# ATM Infrared Wireless LANs: A Proposed Architecture

*Pavlos Theodorou, Jaafar M. H. Elmirghani, and Robert A. Cryan*
*University of Northumbria at Newcastle*

**ABSTRACT** As infrared wireless LANs for in-building applications become more popular because of their many advantages, it is of interest to design such LANs to support the ATM protocol and therefore provide wireless access to fixed ATM networks. The architecture of such a LAN is presented in this article. Emphasis is placed on the protocol stacks of the model to provide seamless operation with the wired network, on the cellular topology, and on the MAC protocol. Under the proposed topology the system performance is revealed in terms of packet dropping probability, average access delay, channel throughput, and statistical multiplexing gain for a range of system parameters.

Over the past decade considerable interest has been given to wireless indoor communications to support multimedia services and therefore to extend the optical wired broadband integrated services digital networks (B-ISDNs) into a wireless environment [1–3]. In B-ISDNs the user information originating in any arbitrary format and comprising continuous or variable-bit-rate voice, image, and data is converted and presented to the network as a sequence of asynchronous transfer mode (ATM) cells. The services supported generally require a large bandwidth; as such, the choice of carrier frequency for a wireless ATM system would need to be between very short wavelength radio bands and infrared (IR). The majority of currently proposed wireless communication services are based on radio waves; however, IR wireless technology is receiving increased attention for indoor applications. IR communication has significant differences from radio communication; some may be advantageous and others disadvantageous when implementing indoor wireless LANs. Radio systems are already used in cellular phone networks, but such links are not capable of delivering the high bandwidths required for multimedia broadband platforms. By contrast, the very high carrier frequency associated with IR systems (300 THz at 1 μm) promises communication channels with abundant bandwidth. Because IR transmission does not interfere with existing radio systems, it does not fall under any regulation of the Federal Communication Commission (FCC), and therefore there is no need for any license, which can help keep the cost low. Moreover, the fact that IR signals do not penetrate opaque objects (e.g., walls) enables an indoor wireless network to provide both a considerable degree of privacy within an area and a very large spatial bandwidth at the same spectrum since the same IR channel can be reused in adjacent areas. Because the only practical technique to transmit information over IR wireless links is intensity modulation with direct detection (IM/DD), the short carrier wavelength and large-area square-law detector lead to efficient spatial diversity that prevents multipath fading. However, distortion due to multipath propagation is present.

The fact that IR radiation does not propagate through walls and other opaque objects can increase cost and system complexity. Outdoor applications using diffused IR links are almost impossible due to intense solar radiation that induces noise in the receiver. Also, in many indoor environments there exists intense infrared noise that arises not only from sunlight but other artificial light sources such as incandescent and fluorescent light [4]. This noise significantly reduces the signal-to-noise ratio (SNR) and calls for high transmitter powers. Unfortunately, high powers can be problematic due to eye safety standards that have to be obeyed and are governed by International Electrotechnical Commission (IEC) standards. Diffuse IR links also suffer from technical problems in terms of multipath and intersymbol interference (ISI), which limit the data rate to speeds much lower than those of the line-of-sight (LOS) topology. Typically, data rates up to 100 Mb/s can be achieved [5–6]. Although these are much lower than what can be achieved in an LOS IR link, the data rates for diffuse (fully mobile) IR systems are much higher than those of radio systems.

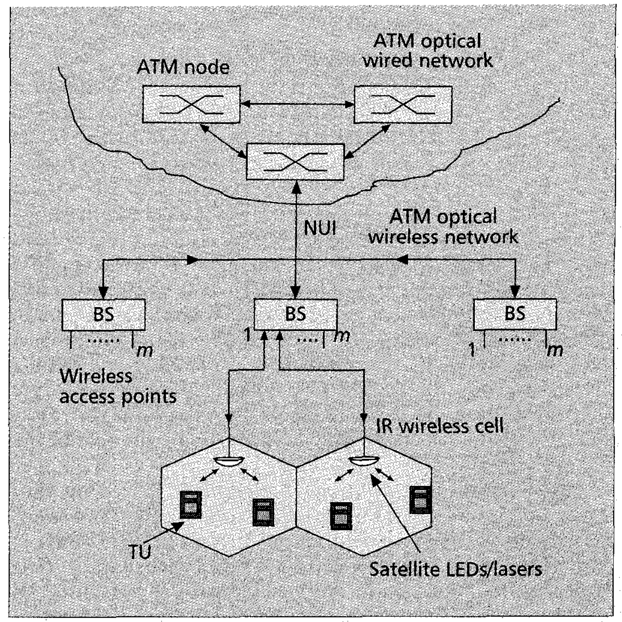
Diffuse IR technology is still in its infancy (however, it is included as part of the IEEE 802.11 standard for wireless LANs); because of this, there are very few manufactures of this technology. Recent research work suggests that using new techniques, it is possible to enhance the technology of diffused IR communication significantly as far as the physical layer is concerned. However, the need arises for research into higher layers in these systems to enable the support of multimedia services, such as ATM, and therefore to extend B-ISDNs into indoor wireless environments. Since ATM was designed for a medium whose bit error rate (BER) is very low, namely optical fibers, the challenge is to extend the ATM capabilities into an IR diffused channel, a medium that does not promise the same quality of service (QoS) as optical fibers. This is because an IR diffused channel is subject to several impairments arising from the noise due to ambient and artificial light, multipath propagation, time-varying ISI, co-channel interference (in a multicell topology), inherent user mobility, and unavoidable changes caused by motion of the surrounding environment. As in the case with radio communication, a data link control (DLC) layer can be introduced in the IR wireless protocol stack for error recovery [1, 2]. It will possibly involve the employment of forward error correction (FEC) for time-sensitive services and automatic repeat request (ARQ) for time-insensitive services. Although several hybrid ARQ/FEC techniques have been reported in the literature concerning radio wireless systems, fundamental differences between radio and the diffused IR channel may change many of the boundary conditions. Hence, more research under these conditions is required. IR links are capable of delivering much higher bandwidths than radio links, yet their capacity is much less than that of optical wired communication systems due to the impairments mentioned above. Therefore, a medium access control (MAC) different than that for the wired network must be employed for the control of the wireless bandwidth to mobile TUs. Finally, to support mobility when a TU is moving from one cell to another, adding mobility functions in the protocol stack is indispensable.

## A Proposed Scenario for ATM IR Wireless LANs

A possible scenario for such a LAN is illustrated in Fig. 1. To cover a large area, such as an open plan office, it needs to be divided into cells; a common way to study such a topology is to use hexagonal cells. To prevent co-channel interference the MAC protocol at the BS ensures that adjacent wireless cells transmit at different times. The channel reuse factor is given by $J = k^2 + l^2 + kl$, where $k$ and $l$ are the coordinates at which the first nearest neighbor is located. Given the radius $r$ of a cell as measured from its center to a vertex, the distance between a cell and its nearest neighbors is the reuse distance and is given by $d = r\sqrt{3n}$. In this article the radius of a cell is taken to be 3 m, a satisfactory size between the interference-limited region and noise-limited region according to the results in [7]. To reduce complexity between the wired and wireless networks as well as processing time, it is important that the protocol stack at the base station (BS) is designed in a way that provides seamless interworking with the wired ATM network. Also, the BS is responsible for providing and managing the mobility functions for the TUs when they move from one cell to another. At any given time a TU belongs to exactly one cell and is said to be associated with the BS at which the wireless access point in this cell is connected. A BS can support up to a certain number ($m$) of wireless access points. Therefore, this scenario is attractive in that the mobility functions and handoff procedures can be allocated at the BSs. This feature makes the proposed wireless system less dependent on the fixed ATM node.

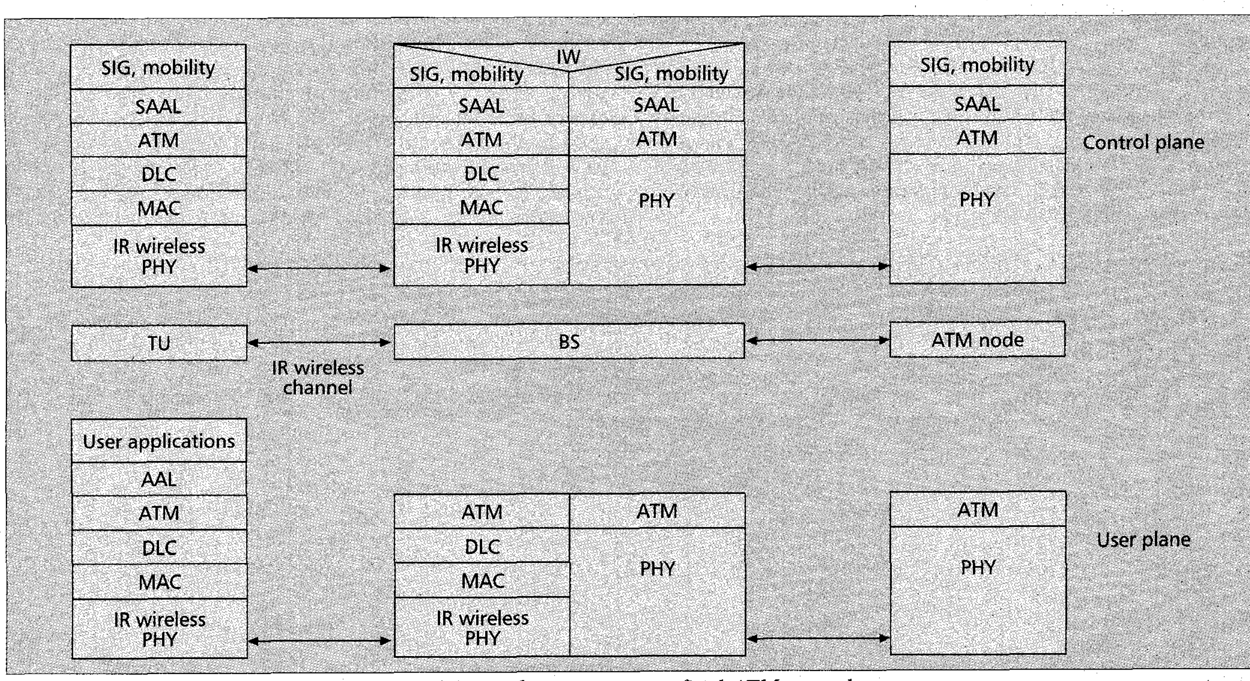
To relax the system of co-channel interference the first nearest neighbor is located at (1, 2), resulting in a reuse channel factor $J = 7$ and a distance between two neighboring cells of 13.7 m. This distance is large enough to ignore co-channel interference because it is much lower than the noise produced by the ambient and artificial light sources under this topology [7]. Finally, we assume that the on-off keying (OOK) transmission bit rate is low enough, ″ 10 Mb/s, so that ISI effects due to multipath phenomenon can be ignored [5].




■ **Figure 1.** *A scenario for an indoor wireless ATM network using IR diffused links.*

## The Protocol Reference Model

The protocol reference model illustrating how the wireless network can be integrated with the fixed ATM network is shown in Fig. 2. It is composed of the user plane, which is responsible for providing user information transfer, and the control plane, which is responsible for setting up and releasing a connection. The user plane does not involve any ATM




■ **Figure 2.** *The protocol stacks architecture of the wireless system into a fixed ATM network.*

adaptation layer (AAL) at the BS; therefore, ATM cells are transported between ATM nodes and TUs as transparently as possible. In other words, this kind of "native approach" treats ATM cells as a payload data field for the DLC layer. The AAL, which is found in the protocol stack of a TU, is designed to act as the interface between user applications and the ATM layer. As such, it is expected to enhance the service provided by the ATM layer, based on the specific requirements of various applications such as voice, video and data. The ATM layer consists of virtual channel (VC) and virtual path (VP) levels, and is responsible for the routing of the cells using identification fields in the cell header. Among the other tasks, the transmitting ATM layer adds the 5-byte header on the ATM cell; the receiving ATM layer processes this header, then strips it away before passing the rest of the cell to the AAL.

Below the ATM layer the DLC and MAC layers are allocated, and, as mentioned in previous sections, these two layers are used to enhance the physical layer transport capability. In particular, the DLC layer is introduced to improve the performance degradation due to the high bit error in a diffused IR link, while the MAC layer is responsible for sharing the capacity of the wireless IR channel according to the bandwidth explicitly required by each TU. Connections, or VCs, between TUs and the rest of the network are established and released with signaling procedures (based on Q.2931 or any ATM Forum user–network interface, UNI, signaling protocol) in the SIG layer of the control plane.

The BS terminates control signaling to/from TUs and carries out call admission control (CAC) then sends the SETUP message to the ATM node after interworking at the BS. A connection is granted when the traffic contract (contains information such as peak cell rate, maximum cell delay, sustainable cell rate, burst tolerance, etc.) of a TU is examined, revealing that the connection can be supported through both networks (wired/wireless) at its required QoS level. The BS based on the information of the traffic contract carries out IR channel control for the TUs. The layer below the SIG is the signaling ATM adaptation layer (SAAL), and supports the transport of the signaling protocol and mobility function protocol.

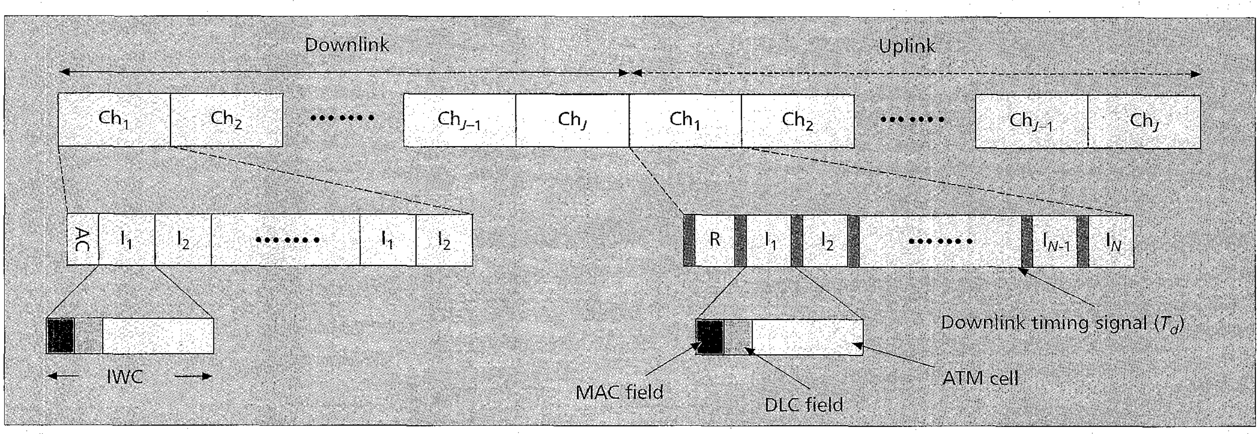
The structure of the overall protocol does not call for the ATM node to carry any function related to the IR channel control. However, the BS gets more complex because it needs to terminate the SAAL, and to carry out CAC, IR channel control, and mobility management. Thus, the BS is equivalent to a small ATM switch, which facilitates adding the proposed system to existing ATM networks.

## The Medium Access Control (MAC) Protocol

Recently, many studies were conducted on time-division multiple access (TDMA) protocols to support integrated services on radio wireless systems. Among them packet reservation medium access (PRMA) [8], a modified R-ALOHA protocol, exploits the talkspurts-silence statistical character of speech streams by means of a speech activity detector to serve more speech TUs. Packet-based transmission facilitates PRMA to accommodate information from diverse sources and to act in harmony with other packet networks such as ATM.

There has been growing interest in multiple access techniques for optical wired LANs using optical codes such as optical orthogonal codes, prime codes, $2^n$ prime codes, and $2^n$ extended-prime codes. We therefore, to prevent collisions when TUs try to reserve a slot in our proposed system, utilize such optical codes. After CAC each TU is dispatched with a unique address code, considered the VP identifier (VPI). Each unique address corresponds to one optical code. By means of a correlative detector at the BS, the characteristics of such a code can be used to distinguish the code of a TU and therefore prevent collisions. Because the optical code is not used to modulate information and its chip rate is the same as the channel bit rate, the system is time-division multiplexed (TDM) and not spread spectrum. The choice of a particular optical code depends on factors such as power loss for the mobile TUs, system complexity, cost, and capacity of the network.

Figure 3 depicts the time organization of the uplink and downlink frame of the MAC protocol. Each frame is divided into $J$ channel frames; $J$ is the channel reuse factor, and each channel frame is divided into slots. An uplink channel frame is composed of a reservation (R) slot and $N$ information (I) slots. TUs request a reservation by transmitting their codes through the R slot. When the requests are more than the free slots in the channel frame, the BS allocates them to TUs with a priority order. Because the chip rate of the optical codes is the same as the IR channel bit rate, the duration of an R slot is defined by the maximum number of TUs to be served in a cell, where an I slot is the transmission time for an IWC. TUs start to transmit in their own time slots by means of short downlink timing signals ($T_d$). A downlink channel frame is composed of an acknowledgment (ACK) slot and $N$ I slots. An ACK slot is used by the BS to broadcast messages for each downlink slot as well as reservation messages after determining the TUs that applied for slot reservation. The duration of an ACK slot is set according to the maximum number of TUs served in a cell.




■ **Figure 3.** *Timing organization for the uplink and downlink frames of the MAC protocol.*

*Here, for convenience*, we assume that the duration of an ACK slot is the duration of an R slot plus the short downlink timing signals in an uplink channel frame.

Once a TU has gained access, it then transmits the remaining message packets (if any) on the reserved slot (or slots) in each uplink frame. After sending the last packet, the TU relinquishes the corresponding slot/slots. Transmission of slot allocation messages in the ACK slot allows TUs to switch themselves to power saving mode without loosing synchronicity or any messages.

If there are no free slots when a TU applies for a reservation, a delivery delay occurs for the packets waiting to be transmitted. For real-time services such as voice and video, terminals will drop the packets that exceed a maximum delay ($D_{max}$). For example, speech packet dropping probability ($P_{drop}$) at which speech quality degradation is almost imperceptible must be less than 1 percent [8]. The MAC protocol allocates a fixed number of slots in each channel frame and therefore in each cell. However, dynamic allocation of the slots in each channel will significantly improve the performance of the system in terms of bandwidth allocation according to the instantaneous needs in each channel or cell.

## ATM Traffic Models

In this article we only consider real-time traffic (RTT), and there are two alternative approaches for bandwidth allocation for such traffic: deterministic and statistical multiplexing. In deterministic multiplexing, each connection is allocated its peak bandwidth and therefore the cells are transferred in constant bit rate (CBR) mode. Doing so causes large amounts of bandwidth to be wasted for bursty services, particularly for those with large peak-to-average bit rate ratios. By contrast, in statistical multiplexing the amount of bandwidth allocated to a variable bit rate (VBR) source in the network is less than its peak, but necessarily greater than its average bit rate.

Modeling VBR sources for ATM networks, thus estimating the value of the statistical bandwidth, has received much attention. A well-known approach to model VBR voice and video sources is the ON/OFF model. The rationale for such a model is that a source is either in an OFF state, transmitting at zero bit rate, or in an ON state, transmitting at its peak rate. Such a source model has the advantage of being both simple and flexible, and can be used to represent connections ranging from bursty to continuous bitstreams. It has been shown that a speech source has talkspurt and silence patterns (ON/OFF, respectively) exponentially distributed [8]. Given the mean duration of talkspurts ($t_1$), silences ($t_2$), and channel frame duration $T_{cf}$, the transition probability from talking to silent state during a channel frame is $\gamma = 1 - \exp(-T_{cf}/t_1)$, while the transition probability from silent to talking state is $\sigma = 1 - \exp(-T_{cf}/t_2)$. Other description parameters for the ON/OFF speech source is the activity ratio $\alpha = t_1/(t_1 + t_2)$ and source burstines $\beta = (t_1 + t_2)/t_1$.

A VBR video source can be described as a superposition of $X$ independent identical ON/OFF sources; however, the calculation of the statistical bandwidth (or equivalent capacity) to guarantee the QoS becomes very complicated. Despite the major effort of researchers to accurately model VBR video sources, characterizing the behavior of the output process of an encoder is still an open research question.

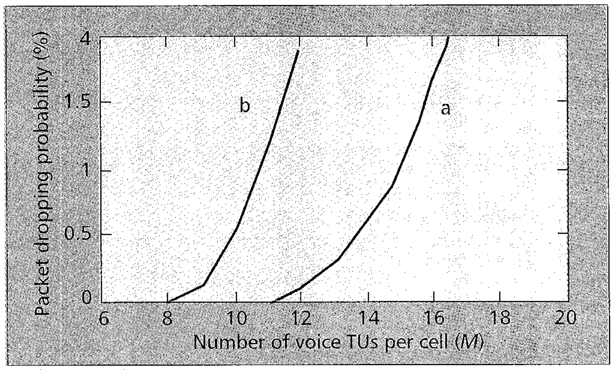
## Performance Investigation

In this section we investigate the performance of the system when a fixed number of VBR voice and video sources are to be served in a cell. The analysis considers only statistical allocation of the time slots. As mentioned previously, using a large channel reuse factor and a channel bit rate no greater than 10 Mb/s, we can ignore co-channel interference and ISI in the physical layer. Consequently, the performance for given system parameters depends on factors such as packet arrival pattern, IWC and MAC structure. To reduce service delay for time-sensitive services, especially when a large channel reuse factor is employed, it is necessary to keep the round-trip duration of each channel short. Therefore, we can consider the time slots in each channel frame as a bunch of parallel servers. The interarrival times and service times obey the exponential distribution; equivalently, the arrival rate and service rate follow a Poisson distribution. Under these features the MAC protocol can be modeled as an $M/M/N/^\circ /M$ queuing model consisting of exponentially distributed durations ($M$) of all spurts and gaps, exponential service ($M$), $N$ parallel servers, infinite storage, and $M$ TUs.

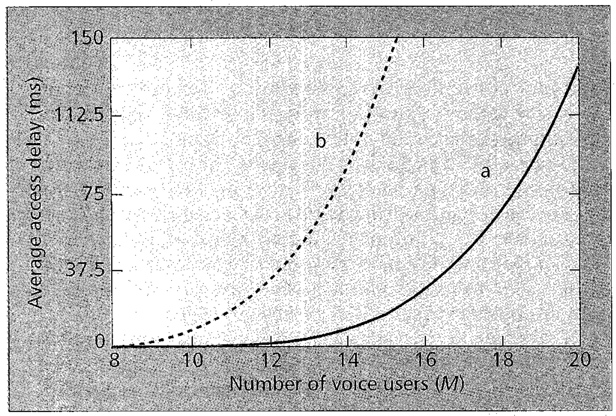
The number of time slots (servers) $N$ per channel frame is given by $N = \text{int}[(R_c T_{cf})/(2JT_{cf}R_s + H)]$, where int[$y$] is the largest integer smaller than or equal to $y$, $R_c$ is the channel bit rate, $R_s$ is the source bit rate, $J$ is the channel reuse factor, $H$ is the header of an IWC, $T_{cf} = (T_f/J) - (\text{ACK}/R_c) - [(NT_d)/R_c]$ is the duration of a channel frame, and $T_f$ is the uplink/downlink frame duration. The integer 2 comes from the fact that we want both uplink and downlink to have the same number of slots. Transmission delays between a TU and the wireless access point of the cell are negligible. The state probability of the queuing model, which defines the probability of the number of TUs in the system at any given time, can be derived using the birth-death theory for finite source queues. Then the system performance in terms of packet dropping probability, throughput, average access delay, and statistical multiplexing gain is evaluated for a range of system parameters. Table 1 shows the various parameters of the system.

| Variable | Notation | Value |
|---|---|---|
| Channel bit rate | $R_c$ | 10 Mb/s |
| Speech peak bit rate | $R_s$ | 64 kb/s |
| Video peak bit rate (coded) | $R_v$ | 320 kb/s |
| Uplink/downlink frame duration | $T_f$ | 3.1 ms |
| R slot duration | $R$ | 280 bits |
| Downlink timing signal | $T_d$ | 4 bits |
| Speech mean ON duration | $t_1$ | 1 s |
| Speech mean OFF duration | $t_2$ | 1.35 s |
| Speech cell maximum time delay | $D_{max}$ | 20 ms |
| Video mean ON duration | $Vt_1$ | 33 ms |
| Video mean OFF duration | $Vt_2$ | 67 ms |
| Video cell maximum time delay | $VD_{max}$ | 150 ms |
| Channel reuse factor | $J$ | 7 |
| Number of identical superimposed ON/OFF sources to model the video traffic | $N_s$ | 5 |
| Header of an IWC (compressed ATM header, DLC field, MAC field, guard band) | $H$ | 70 bits |

**■ Table 1.** *System parameters.*




■ **Figure 4.** *Packet dropping probability vs. number of voice TUs per cell when: a) zero video TUs are in the system; b) one video TU is in the system.*




■ **Figure 5.** *Average access delay vs. number of voice TUs per cell when: a) zero video TUs are in the system; b) one video TU is in the system.*

The area of a hexagonal cell is given by $A = \sqrt{27}r^2/2$, with $r = 3$ m; then $A = 23$ m$^2$. The bandwidth allocated for the uplink/downlink in each cell is given by $B_c = (T_{cf}R_c)/2T_f$. This bandwidth can be used to serve CBR, VBR, and ABR/UBR ATM services. For available/unspecified bit rate (ABR/UBR) services, such as text, idle I slots (slots not used by VBR services) are assigned by means of a dynamic slot allocation MAC protocol.

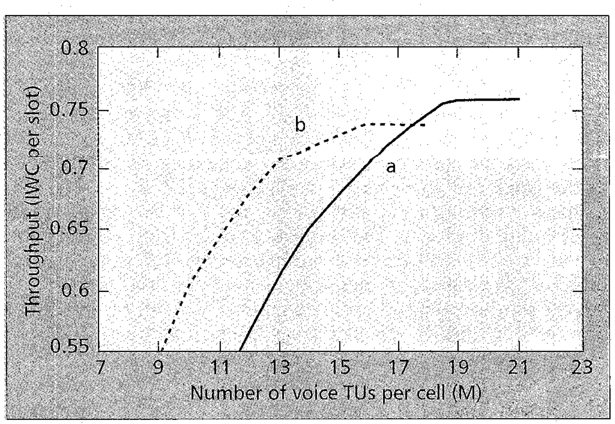
For voice sources the probability of more than one transition, from talking to silent or vice versa, is zero during one channel frame. Talkspurt duration is equivalent to message length or service time of each TU; likewise, gap duration is equivalent to interarrival time of messages. The MAC protocol is designed so that the uplink channel frame rate is identical to the arrival rate of periodic (voice) packets. Considering a slow speech detector the average duration of speech activities is taken from [9] and that of video activities (e.g., videoconference) is taken from [10]. In [10] the statistical bandwidth for the video source has been calculated, and it was found to be about $C = 130$ kb/s for a cell loss probability of $P_{loss} \approx 10^{-4}$. Thus, such a video source would require $(2T_fC)/K$ time slots per channel frame, where $K$ is the number of bits conveyed per time slot.

Figure 4 shows the $P_{drop}$ of the system versus the number of voice TUs ($M$) per cell for various numbers of video sources in the same cell. In both cases it is observed that increasing $M$ results in an increase in $P_{drop}$; this is because more voice packets are to be served by the fixed number of time slots in a channel frame. Curve a depicts the $P_{drop}$ when only voice TUs are in the system; for $P_{drop} = 1$ percent the system is capable of supporting up to 15 TUs simultaneously. If $M$ is above 15, the QoS is not satisfied. When one video source is in the system $M$ is less than 15 for $P_{drop} = 1$ percent, this is depicted by curve b.

Although the performance of a VBR video source may be satisfied when its statistical bandwidth is assigned, the statistical bandwidth does not exactly describe the traffic of the source. In order to precisely allocate the bandwidth for such a source and, therefore, to increase the multiplexing gain of the system, dynamic slot allocation based on the instantaneous capacity requirements of the TUs must be utilized. An algorithm for dynamic slot allocation of the MAC protocol is beyond the scope of this article.

As the number of TUs increases the average access delay ($D$) also increases; this is depicted in Fig. 5. For both curves there is a particular number of $M$ where $D$ starts to increase rapidly. For example, looking at curve a, for $M$ less than 12 the system is in an insensitive region and variation of $M$ has no significant effect. However, when $M$ exceeds 12 the value of $D$ increases rapidly, and the system is said to be in a sensitive region. In this region, the addition of even one more TU dramatically changes the value of $D$. As $D$ increases, voice packets have to wait more than the time limit delay $D_{max}$ to be served, and consequently $P_{drop}$ increases. Figure 6 depicts the system throughput ($\eta$) against $M$. For both cases $\eta$ gradually increases as $M$ increases; eventually it reaches its maximum, where it remains constant even though $M$ keeps increasing. That indicates good system stability.

The statistical multiplexing gain ($G$) of the MAC protocol denotes how many voice TUs can be served per time slot for a given QoS. With only voice TUs in the system and $P_{drop} = 1$ percent, the protocol exhibits a statistical gain of 1.7, which means that in a time slot 1.7 voice sources can be accommodated. When one video sources is in the system, $G$ drops due to the decrease of the number of available slots for voice traffic in a channel frame.




■ **Figure 6.** *System throughput versus number of voice TUs per cell when: a) zero video TUs are in the system; b) one video TU is in the system.*

## CONCLUSIONS

In this article we consider the design of an indoor IR wireless LAN that supports the ATM protocol and therefore can provide wireless access to fixed B-ISDNs. A proposed scenario for such a LAN is presented, and the protocol stacks of the model to provide seamless operation with the wired network, the cellular topology, and the MAC protocol are examined. Given the basic ATM traffic models, we evaluate the system performance for a fixed number of VBR voice and video TUs in a cell over a range of system parameters. Since the channel reuse factor and the chosen bit rate prevent co-channel interference and ISI, respectively, the performance of the system depends mainly on the MAC protocol structure. Original numerical results reveal the system performance in terms of packet dropping probability, system throughput, average access delay, and statistical multiplexing gain.

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## BIOGRAPHIES


PAVLOS THEODOROU (pavlos.theodorou@unn.ac.uk) received his B.Eng. in electrical engineering from the Technological Education Institution (T.E.I.), Thessaloniki, Greece, in 1992. In the same year he joined the Greek National Army, where he specialized in military communication systems. In 1996 he received an M.Sc. in optoelectronic and communication systems from the University of Northumbria, Newcastle, U.K. He is currently engaged in research at the University of Northumbria, leading to a Ph.D. in the area of multimedia optical wireless networks.

JAAFAR M. H. ELMIRGHANI [M] (j.elmirghani@unn.ac.uk) received his Ph.D. in electrical engineering from the University of Huddersfield, United Kingdom, in 1993. In 1991 he worked as a research engineer with the High Speed Optical Fibre Communications group at GEC Hirst Research Centre, U.K. From January to September 1994 he was a post-doctoral research fellow at Manchester Metropolitan University, U.K. Subsequently he joined the University of Northumbria, U.K., as a senior research fellow. In Februray 1995 he was appointed senior lecturer and in February 1998 a principal lecturer, all at the University of Northumbria. He is currently also BT Reader in Telecommunication Systems. He is a member of the IEE, and currently serves as secretary to the IEEE UK&RI Communications Chapter. During 1996 and 1997 he was secretary of the IEEE ComSoc Signal Processing and Communications Electronics (SPCE) Technical Committee and currently serves as vice chair of the same committee. He is an editor of *IEEE Communications Magazine* and has been and is on the technical program committee of several IEEE ICC/GLOBECOM conferences. In particular he will act as chair of the IEEE GLOBECOM '99 Advanced Signal Processing for Communication Symposium. In 1995 he was general chair of the SPIE Chaotic Circuits for Communications Conference and was co-chair of the SPIE Wireless Data Transmission Conference. He was a member of the technical program committee for SPIE Optical Wireless Communication in 1997 and 1998. He currently also serves as technical editor for the Journal of Optical Communications. He has published over 100 technical papers, co-edited *Photonic Switching Technology — Systems and Networks* (IEEE Press, 1998), and has research interests in optical communication systems, signal processing, and communication theory.

ROBERT A. CRYAN [M] (bob.cryan@unn.ac.uk) received a first class honours degree and a Ph.D. in electrical and electronic engineering from the University of Huddersfield in 1986 and 1992, respectively. He is an active member of the IEE, where he served as chair of the Electronics and Control Committee. He is also a member of SPIE and was on the program committee for the OE/FIBERS '92 Multigigabit Fiber Communications Conference and co-chair for SPIE's Symposium on Fibre Networks '93. He was also co-chair of the SPIE All-Optical Communication Systems: Architecture, Control and Network Issues Conference in 1995, 1996, and 1997. Between 1992 and 1994 he was a principal lecturer in the Department of Electrical and Electronic Engineering at Manchester Metropolitan University. Currently he is professor of electrical engineering and head of the School of Engineering, University of Northumbria at Newcastle. He has research interests in optical communications and signal processing.