A SIMULATION STUDY OF GENERIC BROADBAND WIRELESS INTEGRATED NETWORKS

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Abstract

A simulation study of a generic broadband integrated network is introduced. The network provides a wide range of services to multiple classes of users with different quality and transmission rate requirements. In order to achieve the performance objectives, a variety of service protocols are used. These include serving all users on a blocked calls cleared basis, modifying the transmission rate according to the available speed, introducing queuing facilities to wideband messages as well as combination of them. Using a computer simulation model, numerical results for the network performance under such protocols are presented and discussed.

I. Introduction

An important role of telecommunications over the next years is expected and service quality requirements are going to be diverse[1]. With the expected reduction in terminal cost and service charges the satisfaction of the unlimited needs of the business and public sectors will be realized[2]. In order to achieve maximum efficiency, flexibility and cost effectiveness, multiple classes of traffic streams with different characteristics, bandwidth and service requirements are allowed to share the facilities of a common network [3]. As the range of users becomes very broad, the network should have the capability of providing the required bandwidth to requesting users as they demand [4], [5]. An evolution of telecommunication networks toward a generic broadband integrated network with a flexible multi-slot assignment scheme is inevitable so that the potential users can access to any kind of voice, data or image services with great convenience [6], [7], [8]. Using such assignment scheme, the requested bit rates represent multiples of the basic bit rate transmitted in a single time slot. The connection is achieved by assigning a suitable number of time slots to the user according to his transmission rate.

The work presented in this paper describes a simulation study of a generic broadband integrated network. The potential services to be provided by such networks are discussed as well as their quality requirements. Then a simulation model is developed to mimic the operation of the network. A variety of service control protocols is introduced and the obtained results for the network performance under such protocols are presented and discussed.

II. Potential Services

The integration of voice and non-voice services is the primary goal of future generic networks. Enhanced voice services will be provided to the end users [9]. These include call forwarding, voice mail, and displays for caller identification and elapsed time and charges. The mean holding time of voice calls is substantially determined by subscriber behaviour. Voice traffic can tolerate a certain amount of degradation and occasional blocking without becoming objectionable. However, because of its real time property, large transmission delay may disrupt a conversation. Video telephony will become a common service in the near future. It is anticipated that video telephony will first spread among commercial users in offices and then branching its way to public users[6]. Its applications include show each other's faces, domestic scenes, objects, graphic materials and procedures.

Videotex is another service that is expected to be popular. It is an interactive access to a remote database, controlled by a videotex computer centre, by a person at a suitable terminal [8], [10]. These databases contain on-line telephone book, hundreds of magazine titles, government reports, statistics, indexes, demographic data, companies information and marketing studies [2], [6]. For this type of service the mean holding time is determined mostly by the subscriber response time and the transmission speed. Although this service is not strictly a real time one, it has certain delay limitations.

Another service is the video-on-demand (VOD) that supports retrieval of video programs (e.g. a feature film, video show, educational or entertainment programs) from centralised video libraries, and display on the customer's monitor [6], [11]. To succeed in its competition with the business of video tapes renting, VOD must compete not only on price but also on convenience, image quality, availability and reliability. This necessitates a considerably higher bandwidth or bit rate than ordinary telephone and in addition the user's waiting time to get service should be small.

Data transfer service between users will be offered more frequently in future networks [1], [6]. An example for this service is people who are working together on a project commonly need to share files. The service requirement in general terms entails zero information loss during transmission and bounded expected delay. The mean holding time in this case depends on the quantity of information and the transmission rate. The tolerance of the user to the perceived delay varies according to the type of application. In interactive applications, the response time to the user command is the key performance

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parameter while for file transfer type applications, the time required to transfer the whole file would be the key parameter. A special application is the transmission of fully assembled pages of newsprint as a facsimile or transmission of printing-plate data that will result in a more up-to-the-minute information as well as faster and cheaper newspaper distribution. As costs go down, a boom in video teleconferencing is expected [6]. Long distance travel and face-to-face meetings are displaced by allowing groups of individuals to communicate without the necessity of their congregating in one location. It can also serve close-in communities of interest, e.g. at companies with multiple locations or at universities with multiple campuses.

III. Simulation Model

Due to the sophistication of the generic broadband network, it is not amenable to tractable mathematical analysis. Most of the existing models consider only two parcels of traffic with different bandwidth and service requirements [12-17]. However, digital computer simulation provides a useful and effective alternative to evaluate the network performance. As the function of the network is the transmission of information through shared facilities, it is possible to mimic the operation of the network and get information about its behaviour under different conditions. The simulation process is driven by the occurrence of random events such as arrival of messages with different lengths and bandwidth requirements. The order of execution and handling methods of arriving messages are controlled by different scheduling policies that may be adopted.

A simulation model of a generic broadband network with N basic channels (time slots) has been developed. The model assumes that the network provides service to I classes of users. Each class is originated from an independent Poisson input stream with an average arrival rate λ_i so that the mean interarrival time of messages in the $i-th$ stream is $/\lambda_i$. The service time for each class is exponentially distributed with mean $(1/\mu_i)$ that varies from one class to the other. An arrival in class 1 is a digitally encoded narrowband voice call that requires a channel having a bandwidth corresponds to the basic bit rate of a single time slot and is served on a blocked calls cleared basis (BCC), i.e., an attempt that fails to get service leaves the network and make no immediate reattempt. An arrival in class i ; $i>1$, is a wideband message that requires n_i basic channels, or equivalently n_i time slots. The value of n_i is a positive integer. Class i messages; $i>1$, have different service policies that will be discussed in the next section. The simulation model is a discrete event simulation. If the event is an attempted access of a message of type i , the time to the next call is determined and the network state is updated. If the attempt succeeds, the required service time is specified and a new message record is defined so that a free user from the corresponding pool is identified as busy and considered as the requester. Then each user is tracked separately as the simulation progresses to follow up the state of the network. Once the message is over, the occupied channels are released simultaneously and the user record is deleted. If the access attempt fails, the message is treated according to the adopted service controlling protocol.

IV. Service Control Protocols

The performance of the network is mainly dependent on the method of distributing the available channels between the different classes of users. The access control protocol is of great importance as it is required to achieve the best utilisation of the network resources while satisfying the service quality requirements for each class of users. Based on the performance objectives, different control schemes may be adopted.

A. Scheme I

In this scheme, all classes of users share the network channels without preference. All messages are handled on a BCC basis with the future behaviour of the blocked user is completely unaffected by having been blocked. A narrowband call is blocked only if all channels in the network are being busy, while a wideband message is admitted into service only when all required channels are simultaneously available. A representation of this service control protocol is displayed in Figure 1. The traffic offered by class i is

$$A_i = n_i \frac{\lambda_i}{\mu_i} = n_i \rho_i \tag{1}$$

and the total traffic offered to the network is given by

$$A = \sum_{i=1}^{I} n_i \rho_i \tag{2}$$

B. Scheme II

In the former scheme, a wideband message may be blocked while network channels are unutilized because the available number of channels is insufficient to satisfy its demand. To increase the successful chance of wideband messages and improve channels occupancy we allow a wideband message to use the available channels upon its arrival. When a wideband message requires n_i channels for its connection while only $k<n_i$ channels are free, the requesting user reduces its transmission rate by a factor of (k/n_i) upon receiving a real-time control signal from the controller; presuming that the message source is capable to do so. A schematic representation of this protocol is shown in Figure 2. In some cases it is possible for the source coder to reduce its transmission rate at the penalty of degraded quality without the need to increase its

service time. On the other hand, when a conservation of the information quantity is necessary, it is inevitable to increase the service time by a factor of (n_i / k). In our simulation we consider the second situation as this is the general case.

C. Scheme III

In this scheme, queuing facilities are introduced for wideband messages. A narrowband call upon its arrival enters service if there is at least one free channel; otherwise, it is blocked and cleared from the network. If sufficient free channels are not available upon the arrival of a wideband message, it joins a queue until it can be served in order of arrival as shown in Figure 3. Considering an infinite queue, the message that joins the queue is assumed to wait as long as necessary to get service, while if a finite queue is implemented a wideband message finding a full queue is lost and cleared from the network without effect.

D. Scheme IV

This scheme integrates the benefits of introducing queuing facilities for wideband messages and the advantages of transmitting at the available speed. Based on a control signal, a wideband message upon its arrival can modify its transmission rate according to the available number of free channels; should all channels are occupied, the message can join a queue.

V. Results

A generic broadband network with three classes of users is studied. Class 1 represents narrowband calls that occupy one channel only while class 2 and 3 represent wideband messages that need more than one channel. Figure 4 depicts the blocking probabilities of all classes when the network adopts Scheme I (i.e., BCC) and a message belonging to class 2 needs two channels while that of class 3 requires four channels for connection. The total traffic offered is 36 Erlang; fifty percent of such traffic is considered as due to narrowband calls and the rest is divided between the two wideband classes. The number of channels available is taken as a variable. As expected, because a narrowband call is admitted into service if at least one free channel is available, the blocking probability of class 1 is the least. Accordingly, class 1 calls leave insufficient channels to accommodate the arriving wideband messages leading to higher blocking rate specially for class 3 messages. This greedy desire of narrowband calls is more pronounced in Figure 5 where a call of class 2 needs eight channels and that of class 3 needs sixteen channels. Once a wideband message is over, it releases eight or sixteen simultaneous channels, depending on its class. The incoming narrowband calls occupy these recently released channels as soon as they appear resulting in a reduction of class 1 blocking. This process continues until an arriving wideband message could find sufficient number of simultaneous channels that satisfies its demand. Then class 1 blocking increases again. The effect of such phenomenon appears as valleys in the blocking probability of class 1. Increasing the number of available channels reduces the blocking of all classes as the capacity of the network becomes more capable to accommodate higher traffic.

To relieve this harmful effect on wideband messages and reduce the probability that free (but insufficient) channels may be available while wideband access is denied we use Scheme II. Consider a network with 50 channels and an offered load of 36 Erlang. The required channels for class 1, 2 and 3 are one, two and four channels, respectively. A fraction α of the total traffic is due to narrowband and 70% of the rest is due to class 2. The performance of such network when Scheme I is applied is shown in Figure 6. Adopting Scheme II, based on a control signal the transmission rate is modified according to the available free channels and the service time is extended. This scheme gives a fair chance for all classes as shown in Figure 7 and wideband messages represent great rivals to narrowband calls resulting in blocking probabilities approximately of the same order.

Introducing queuing facilities for wideband messages increases their access chance. Upon finding insufficient number of channels, the wideband message joins a queue and as soon as the required channels become available it occupy them. However, the blocking probability of narrowband calls is affected by this persistent behaviour of wideband messages as shown in Figure 9 where the delay probabilities of class 2 and 3 as well as the blocking probability of class 1 are displayed. The performance of the network operating with Scheme IV is shown in Figure 6. As wideband messages are not queued when at least one free channel is available, the probabilities of delay of wideband messages and that of blocking of narrowband calls trace each other. As expected, blocking rate of narrowband calls is higher than that in Scheme I (Fig. 6) and sometimes wideband messages harm the access of narrowband calls resulting in delay probability that is lower than blocking chance of class 1.

VI. Conclusion

The future generic broadband networks must provide service to different categories of users with different quality and bit rate requirements. Such networks give users the opportunity to access to any kind of voice, data and image service. The paper describes a simulation study of the performance of a generic broadband network under a variety of service protocols. The choice between these service control protocols is substantially dependent on the performance objectives. We expect that the network provider should define a weighting function for the overall grade of service f the network. This weighting function is influenced by the service charge of each class, the quality requirements and users' satisfaction and habits. The weighting function may be given by

$$GOS_{overall} = \sum_{i=1}^{I} \xi_i . GOS_i \qquad (3)$$

where $GOS_{overall}$ is the overall grade of service of the network, GOS_i is the grade of service of class i and ξ_i is a weighting factor of class i with $0 \leq \xi_i \leq$ and $\sum_{i=1}^{I} \xi_i = 1$. The future generic broadband integrated networks should adopt an intelligent protocol that efficiently allocates the network channels to potential users by switching between these service protocols according to the weighting function defined.

VII. References

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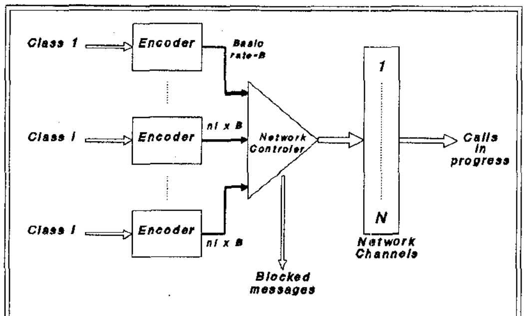

Fig. 1. Service control protocol of Scheme I.

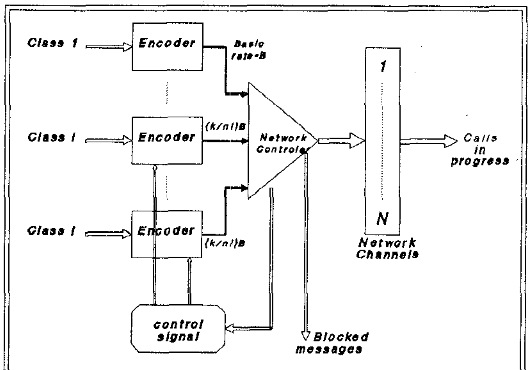

Fig. 2. Service control protocol for Scheme II.

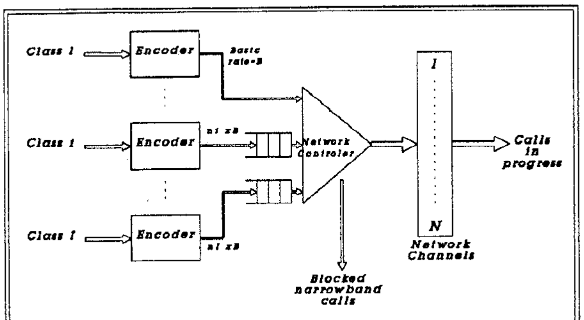

Fig. 3. Service control protocol of Scheme III.

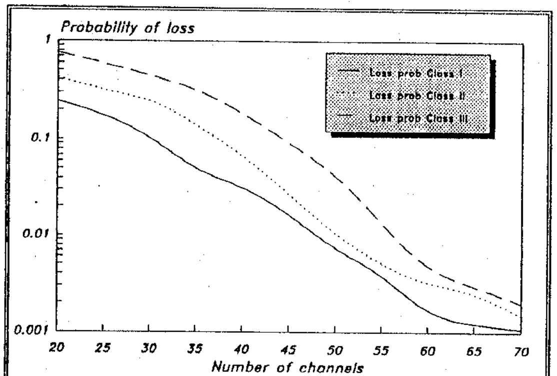

Fig.4. Loss prob.(Sch.I) when channels required are 1,2,4.

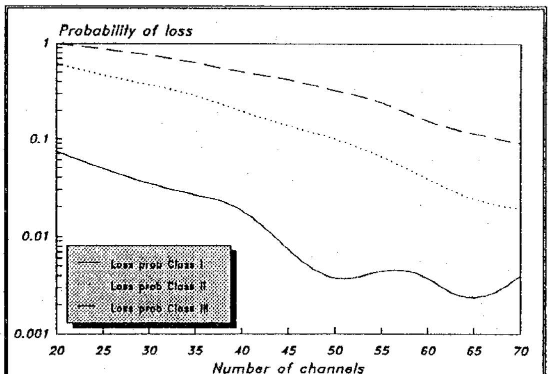

Fig.5. Loss prob.(Sch.I) when channels required are 1,8,16.

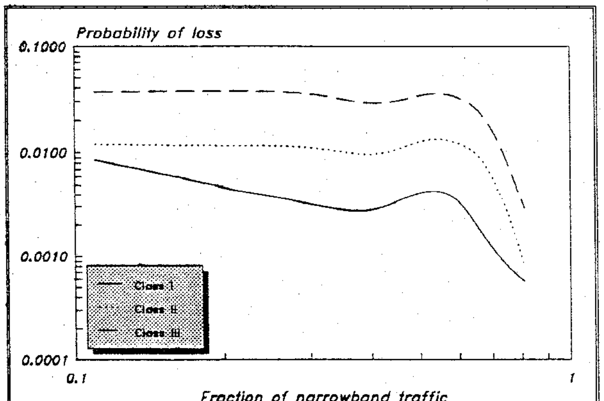

Fig.6. Loss prob.(Sch.I) for N=50, ni=1,2,4.

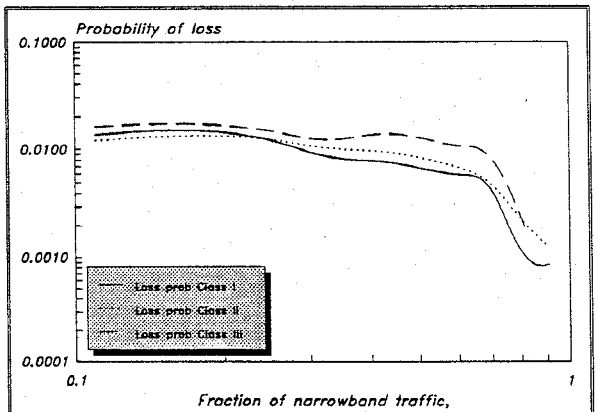

Fig.7. Loss prob. when Scheme II is adopted, N=50, ni=1,2,4.

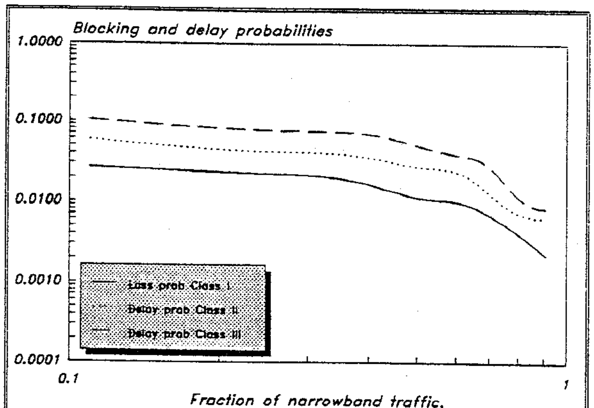

Fig.8. Loss and delay prob. when Scheme III is adopted.

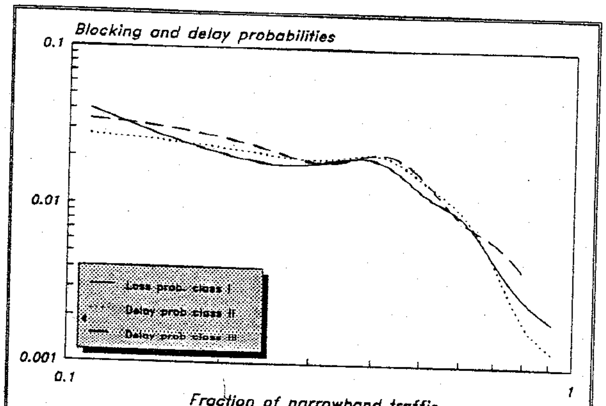

Fig.9. Loss and delay prob. when Scheme IV is adopted.