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Dynamic bandwidth allocation for 3G wireless systems—A fuzzy approach

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Abstract

3G Wireless systems are to support multiple classes of traffic with widely different characteristics and quality of service (QoS) requirements. A major challenge in this system is to guarantee the promised QoS for the admitted users, while maximizing the resource allocation through dynamic resource sharing. In the case of multimedia call, each of the services has its own distinct QoS requirements concerning probability of blocking (P_B), service access delay (SAD), and access delay variation (ADV). The 3G wireless system attempts to deliver the required QoS by allocating appropriate resources (e.g. bandwidth, buffers), and bandwidth allocation is a key in achieving this. Dynamic bandwidth allocation policies reported so far in the literature deal with audio source only. They do not consider QoS requirements. In this work, a fuzzy logic (FL)-based dynamic bandwidth allocation algorithm for multimedia services with multiple QoS (P_B, SAD, ADV, and the arrival rate) requirements are presented and analyzed. Here, each service can declare a range of acceptable QoS levels (e.g. high, medium, and low). As QoS demand varies, the proposed algorithm allocates the best possible bandwidth to each of the services. This maximizes the utilization and fair distribution of resources. The proposed allocation method is validated in a variety of scenarios. The results show that the required QoS can be obtained by appropriately tuning the fuzzy logic controller (FLC).
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Keywords: 3G wireless systems; Dynamic bandwidth; Quality of service (QoS); Service access delay (SAD); Access delay variation (ADV); Fuzzy logic controller (FLC)

1. Introduction

Wireless mobile systems, because of their appeal and utility have attractive market. Their appeal has sparked extensive developments in technology and deployment of powerful standards for various mobile applications over the last decade. 2G cellular systems, specifically with GSM, have been a tremendous success in the market. We are now poised to introduce 3G systems in a bid to extend service offerings, particularly in mobile data applications. The industry has defined and elaborated several 3G radio standards, all geared toward satisfying operators' needs.

Table 1 shows the various technologies in IMT-2000; CDMA, TD-CDMA, TD-SCDMA and EDGE. 3GPP2 handles CDMA2000. These technologies are considered to be of considerable significance to 3G.

For many years, 2G systems—GSM have been operated successfully all over the world. The anticipated demand for mobile data services providing high throughput, excellent quality of service (QoS), and improved system capacity, has prompted operators to begin screening their options for the best choice in a 3G mobile environment. A look at the potential 3G candidates will reveal that they all have close relation to their 2G predecessors.

In this paper, it has been taken into account that a total bandwidth of 5 MHz is available for utilization with a channel bit rate of 5 Mbps, which can be obtained by using any of these 3G systems.

1.1. The UMTS systems

Third Generation Partnership Project (3GPP) specification group has defined the Universal Mobile Telecommunication (UMTS) in recent years. It provides a new radio network architecture including W-CDMA (FDD) and TD-CDMA (TDD) radio technologies. GSM/GPRS/EDGE enabled

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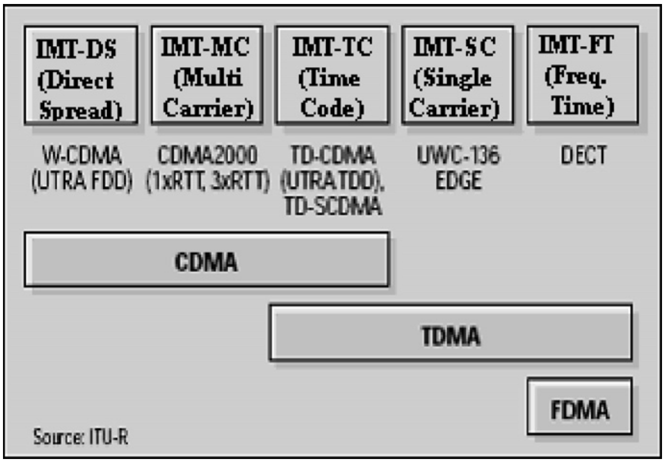
Table 1
IMT-2000 terrestrial radio interfaces and categories



services can interwork with GSM and can support advanced features such as multi-rate wideband voice codec, IP-based multimedia services (IMS), and high speed downlink packet access (HSDPA).

1.2. The CDMA2000 system

CDMA2000 [12] is another major 3G CDMA technology. It is an evolutionary approach based on IS-95. It is designed to increase the data transmission rates and voice capacity of legacy IS-95 networks. In contrast to W-CDMA, it includes a multi-carrier CDMA concept designed for 1.25 MHz carrier bandwidth with a chip rate of 1.2288 Mcps. A single 5 MHz band can accommodate three carriers.

1.3. 3G wireless system—service classes

A fully developed 3G wireless network can offer bandwidth on demand to users in a form that matches their QoS needs, from isochronous constant bandwidth to sporadic high bandwidth bursts. Emerging high speed networks integrate a wide variety of applications with different traffic characteristics and quality of service (QoS) requirements. For this purpose, network applications can be classified into different categories otherwise called as service classes [5,6] with respect to their QoS requirements.

1.3.1. Service classes—description

1.3.1.1. Constant bit rate (CBR). This class is used for voice applications where the bit rate is constant. CBR applications are quite sensitive to call-delay variations. Examples of applications that use CBR are telephone traffic (namely, 64 kbps), television and video-conferencing.

1.3.1.2. Variable bit rate-nonreal time. This class allows users to send traffic at the rate (VBR-nrt) that varies with time depending on the availability of user information. Multimedia e-mail service is an example of this class.

1.3.1.3. Variable bit rate-real time. This class is similar to VBR-nrt but is designed (VBR-rt) for applications which are sensitive to call-delay variations. Examples for VBR-rt services are voice service with speech activity detection and interactive compressed video.

1.3.1.4. Available bit rate (ABR). This class provides a rate-based flow control and is aimed at data traffic such as file-transfer and e-mail. Although, this standard does not require the call transfer delay and information loss to be guaranteed or minimized, it is always desirable to minimize the delay and the loss as much as possible.

1.3.1.5. Unspecified bit rate (UBR). This service is designed to accommodate some of the current common computer applications, such as file transfers that occur in spaced bursts of bits and have relatively few service requirements. It provides no QoS guarantees and in turn not subjected to guaranteed bandwidth application. Table 2 gives the QoS requirements for various traffic classes and applications.

1.4. Call admission control

Connection or call admission control [2] is a technique used by the wireless network for protecting itself from excessive loads. In essence, when a new connection is requested, the user must specify the traffic characteristics for that connection. The user selects traffic characteristics by selecting a QoS from among the QoS classes that the network provides. The network accepts the connection only if it can commit the resources necessary to support that traffic level while at the same time maintaining the agreed upon QoS of existing connections. Once the connection is accepted, the network continues to provide the agreed upon QoS as long as the user complies with the traffic contract.

This kind of controlling of the number of incoming calls is done because of the specific characteristics of wireless networks. In wireless networks, most traffic sources are bursty. A bursty source may generate bits at a near-peak rate for a very short period of time and immediately afterwards may become inactive, generating no bits. Such a bursty traffic does not need a bandwidth equal to its peak rate continuously.

Since the 3G wireless network supports a large number of such bursty sources, allowing more traffic sources to share the bandwidth at a time, i.e. multiplexing them can be used to gain bandwidth efficiency. But if large number of traffic sources becomes active simultaneously, severe network congestion can result. Due to these effects, congestion control is a challenge for wireless networks.

Also, control schemes that adjust the transmission rate of the source based on the feedback from destination may not work in wireless networks. Traditionally used congestion control schemes fall in the class of reactive controls. Reactive control scheme reacts to the congestion to a certain level only, which is not suitable for wireless networks.

Preventive control, unlike reactive control does not wait until congestion actually occurs, but rather tries to prevent the network from reaching unacceptable level of congestion. The

Table 2
3G Wireless system—UMTS QoS classes

Class	Traffic class	Class description	Example	Relevant QoS requirements
1	Conversational	Preserves time relation between entities making up the stream conversational pattern based on human reception; real time	Voice, voice telephony, video gaming, video conferencing	Low jitter, low delay
2	Streaming	Preserves time relation between entities making up the stream; real time	Multimedia, video on demand, web cast, real time video	Low jitter
3	Interactive	Bounded response time, preserves the payload content	Web browsing, data retrieval	Low round trip delay time, low BER
4	Background	Preserves the payload content	E-mail, SMS, file transfer	Low BER

most common and effective approach is to control the traffic flow at entry point. Here the decision to admit new traffic can be based on the knowledge of the capacity of the channel in which the traffic would flow. Preventive control in wireless networks can be performed in two ways: admission control and bandwidth allocation.

Admission control determines whether to accept or reject a new connection at the time of call set-up, as has already been described. Request for the new connection specifies the required QoS parameters. The decision on whether or not the request should be granted is based on the current network load and the required QoS.

1.5. Trunking and grade of service

Cellular radio systems rely on trunking to accommodate a large number of users in a limited radio spectrum. The concept of trunking [11] allows a large number of users to share the relatively small number of channels in a cell by providing access to each user on demand from a pool of available channels. In a trunked radio system, each user is allocated a channel on a per call basis, and upon termination of the call, the previously occupied channel is immediately returned to the pool of available channels.

The grade of service (GOS) is a measure of the ability of a user to access a trunked system during the busiest hour. GOS is typically given as the likelihood that a call is blocked, or the likelihood of a call experiencing a delay greater than a certain queuing time.

There are two types of trunked systems which are commonly used. The first type offers no queuing for call requests. That is, for every user who requests service, it is assumed there is no setup time and the user is given immediate access to a channel if one is available. If no channels are available, the requesting user is blocked without access and is free to try again later. This type of trunking is called Blocked Calls Cleared.

In this paper, the second kind of trunked system, namely Blocked Calls Delayed is used. In this system the calls are held in a queue (buffer) if the channel is not available immediately. In such case the calls are delayed.

1.6. QoS parameters

When a new connection is requested, the network needs to know the traffic characteristics of the new connection in order to accurately predict its ability to maintain a certain performance level. A set of traffic descriptors and the acceptable level of QoS parameters are given to the network by the user. The network decides whether to admit a call or not and how much bandwidth to allocate based on the traffic descriptors and QoS parameters.

The QoS parameters considered in this work are blocking probability (P_B), service access delay (SAD) and access delay variation (ADV) or also called jitter.

1.6.1. Probability of blocking (P_B)

Wireless cellular networks derive their name from the fact that the service area of these networks is divided into a group of cells, with each cell being controlled by a base station (BS). When a user requests a service (of any traffic type—voice, audio, data or video) from the network through a mobile terminal (MT), his request may be accepted or denied based on the availability of required resources in the user's resident cell. This denial of service is known as call blocking, and its probability is called call blocking probability P_B.

Thus the blocking probability, P_B is defined as the ratio of the number of calls blocked to the total number of calls arrived.

1.6.2. Service access delay (SAD)

Service access delay (SAD) [10] is the time between sending of complete address information to the base station and receipt of call set-up notification to the user handset. This service access delay (for any type of traffic—voice, audio, data or video) consists of three components, i.e.:

$$\mathrm{SAD} = d_r + d_q + d_t$$

where, d_r is the average time of successfully sending a call transmission request, d_q the queuing time before a channel is allocated to the call requested, and d_t is the transmission time of information after the traffic channel is allocated.

These transmission time delays d_r and d_t cannot be controlled through simulation and they depend on many factors, such as nature of wireless channel (for example, AWGN) through which information is transmitted and also on multipath effects. The average value of total transmission delay is assumed to be 25 ms [3].

1.6.3. Access delay variation (ADV) or jitter

ADV is also called 'jitter'. It is an important issue to be taken care of while using multimedia services in wireless system. ADV is defined as a measure of variance of the service access delay. High variation implies larger buffering for delay sensitive traffic, such as audio and video. When audio, video, voice, and data traffic are simulated and ADV for all the above services are determined it is found that ADV is negligible for voice and data. The ADV values are high for audio and video traffic. So in this paper, jitter is taken into consideration for audio and video traffics only.

Jitter is an important parameter used in defining the QoS parameters. The end-to-end delay of the ith information bit can be specified as $D + W_i$, where D is a constant and W_i, a stochastic variable determining jitter. W_i arises out of buffer overflows within the network caused by asynchronous multiplexing as well as information losses arising from bit-errors which can not be corrected.

1.7. Overview of the literature

3G wireless systems are the area in which voluminous research has been done and there is still room for improvement, as it becomes part of everyday life. Many research papers that have been published deal with homogeneous sources mainly video because of its importance. In Ref. [4], the modeling of video using diffusion approximation is explained. The concept of introducing QoS index depending on the source requirements are found in [2,8,10].

The importance of bandwidth allocation and the types of traffics, such as video, audio, data and voice and their real time traffic and delay parameters considered for multimedia services are reported in [1,3].

The different types of 3G systems and their characteristics and standards are dealt in Ref. [7,9]. The concept of fuzzy logic and the design of fuzzy logic controller are dealt in Refs. [13,14].

The 3G wireless systems utilization has started in wireless communication. To allocate dynamic bandwidth for heterogeneous sources with various QoS parameters fuzzy logic controller has been implemented. This approach has been adopted because of the ability of the fuzzy logic in handling multiple inputs and outputs with ease.

Here, novel dynamic bandwidth allocation of multimedia traffic in 3G wireless networks using fuzzy logic (FL) is proposed. A fuzzy logic controller (FLC) can provide algorithms which can convert the linguistic control strategies based on intuition, heuristic learning and expert knowledge into an automatic control strategy for connection admission control and bandwidth management purposes. In particular, the methodology of FLC appears to be very useful when the processes are too complex for analysis by conventional quantitative techniques, and when the available information cannot be interpreted correctly and with certainty.

The goal is to design an efficient, simple, and robust algorithm by tuning the FLC which adapts itself to the variability of the traffic rate and the QOS requirements. Moreover, class multiplexing among different classes of traffic (i.e. voice, audio, data and video) has been included in the design. This is achieved with separate FL units for each type of traffic. Since this approach does not rely on specific traffic parameters, robustness is achieved. Moreover, FLs do not require an accurate mathematical modeling of the system in contrast to conventional mathematical approaches and also their processing speed is extremely fast.

The rest of the paper is organized as follows: in Section 2, we present the design of dynamic bandwidth allocation. Section 3 gives the design of FLC. Section 4 explains the modeling of traffic sources. Section 5 deals with the simulation of QoS parameters for the generation of the rule base, and results and conclusions are given in Section 6.

2. Dynamic bandwidth allocation

A 3G wireless link with C_{link} bps capacity is considered for multiple sources with multiple classes of traffic (i.e. different traffic characteristics and QoS requirements). The measures of QoS are blocking probability (P_{B}), service access delay (SAD), and access delay variation (ADV). Only four virtual paths are considered to be supported over the link. Each virtual path supports sources with similar class of traffic. For example, video sources with different QoS will be supported on first virtual path, audio sources on second virtual path, voice sources on third virtual path, and data on the fourth virtual path. The reason for this classification is to simplify the FLC design so that the number of membership functions and the fuzzy rules can be reduced. Each of the voice, audio, video and data services is divided into K subclasses. To admit a new class k call, the FLC computes the bandwidth C_k ($k = 1, 2, \ldots, K$) to be logically allocated to class k with acceptable QoS and the condition $\sum_{k=1}^{k} C_k \leq C_{\text{link}}$ is satisfied. The FLC, as shown in Fig. 1, explicitly determines the bandwidth C_k by taking into account the statistical multiplexing gain amongst the connections within each class. In this paper, it is assumed that the FLC does take advantage of statistical multiplexing between different classes. Statistical multiplexing of bandwidth between different connections is a multidimensional problem and is dealt here using FLC. We also assume that there is no sharing of buffers between different classes since such sharing may compromise the QoS guarantee for some classes.

It is important to note that different classes of traffic (e.g. CBR, VBR, etc.) exhibit different traffic characteristics and have different QoS requirement as statistical multiplexing is best achieved among connections with similar statistical characteristics and QoS requirements. Each class k is assigned a minimum service rate computed by FLC to guarantee its specified QoS. The bandwidth for the classes with more stringent QoS requirements is allocated by dynamically sharing the available bandwidth from other classes, whereas for the classes with less stringent QoS requirements less bandwidth is allocated less than they would need, with the assumption that the shortfall will be made-up by dynamic sharing. The FLC proposed in this work provides improved bandwidth utilization by satisfying the QoS requirements for video (nrt-VBR), voice (rt-CBR), audio (nrt-CBR) and data (nrt-VBR).

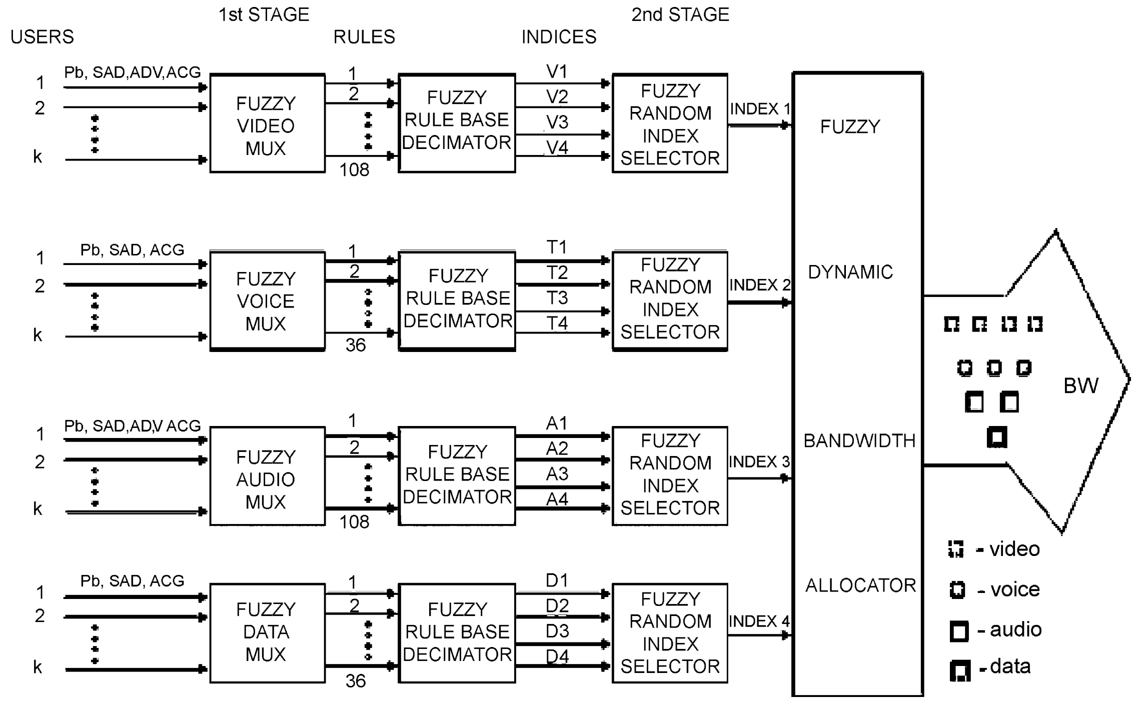


Fig. 1. Fuzzy logic controller.

3. Design of fuzzy logic controller

For systems with little complexity, hence less uncertainty, closed form mathematical expressions provide precise description. But, for most complex systems, where ambiguous or imprecise information only are available, the fuzzy reasoning provides a way to understand the system behavior by allowing us to interpolate approximately between observed input and output. If a fuzzy system has 'n' inputs and single output, then its fuzzy rules can be of the following general format.

If X_1 is A_{1j} AND X_2 is A_{2j} AND X_3 is $A_{3j}, \ldots, X_m$ is A_{mj} THEN Y is B_j, where A_{ij} are the fuzzy sets of the input linguistic variable X_i, and B_j is called the set of the output linguistic variable Y.

Generally, in 3G wireless networks the average number of calls generated (ACG) and the QoS parameters exhibit stochastic behavior, and the bandwidth to be allocated depends on these parameters. In the absence of FLC, the random changes in the arrival rate leads to the degradation of promised QoS parameters, which necessitates dynamic bandwidth allocation. This uncertain nature of network traffic process can be adequately captured by fuzzy approach with ACG, P_B, SAD and ADV as multiple input variables and bandwidth as the single output variable, i.e. the process can be considered as multiple input and single output (MISO) system.

The FLC consists of two stages. In the first stage, there are four fuzzy subsystems arranged in parallel corresponding to four types of sources: video, voice, audio and data as shown in Fig. 1. For the first stage fuzzy systems of video and audio, there are four inputs and for voice and data, three inputs since ADV is not considered for them as input and one output for each call. The inputs are the three QoS parameters and the number of calls generated per time instant. The output is an index corresponding to the rules.

For each of the three QoS inputs, three trapezoidal membership functions namely "low", "medium", "high" are defined, whereas for the fourth input four trapezoidal membership functions, viz. "low", "medium", "high" and "very high" are defined in order to capture the wide variation in the arrival rate. The trapezoidal membership functions are chosen so that many rules can have maximum membership function value. Therefore, the number of possible combinations or indices of the possible states for the input variables is equal to $3 \times 3 \times 3 \times 4 = 108$ (for voice and data, $3 \times 3 \times 4 = 36$), which is equal to the number of rules needed for the first stage of the FLC. These fuzzy rules are given in Appendix A for

Table 3
Grouping of indices for first stage

Video		Voice		Audio		Data	
Rule no.	Index	Rule no.	Index	Rule no.	Index	Rule no.	Index
1–27	VI	1–9	T1	1–27	A1	1–9	D1
28–54	V2	10–18	T2	28–54	A2	10–18	D2
55–81	V3	19–27	T3	55–81	A3	19–27	D3
82–108	V4	28–36	T4	82–108	A4	28–36	D4

Table 4
Input and output ranges for the first stage

Type of source	Input				Output
	P_B	SAD (s)	ADV (s)	ACG (traffic-kbps)	Index
Video	0–0.71	0.025–0.42	0–0.2	0–7300	1–108
Voice	0–0.38	0.025–1.76	–	0–520	1–36
Audio	0–0.58	0.025–1.34	0–0.26	0–1410	1–108
Data	0–0.38	0.025–0.99	–	0–2440	1–36

video sources. Similar rules are generated for audio and data sources also.

Each of voice and data has 36 indices. Audio and video has 108 indices, which are fed as inputs to the second stage FLC. In order to simplify the second stage of the FLC, the simulated data of each source is analyzed and the statistics of occurrence of each index is observed. Based on these observations, the indices are grouped, as shown in Table 3 for voice, audio, video and data. Hence, in the rule base decimator, the 36 or 108 indices are reduced to 4 indices for each of voice, audio, video and data. Now, the total number of rules is reduced to $4 \times 4 \times 4 \times 4 = 256$, which form the knowledge based rules for the second stage of FLC. Sample of these rules for video index 1 is shown in Appendix B. The membership functions for these indices are assumed to be trapezoidal. Using the knowledge, the indices are randomly selected, one each for voice, video, data, and audio. The idices are fed as inputs to fuzzy bandwidth allocator. The output (bandwidth) membership functions for each of the sources are assumed to be again Trapezoidal. The center average defuzzifier is used to estimate the crisp output bandwidth value for each of the sources from these membership functions. The input output ranges for the first stage of FLC is given in Table 4 and for the second stage in Table 5. The FLC design is implemented using Matlab 6.0 version.

Whenever there is a change in the user QoS requirement or in the arrival rate, only some rules out of the 36 or 108 rules are fired leading to changes in the indices which in turn changes the bandwidth dynamically as per the user's requirements.

4. Modeling of traffic sources

As mentioned earlier, 3G wireless cellular systems must support various communication services, each having its own traffic characteristics and QoS requirements. For evaluation, accurate source modeling is required. Several traffic models have been proposed in the literature. ON–OFF and IPP, alternate state renewal process, MMPP, MMRP are a few examples of Markov and embedded Markov processes. Regression models like autoregressive moving average models and integrated moving average values, and long-range-dependent models are some other examples. In our project, we use Markovian models which are the most analyzed, studied and widely applied in the field owing to their tractability.

Table 5
Input–output ranges for second stage

Type of source	Input (index)	Output (bandwidth (bits/s))
Video	1–108	0.6E+6–3.89E+6
Voice	1–36	0.05E+6–0.375E+6
Audio	1–108	0.145E+6–1.03E+6
Data	1–36	0.21E+6–1.7E+6

4.1. Voice, audio and data source modeling

When N independent sources are multiplexed, aggregated call arrivals are governed by the number of sources. ON–OFF model is a special case of MMPP model in which no call arrives in the OFF period. Such ON–OFF models as shown in Fig. 2 are best used to approximate voice, audio and data services wherein, during the ON state, call arrives at constant rate for CBR traffics or at different rates for VBR traffics. The source remains OFF otherwise.

4.1.1. Voice source modeling

In this work, Voice is modeled as rt-CBR as shown in Fig. 3. The voice sources are simulated using the ON–OFF model for a single source and aggregating many such sources. The voice source generation follows exponential inter-arrival times with constant arrival rate.

The design parameters considered for simulation are:

- The percentage of voice calls among all calls generated is 60% [3].

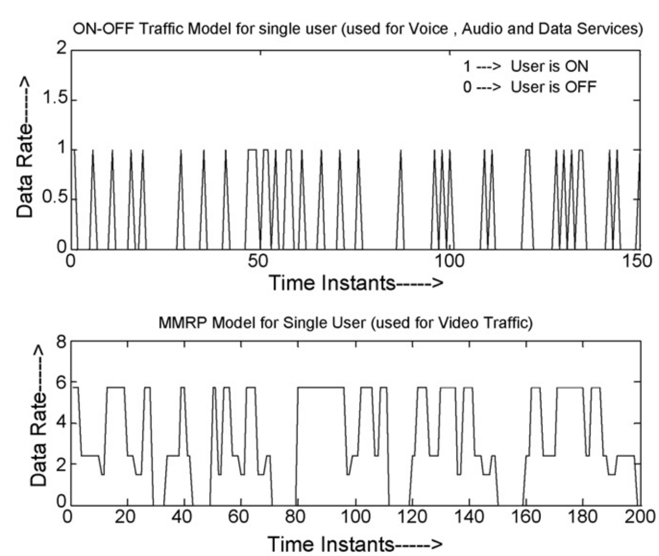


Fig. 2. ON–OFF and MMRP models Used for generating sources.

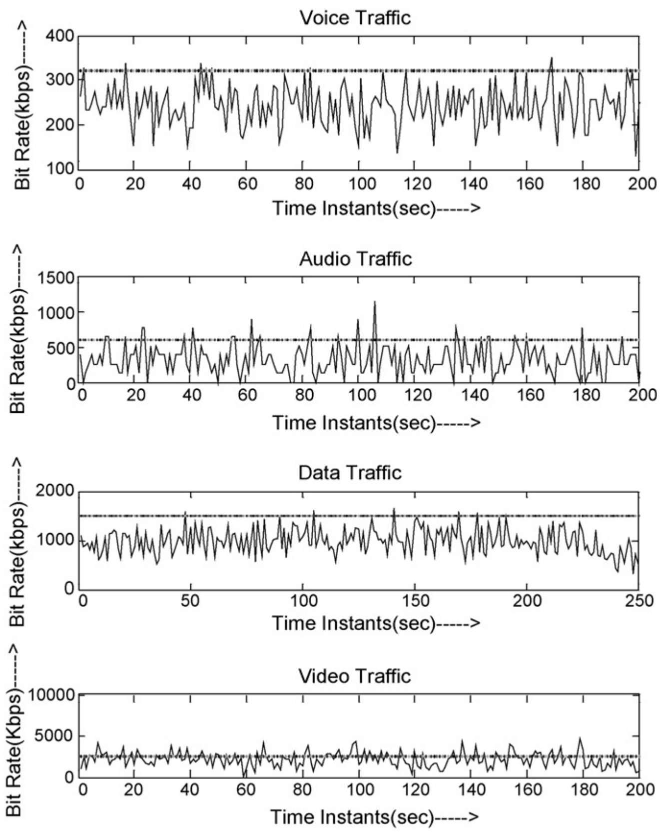


Fig. 3. Traffic generated by the multimedia sources.

- The duration of a voice connection is exponentially distributed with an average equal to 180 s [3].
- The bit rate provided for each voice connection is equal to 8 kbps.

4.1.2. Audio source modeling

Audio is modeled as nrt-CBR as shown in Fig. 3. The audio sources are simulated using the ON–OFF model for a single source and aggregating many such sources. The audio source generation follows exponential inter-arrival times with constant arrival rate.

The design parameters considered for simulation are:

- The percentage of audio calls among all calls generated is 5% [3].
- The duration of an audio connection is exponentially distributed with an average equal to 180 s [3].
- The bit rate provided for each audio connection is equal to 128 kbps [3].

4.1.3. Data source modeling

Data is modeled as nrt-VBR as shown in Fig. 3. The data sources are simulated using the ON–OFF model for a single source and aggregating many such sources. Similar to the generation of voice and audio sources, data source generation too follows exponential inter-arrival times, but unlike the case of voice or audio, where bit rate is constant, for data source it varies randomly.

The design parameters considered for simulation are:

- The percentage of data calls among all calls generated is 20% [3].
- The duration of an data connection is exponentially distributed with an average equal to 360 s [3].
- The bit rate provided for each data connection is assumed to vary from 0 to a maximum of 100 kbps.

4.1.4. Video source modeling

Video sources are simulated by MMRP model as shown in Fig. 2. Assuming there are "k" sources, each is considered to be an MMRP. Each video source is governed by an m-state Markov chain with probability transition matrix $P = [p_{ij}]$, where $i, j = 0, 1, 2, \ldots, m-1$. When a source is in state m, it generates R_m bps. The duration or the holding time of state m has a general distribution with mean α_m^{-1} and variance σ_m^2. When the source exists in state i, it moves to state j with probability P_{ij}. The probability transition matrix for each MMRP source is given below:

$$P = \begin{bmatrix} 0 & 0 & 1/2 & 1/2 \\ 1/6 & 0 & 1/3 & 1/2 \\ 2/5 & 1/5 & 0 & 2/5 \\ 1/5 & 3/10 & 1/2 & 0 \end{bmatrix}$$

Here, the VBR source rates are assumed to be 120 kbps, 180 kbps, 240 kbps, and 420 kbps. The holding time for each state is assumed to be exponentially distributed with mean of 160 ms.

The percentage of video calls among all calls generated is 15% [3].

The duration of a Video connection is exponentially distributed with an average equal to 180 s [3].

5. Simulation of QoS parameters for rule base generation

In order to enable the FLC to allocate the bandwidth dynamically, the inference engine has to be fed with knowledge-based rules. These rules are derived by simulation. The simulation details are given below.

At every instant, each of the N sources generates a certain number of bits and these bits are added to obtain the total number of bits at that instant. The channel capacity Clink as a function of average bps is assumed, such that the QoS parameters measured using the channel fall within the specified range. This channel is assumed to be constant over the simulation period. The QoS parameters which are used for defining the input membership functions for the FLC are computed as given below:

$$\text{blocking probability } (P_{\mathrm{B}}) = \frac{\text{number of blocked calls in a cell}}{\text{number of new call arrivals at that cell}} = \frac{N_{\mathrm{B}}}{N}$$

Then the average delay, d_{q} of those calls which are queued is given by:

$$d_{\mathrm{q}} = \frac{P_{\mathrm{r}}[C_{\mathrm{D}}]H}{C - A}$$

where, H is Average call holding time = A/λ and C_D is the call being delayed:

$$P_r[C_D] = \frac{A^c}{A^c + C!(1 - A/C)\sum_{k=0}^{C-1}(A^k/k!)}$$

where, A is the channel capacity in Erlangs, C the number of trunked channels in system and λ is the average call arrival rate:

$$\text{service access delay (SAD)} = d_q + 25\,\text{ms}$$

$$\text{access delay variation (ADV)} = \frac{1}{M}\sum_{i=1}^{M}(\text{SAD}_{i+1} - \text{SAD}_i)$$

where, M is the number of calls waiting in the queue, SAD_i is the service access delay of the ith call.

The variation of QoS parameters with respect to the call arrival rate for video source is shown in Fig. 4.

6. Results and conclusions

In Fig. 5, the mean bandwidth required for each source and total bandwidth required to satisfy QoS parameters, i.e. the bandwidth that has been generated from simulation and the fuzzy allocated bandwidth are plotted in the y-axis against the time instants on x-axis.

From these figures, it is observed that for all the QoS requirements to be satisfied, ideally bandwidth allocated should be equal to the total traffic generated (by all sources) at every instant. But, while simulating, bandwidth is assumed to be a function of the average number of calls generated at every instant and this was kept constant throughout the period of time (number of instants used to calculate the average). This resulted in the allocated bandwidth being lesser than the estimated bandwidth stated in order for the QoS parameters to fall within

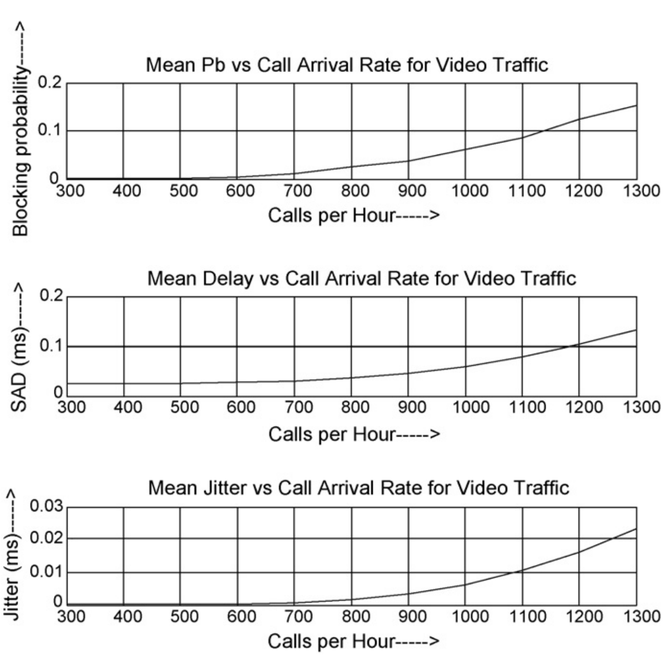


Fig. 4. Variation of QoS parameters for video source.

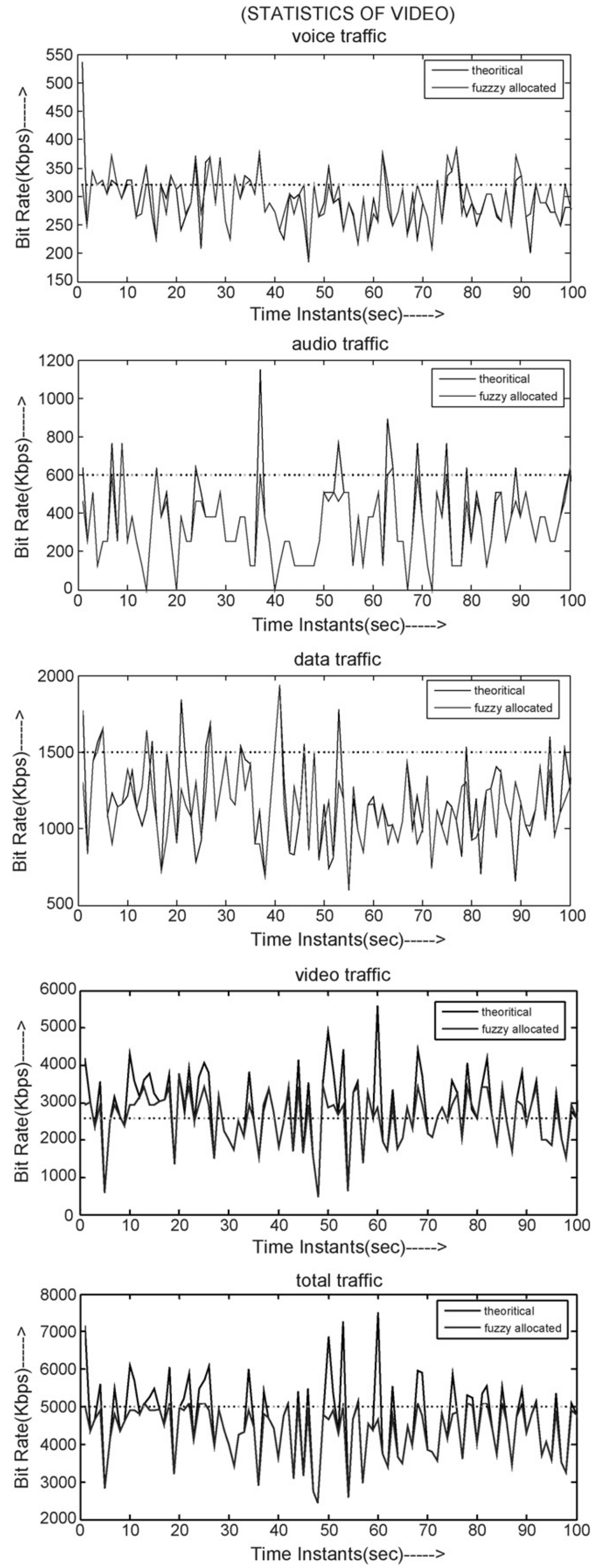


Fig. 5. Comparison of theoretical and fuzzy allocated bandwidth for voice, audio, data, video sources and the total traffic at high call arrival rates (i.e. during busy periods).

Table 6
Percentage deviation between theoretical (estimated) and fuzzy allocated bandwidth at different call arrival rates

Call arrival rate (calls/h)	Theoretical bandwidth (kbps)	Fuzzy allocated bandwidth (kbps)	Percentage deviation (%dev)
300	1158.1	1158.1	0
500	1905.4	1905.4	0
700	2707.3	2706.5	0.0414
900	3439	3423.9	0.439
1000	3849	3801	1.247
1100	4224	4106.4	2.784
1200	4599.4	4338.4	5.672
1300	5061.5	4585.5	9.415

the specified range over the entire period. In order to meet the required QoS parameters FLC allocates bandwidth which is slightly deviating from the ideal (estimated).

6.1. Inference

At low and medium call arrival rates, FLC allocates bandwidth dynamically satisfying the required QoS.

But, at high call arrival rates, there is deviation between estimated and allocated bandwidth even though the required QoS is achieved. This deviation is due to the inherent property of the FLC in limiting the number of users entering the system at high call arrival rates and available bandwidth of 5 Mbps.

From Table 6, it is seen that the deviation is lying below 10% even at very high bandwidth at higher call arrival rates call arrival rate of 1300 calls/h. This deviation is more for video, since video service alone requires more bandwidth (>5 Mbps). At high call arrival rates all the services are allocated fair bandwidths. Thus fairness in bandwidth allocation is achieved.

Fuzzy logic controller ensures dynamic allocation, i.e. allocated bandwidth varying instantaneously with varying QoS requirements of users and varying traffic which is bursty in nature. Since all the QoS parameters are considered simultaneously along with the arrival rates the user's quality expectations are satisfied.

Appendix A. Rules for the first stage of FLC (video service)

S. no.	P_B	SAD	ADV	ACG	Index
1	L	L	L	L	M1
2	L	L	M	L	M1
3	L	L	H	L	M1
4	L	M	L	L	M1
5	L	M	M	L	M1
6	L	M	H	L	M1
7	L	H	L	L	M1
8	L	H	M	L	M1
9	L	H	H	L	M1
10	M	L	L	L	M1
11	M	L	M	L	M1
12	M	L	H	L	M1
13	M	M	L	L	M1
14	M	M	M	L	M1
15	M	M	H	L	M1
16	M	H	L	L	M1
17	M	H	M	L	M1

Appendix A (*Continued*)

S. no.	P_B	SAD	ADV	ACG	Index
18	M	H	H	L	M1
19	H	L	L	L	M1
20	H	L	M	L	M1
21	H	L	H	L	M1
22	H	M	L	L	M1
23	H	M	M	L	M1
24	H	M	H	L	M1
25	H	H	L	L	M1
26	H	H	M	L	M1
27	H	H	H	L	M1
28	L	L	L	M	M2
29	L	L	M	M	M2
30	L	L	H	M	M2
31	L	M	L	M	M2
32	L	M	M	M	M2
33	L	M	H	M	M2
34	L	H	L	M	M2
35	L	H	M	M	M2
36	L	H	H	M	M2
37	M	L	L	M	M2
38	M	L	M	M	M2
39	M	L	H	M	M2
40	M	M	L	M	M2
41	M	M	M	M	M2
42	M	M	H	M	M2
43	M	H	L	M	M2
44	M	H	M	M	M2
45	M	H	H	M	M2
46	H	L	L	M	M2
47	H	L	M	M	M2
48	H	L	H	M	M2
49	H	M	L	M	M2
50	H	M	M	M	M2
51	H	M	H	M	M2
52	H	H	L	M	M2
53	H	H	M	M	M2
54	H	H	H	M	M2
55	L	L	L	H	M3
56	L	L	M	H	M3
57	L	L	H	H	M3
58	L	M	L	H	M3
59	L	M	M	H	M3
60	L	M	H	H	M3
61	L	H	L	H	M3
62	L	H	M	H	M3
63	L	H	H	H	M3
64	M	L	L	H	M3
65	M	L	M	H	M3
66	M	L	H	H	M3

Appendix A (*Continued*)

S. no.	P_B	SAD	ADV	ACG	Index
67	M	M	L	H	M3
68	M	M	M	H	M3
69	M	M	H	H	M3
70	M	H	L	H	M3
71	M	H	M	H	M3
72	M	H	H	H	M3
73	H	L	L	H	M3
74	H	L	M	H	M3
75	H	L	H	H	M3
76	H	M	L	H	M3
77	H	M	M	H	M3
78	H	M	H	H	M3
79	H	H	L	H	M3
80	H	H	M	H	M3
81	H	H	H	H	M3
82	L	L	L	VH	M4
83	L	L	M	VH	M4
84	L	L	H	VH	M4
85	L	M	L	VH	M4
86	L	M	M	VH	M4
87	L	M	H	VH	M4
88	L	H	L	VH	M4
89	L	H	M	VH	M4
90	L	H	H	VH	M4
91	M	L	L	VH	M4
92	M	L	M	VH	M4
93	M	L	H	VH	M4
94	M	M	L	VH	M4
95	M	M	M	VH	M4
96	M	M	H	VH	M4
97	M	H	L	VH	M4
98	M	H	M	VH	M4
99	M	H	H	VH	M4
100	H	L	L	VH	M4
101	H	L	M	VH	M4
102	H	L	H	VH	M4
103	H	M	L	VH	M4
104	H	M	M	VH	M4
105	H	M	H	VH	M4
106	H	H	L	VH	M4
107	H	H	M	VH	M4
108	H	H	H	VH	M4

Appendix B. Sample rules for second stage of the dynamic bandwidth allocator (FLC)

S. no.	Index voice	Index audio	Index data	Index video	Voice BW	Audio BW	Data BW	Video BW
1	M1	M1	M1	M1	L	VL	VL	VL
2	M1	M1	M2	M1	L	VL	L	VL
3	M1	M1	M3	M2	L	VL	M	L
4	M1	M1	M4	M3	L	VL	H	LM
5	M1	M2	M1	M4	L	L	VL	M
6	M1	M2	M3	M1	L	L	M	VL
7	M1	M2	M4	M2	L	L	H	L
8	M1	M3	M1	M3	M	L	IL	M
9	M1	M3	M2	M4	M	L	MH	LM
10	M1	M3	M4	M1	M	L	H	VL
11	M1	M4	M1	M2	M	M	IL	L
12	M1	M4	M2	M3	M	L	MH	LM
13	M1	M4	M3	M4	M	L	MH	LM
14	M2	M1	M1	M1	L	VL	VL	VL
15	M2	M1	M2	M2	L	VL	L	L
16	M2	M1	M3	M3	L	L	L	M

Appendix B (*Continued*)

S. no.	Index voice	Index audio	Index data	Index video	Voice BW	Audio BW	Data BW	Video BW
17	M2	M1	M4	M4	M	VL	H	LM
18	M2	M2	M2	M1	L	L	L	VL
19	M2	M2	M3	M2	L	L	M	L
20	M2	M2	M4	M3	M	L	M	LM
21	M2	M3	M1	M4	M	L	VL	M
22	M2	M3	M3	M1	M	M	MH	VL
23	M2	M3	M4	M2	M	M	H	L
24	M2	M4	M1	M3	M	L	IL	M
25	M2	M4	M2	M4	M	L	MH	LM
26	M2	M4	M4	M1	M	H	H	VL
27	M3	M1	M1	M2	H	VL	IL	L
28	M3	M1	M2	M3	M	VL	L	M
29	M3	M1	M3	M4	H	L	M	LM
30	M3	M2	M1	M1	H	L	IL	VL
31	M3	M2	M2	M2	M	L	L	L
32	M3	M2	M3	M3	M	L	M	LM
33	M3	M2	M4	M4	M	L	M	LM
34	M3	M3	M2	M1	H	M	LM	VL
35	M3	M3	M3	M2	H	M	MH	L
36	M3	M3	M4	M3	H	L	MH	LM
37	M3	M4	M1	M4	H	L	IL	M
38	M3	M4	M3	M1	H	H	MH	VL
39	M3	M4	M4	M2	H	H	H	L
40	M4	M1	M1	M3	H	VL	VL	M
41	M4	M1	M2	M4	H	VL	L	M
42	M4	M1	M4	M1	H	VL	H	VL
43	M4	M2	M1	M2	H	L	VL	L
44	M4	M2	M2	M3	H	L	M	LM
45	M4	M2	M3	M4	H	L	M	LM
46	M4	M3	M1	M1	H	M	IL	VL
47	M4	M3	M2	M2	H	M	LM	L
48	M4	M3	M3	M3	H	L	M	LM
49	M4	M3	M4	M4	H	L	M	LM

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