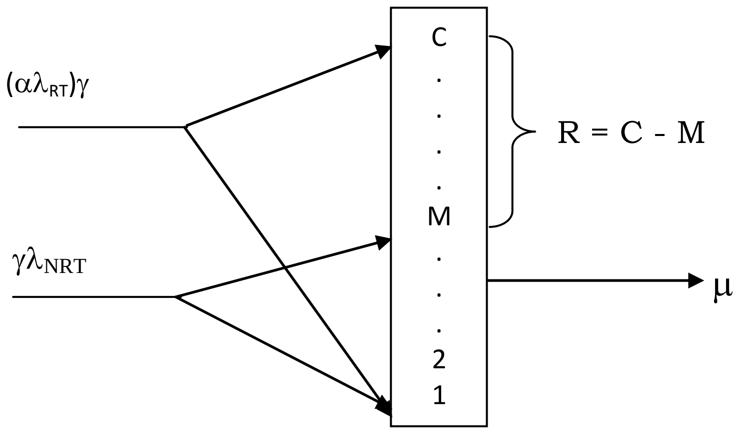

Figure 1: System model

$$\lambda_{RT} = \gamma_{RT}\lambda_{RT} \quad - \; - \; - \; - \; - \; - \; - \; - \; - \; - \tag{1}$$

$$\lambda_{n} = \gamma_{NRT}\lambda_{NRT} \quad - \; - \; - \; - \; - \; - \; - \; - \; - \; - \tag{2}$$

It is assumed that γ_{NRT} and γ_{RT} are the same and denoted as γ. Therefore

$$\lambda_{N} = \gamma\lambda_{RT} \quad - \; - \; - \; - \; - \; - \; - \; - \; - \; - \tag{3}$$

$$\lambda_{n} = \gamma\lambda_{NRT} \quad - \; - \; - \; - \; - \; - \; - \; - \; - \; - \tag{4}$$

The effective service time for states zero to M is given as

$$\mu = \mu_{NRT} + \mu_{RT} \quad - \; - \; - \; - \; - \; - \; - \; - \; - \; - \tag{5}$$

The effective service time for states M+1 to C is given as μ_H

The state transition diagram for the model is shown in fig. 2.

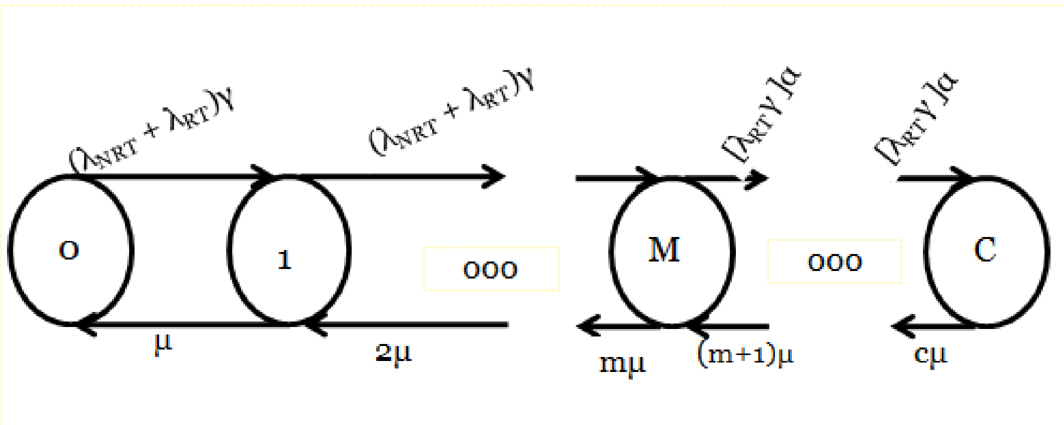

Figure 2: State transition diagram for system model.

$$P_{RT} = \frac{\frac{1}{R!M!}\left[\frac{(\lambda_{NRT}+\alpha\lambda_{RT})\gamma}{\mu_{NRT}+\mu_{RT}}\right]^{M}\left[\frac{\alpha[\lambda_{RT}\gamma]}{\mu_{RT}}\right]^{(R)}}{\sum_{S=0}^{M}\frac{1}{S!}\left[\frac{(\lambda_{NRT}+\alpha\lambda_{RT})\gamma}{\mu_{NRT}+\mu_{RT}}\right]^{S}+\sum_{S=(M+1)}^{C}\frac{1}{K!}\left[\frac{(\lambda_{NRT}+\alpha\lambda_{RT})\gamma}{\mu_{NRT}+\mu_{RT}}\right]^{M}\frac{1}{(S-M)!}\left[\frac{\alpha[\lambda_{RT}\gamma]}{\mu_{RT}}\right]^{(S-M)}} \qquad (6)$$

Statistical Approach

Four possible events can occur in the multimedia calls in CDMA system. These events are

1 The event that call is successful
2 The event that the call is blocked
3 The event that call is congested and
4 The event that the call is dropped

In view of the above, let,
Y refers to the event that the call is successful
X_1 refers to the event that the call is blocked
X_2 refers to the event that the call is congested
X_3 refers to the event that the call is dropped
Accordingly, let,
X_1^I be the event that there is no call block
X_2^I be the event that there is no call congestion
X_3^I be the event that there is no call drop

Y is desirable while X_1, X_2 and X_3 are all undesirable events. It is also worthy to note that events X_1, X_2 and X_3 are the major setbacks for event Y in the CDMA system.

It is considered that event Y is mutually exclusive of X_1, X_2 and X_3 while X_1^I , X_2^I *and* X_3^I are all independent events.

Applying set theory,

Y = X_1^I *and* X_2^I *and* X_3^I - - - - - - - - - (7)

Y = $X_1^I \cap X_2^I \cap X_3^I$ - - - - - - - - - - (8)

If the probability of Y is P[Y]
Then,

P[Y] = P[X_1^I] X $P[X_2^I]$ X $P[X_3^I]$ - - - - - - - - (9)

P[X_1^I] = 1 – P[X_1] = 1- P_{RT} - - - - - - - - (10)

P[X_2^I] = 1 – P[X_2] - - - - - - - - - (11)

P[X_3^I] = 1 – P[X_3] - - - - - - - - - (12)

Substituting equation 6 in 10

$$P[X_1^I] = 1 - \frac{\frac{1}{R!M!}\left[\frac{(\lambda_{NRT}+\alpha\lambda_{RT})\gamma}{\mu_{NRT}+\mu_{RT}}\right]^{M}\left[\frac{\alpha[\lambda_{RT}\gamma]}{\mu_{RT}}\right]^{(R)}}{\sum_{S=0}^{M}\frac{1}{S!}\left[\frac{(\lambda_{NRT}+\alpha\lambda_{RT})\gamma}{\mu_{NRT}+\mu_{RT}}\right]^{S}+\sum_{S=(M+1)}^{C}\frac{1}{M!}\left[\frac{(\lambda_{NRT}+\alpha\lambda_{RT})\gamma}{\mu_{NRT}+\mu_{RT}}\right]^{M}\frac{1}{(S-M)!}\left[\frac{\alpha[\lambda_{RT}\gamma]}{\mu_{RT}}\right]^{(S-M)}} \quad - - - \quad (13)$$

The next thing is to determine the probability of congestion.

P[X_2^I] = 1 – P[X_2] - - - - - - - - - (14)

It is known that, congestion varies directly with traffic,
i.e

P[X_2] $\propto \rho$ - - - - - - - - - - (15)

P[X_2] = $\Omega \,.\, \rho$ - - - - - - - - - - (16)

Where Ω is the traffic constant and ρ is the traffic intensity.
Therefore

P[X_2^I] = 1 - $\Omega \,.\, \rho$ - - - - - - - - - (17)

In [10], the dropping probability was found to be

$P[X_3] = \frac{(d_r t)^n}{n!}\rho^{-d_r t}$ - - - - - - - (18)

$P[X_3^I] = 1 - \left[\frac{(d_r t)^n}{n!}\rho^{-d_r t}\right]$ - - - - - - - (19)

Substituting equations 10, 11 and 12 in 9, we have,

$P[Y] = (1 - P[X_1])\,(1 - P[X_2])\,(1 - P[X_3])$ - - - - - - - (20)

$P[Y] = 1 - (P[X_1] + P[X_2] + P[X_3]) + P[X_1]\,P[X_2] + P[X_1]\,P[X_3] + P[X_2]\,P[X_3] - P[X_1]\,P[X_2]\,P[X_3]$ (21)

Substituting $P[X_1]$, $P[X_2]$ and $P[X_3]$ in equation 21

$$P[Y] = 1-$$

$$\left[\frac{\frac{1}{R!K!}\left[\frac{(\lambda_{NRT}+\alpha\lambda_{RT})\gamma}{\mu_{NRT}+\mu_{RT}}\right]^{K}\left[\frac{\alpha[\lambda_{RT}\gamma]}{\mu_{RT}}\right]^{(R)}}{\sum_{S=0}^{K}\frac{1}{S!}\left[\frac{(\lambda_{NRT}+\alpha\lambda_{RT})\gamma}{\mu_{NRT}+\mu_{RT}}\right]^{S}+\sum_{S=(K+1)}^{C}\frac{1}{K!}\left[\frac{(\lambda_{NRT}+\alpha\lambda_{RT})\gamma}{\mu_{NRT}+\mu_{RT}}\right]^{K}\frac{1}{(S-K)!}\left[\frac{\alpha[\lambda_{RT}\gamma]}{\mu_{RT}}\right]^{(S-K)}} + \Omega\,.\,\rho + \cdots \quad \cdots + \left[\frac{(d_r t)^n}{n!}\rho^{-d_r t}\right]\right] +$$

$$\left[\frac{\frac{1}{R!M!}\left[\frac{(\lambda_{NRT}+\alpha\lambda_{RT})\gamma}{\mu_{NRT}+\mu_{RT}}\right]^{M}\left[\frac{\alpha[\lambda_{RT}\gamma]}{\mu_{RT}}\right]^{(R)}}{\sum_{S=0}^{M}\frac{1}{S!}\left[\frac{(\lambda_{NRT}+\alpha\lambda_{RT})\gamma}{\mu_{NRT}+\mu_{RT}}\right]^{S}+\sum_{S=(M+1)}^{C}\frac{1}{M!}\left[\frac{(\lambda_{NRT}+\alpha\lambda_{RT})\gamma}{\mu_{NRT}+\mu_{RT}}\right]^{M}\frac{1}{(S-M)!}\left[\frac{\alpha[\lambda_{RT}\gamma]}{\mu_{RT}}\right]^{(S-M)}} \; X\,\Omega\,.\,\rho\right] +$$

$$\left[\frac{\frac{1}{R!M!}\left[\frac{(\lambda_{NRT}+\alpha\lambda_{RT})\gamma}{\mu_{NRT}+\mu_{RT}}\right]^{M}\left[\frac{\alpha[\lambda_{RT}\gamma]}{\mu_{RT}}\right]^{(R)}}{\sum_{S=0}^{M}\frac{1}{S!}\left[\frac{(\lambda_{NRT}+\alpha\lambda_{RT})\gamma}{\mu_{NRT}+\mu_{RT}}\right]^{S}+\sum_{S=(M+1)}^{C}\frac{1}{K!}\left[\frac{(\lambda_{NRT}+\alpha\lambda_{RT})\gamma}{\mu_{NRT}+\mu_{RT}}\right]^{M}\frac{1}{(S-M)!}\left[\frac{\alpha[\lambda_{RT}\gamma]}{\mu_{RT}}\right]^{(S-M)}} \; X \; \frac{(d_r t)^n}{n!}\rho^{-d_r t}\right]$$

$$+$$

$$\left[\Omega\,.\,\rho \; X \; \frac{(d_r t)^n}{n!}\rho^{-d_r t}\right] -$$

$$\left[\frac{\frac{1}{R!M!}\left[\frac{(\lambda_{NRT}+\alpha\lambda_{RT})\gamma}{\mu_{NRT}+\mu_{RT}}\right]^{M}\left[\frac{\alpha[\lambda_{RT}\gamma]}{\mu_{RT}}\right]^{(R)}}{\sum_{S=0}^{M}\frac{1}{S!}\left[\frac{(\lambda_{NRT}+\alpha\lambda_{RT})\gamma}{\mu_{NRT}+\mu_{RT}}\right]^{S}+\sum_{S=(M+1)}^{C}\frac{1}{M!}\left[\frac{(\lambda_{NRT}+\alpha\lambda_{RT})\gamma}{\mu_{NRT}+\mu_{RT}}\right]^{M}\frac{1}{(S-M)!}\left[\frac{\alpha[\lambda_{RT}\gamma]}{\mu_{RT}}\right]^{(S-M)}} \, X\,\Omega\,.\,\rho \; X \; \frac{(d_r t)^n}{n!}\rho^{-d_r t}\right] \quad (22)$$

Equation 322 is the proposed success probability model of multimedia calls (SPMMC)in the CDMA.

IV. CONCLUSION

The proposed scheme provides a quality of service guarantee to the multimedia calls and at the same time the exploitation of priority resources to accommodate blocked real time multimedia in order to improve the performance of the network. In this paper, it can be predicted that by integrating the concept of real time multimedia calls priority to the M+G, the real time call failure probability has been considerably reduced. Statistical approach using the continuous-time single dimensional birth–death process (Markov chain) was helpful in developing the mathematical model (SPMMC). This model when implemented will help to assess the performance of the proposed scheme.

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