Multiservice Communications over TDMA/TDD Wireless LANs*

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Abstract. In recent years, several wireless LAN technologies making use of TDMA/TDD MAC have been designed. In this type of environment, there is the need of a central controller responsible of allocating the bandwidth among all the active mobile terminals. In order to properly carry this task, the use of simple but effective signaling and bandwidth allocation mechanisms is a must. In this way, the mobile terminals can let know the controller of their needs in terms of the bandwidth to be allocated. In turn, the central controller has to properly allocate the available bandwidth among all the competing mobiles taking into account their QoS requirements. In this work, we undertake the design and performance evaluation of the signaling protocols and bandwidth allocation mechanisms. We have paid particular attention to study the amount of overhead introduced by these mechanisms: an important feature when designing control mechanisms for wireless environments. We validate the effectiveness of our proposed schemes when supporting a multi-service environment comprising four types of services: video, voice, best-effort and background.

Keywords: TDMA/TDD, WLAN, QoS, Resource Request, HIPERLAN/2

1 Introduction

Nowadays wireless local networks represent an alternative to wired LANs. Current wireless LANs operate at transmission rates able to support all types of applications: data, voice, video, etc. It is widely recognized that one of the main advantages of WLANs is their great flexibility: the wire is suppressed allowing users to freely connect to the network. Standardization efforts have resulted on the definition of wireless LANs standards whose one of their main aims is to guarantee the interoperability among equipment developed by different vendors. To date, two of the most important wireless LANs standards are: the IEEE 802.11 standard [1] and the HIPERLAN/2 standard [2] developed by the ETSI (European Telecommunications Standards Institute). Furthermore, the development of wireless communications is now spanning into the area of metropolitan area networks (MANs) where standards, such as, IEEE 802.16 [3], are under study.

The ETSI within the framework of its project BRAN (Broadband Radio Access Networks) has developed various standards for wireless LAN and MAN. One of these is the HIPERLAN/2 standard, which operates in the band of 5 GHz with transmission rates from 6 up to 54 Mbit/s. HIPERLAN/2 supports both operating modes; infrastructure and ad-hoc modes. When operating under the infrastructure mode, the standard distinguishes between two types of devices: the Access

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Point (AP) and the Mobile Terminal (MT). The AP is responsible of providing connectivity with the core network as well as of adapting the users requirements by taking into account the characteristics of the core network and the services offered by HIPERLAN/2. On the other hand the AP takes care of the distribution of the resources and the coordination of all the MTs located within the cell.

The fact that the access control is taken in charge by a unique device, facilitates the design and deployment of mechanisms capable of satisfying the QoS requirements of various applications. Based on the MAC protocol defined by the HIPERLAN/2, it is possible to build up QoS mechanisms capable of providing the guarantees needed by various applications. The HIPERLAN/2 also defines the structure and sequence of the control messages between the MTs and the AP. However, the HIPERLAN/2 does not define the specifics regarding the timing and numerical values of the system parameters, such as the bandwidth to be reserved for a specific type of connection. Nor it is object of the standard the description of the specifics of the algorithm to request and grant bandwidth. Therefore, HIPERLAN/2 has intrinsical characteristics allowing it to support several traffic classes with different QoS requirements.

One of the first issues to solve when developing a structured set of resource allocation mechanisms is how to make available the applications requirements to the AP. In this paper, we show that making use of a set of resource request mechanisms designed taking into account the requirements and characteristics of the applications can indeed improve the network performance.

Another major issue to be addressed is the definition of the resource granting mechanisms. In TDMA/TDD wireless LANs, such as HIPERLAN/2 and IEEE 802.16 , the AP has to inform the MTs on the bandwidth assigned to each active connections. This information is transmitted to the MT by including it into the frame. It follows that as the number of active connections increases, the frame overhead will increase accordingly. In order to make a proper use of the network bandwidth, it is important to limit the amount of overhead. In this work, we will analyze various bandwidth allocation schemes. This refers to the impact of such mechanisms on the structure of the frame, and in particular on the amount of overhead introduced into it.

The main objectives of this work can be simply stated as follows. First, we should define a taxonomy of the various service classes to be supported. This taxonomy will allow us to define the resource request mechanisms according to the needs of each service class. Second, we should show that by properly matching the proposed resource request mechanisms to the various types of applications under consideration, our schemes should fulfill the applications requirements and enhance the overall system performance. Third, we should show that the overhead introduced into the frame structure can be significantly reduced by making use of a bandwidth allocation mechanism that takes into account the traffic characteristics of the applications.

The article is organized as follows. Section 2 provides a short overview of the HIPERLAN/2 standard. In Section 3 we review the related bibliography on the area of MAC algorithms for TDMA/TDD wireless networks. The QoS framework presented in this work is described in Section 4. The results of our performance evaluation study are given in Section 5. Finally, Section 6 concludes the paper.

2 HIPERLAN/2 MAC Protocol

The HIPERLAN/2 MAC protocol [4] is based on a dynamic TDMA/TDD scheme with centralized control, using as logical transmission unit frames of $2\,ms$. Given that the allocation of the frame resources to each MT is made by the AP, the requirements of the application resources have to be known of these entities, which are responsible of allocating the available resources according to the user needs. Towards this end, each MT has to request to the AP the required resources by issuing a *Resource Request (RR)* message, while the AP informs the MT of the positive outcome by using a *Resource Grant (RG)* message.

The HIPERLAN/2 frame is divided into four phases, each phase being composed by a group of transport channels. A transport channel is a logical entity and its classification depends on the type of data that it conveys.

The phases of a frame are:

1. *Broadcast phase*: this phase is used for the communications taking place on the downlink. It contains the configuration parameters of the frame, the resource grant (RG) messages for each active connection in the frame, and the information regarding the number of collisions having occurred in the previous frame.
2. *Downlink phase & Uplink phase*: these phases are formed by a group of PDU trains, which are formed by a preamble and a variable number of Short transport CHannels (SCHs) and Long transport CHannels (LCHs) dedicated to each one of those connections with resources granted in the frame. The LCH channels transport user data and the SCH channels convey error control or resource request messages.
3. *Random Access (RA) phase*: consists of a number of Random CHannels (RCH), which can be used for transmission of resource request messages. A contention process based on a Slotted-ALOHA scheme is used to access the RCH channels.

It is important to note that not all of the transport channels have the same size and that this one depends on the channel type [4].

3 Related Work

Given the central role played by the MAC algorithms on enabling the provisioning of the QoS requirements to the various applications, there has been a large number of studies focusing on the design and evaluation of QoS-aware MAC protocols for TDMA/TDD networks. In [5], the authors describe a resource reservation protocol. In this protocol, the MTs issue the first request by using a contention-based protocol, similar to S-ALOHA. Once having been allocated a number of channels, the data packets convey, via "piggybaking', the following MT's resource requests. Whenever, a MT does not have anymore packets to transmit, the resources (bandwidth) are freed. At a latter time, when the MT becomes once again active, this one has to start the reservation process by issuing a first request via the contention process. The MASCARA algorithm [6] makes use of a resource request mechanism similar to the one described in [5]. The mechanism makes use of a "scheduler" based on the "token bucket" scheme to distribute the resources among the active connections. One of the main drawbacks of the proposed request mechanisms in [5] and [6] is that they have been built around a contention-based MAC. It is well known that the performance of these mechanisms severely degrades as the number of active connection increases. This may the allocation mechanism prone to delay and losses; an undesirable condition when developing QoS mechanisms. Furthermore, the "piggybacking" mechanism can not be implemented by HIPERLAN/2. More recent works [7],[8], focusing on the HIPERLAN/2 standards, have proposed several algorithms to distribute the bandwidth resources to different types of connections. Within a given type of connection, the bandwidth is allocated by taking into account the status of the buffers or the connection parameters provided at the time of establishing the connections. However, the authors do not specify the resource request (signaling) protocol. We can implement a resource request protocol with a "polling" mechanism or making use of a fixed number of control channels as proposed in [9, 10]. The drawback of the latter approach is the excessive overhead introduced as the number of uplink connections increases. Another proposed option, shown in [9, 10], allocates a fixed number of control channels, whose associated connection can use to further request more resources (bandwidth).

4 A QoS Framework

4.1 Service Classes and Resource Request Mechanisms

Regarding the underlying HIPERLAN/2 mechanisms, we focus on a two level hierarchy: connection establishment procedures and signaling primitives. Regarding the former, we enable the provisioning of contracted and non-contracted services. These two types of services relate to the provisioning of the bandwidth required to convey the signaling primitives. In particular, when making use of contracted services, the MTs are assigned a number of SCHs (at least one) for signaling and/or a number of LCHs for data transmission, while in the case of a non-contracted services, the MTs are either polled by the AP or have to go through a contention mechanism to place their resource requests. Our main objective is therefore to propose a general framework for provisioning the HIPERLAN/2 standards of a comprehensive set of QoS mechanisms. It is worth to mention that, to the authors knowledge, past work in this area has been limited to the definition of resource request mechanisms to support time-constrained services [9], [10]. However, no attempts have been made to define an overall framework integrating all service classes.

Based in the type of resource request mechanisms used by a connection, we can defined four types of them:

- *Type 1*: In this case, the MT operates under a contracted service policy: a certain number of SCH and LCH channels are assigned per frame or every given number of frames to the MT.
- *Type 2*:Under this second type, the resource request mechanism is initiated by the AP through a *polling* mechanism. The AP polls the MT at the beginning of the connection allowing the MT to request the number of SCHs and/or LCHs that it needs in the following frame. The AP may require more than one frame to allocate the LCH channels requested by the MT depending on the network load. As soon as the AP finishes granting the total number of LCH channels requested, the AP polls the MT once again. In order to avoid excessive delays, the AP initializes a timer as soon it polls the MT. When the timer expires, the AP polls once again the MT. This mechanism attempts to compensate for any extra delay incurred during the resource granting process.
- *Type 3*: Under this connection class, the MT operates under a non-contracted service policy. The MT has then to request its resources by sending a message using an RCH channel. The access to this channel is done using a contention process. Once having finished the allocation of the LCH channels required by the MT, the AP, similarly to the Type 2 mechanism, allocates a SCH to the MT. In other words, as far as the MT remains active, a SCH is assigned to it. Otherwise, the MT will have to go through the contention process to send a RR message after an idle period.
- *Type 4*: The main difference between this type when compared to Type 3 comes by the fact that regardless of the activity of the connection, the MT has to go through a contention process, via a RCH channel, to place its resource request. In particular, different to the previous type, Type 3, once the AP has finished to fulfill the MT requirement, the MT has to go once again through a competition process to place its request, conveyed via an RCH channel.

4.2 Connection Types vs. Applications

Each one of the connections previously described have been defined bearing in mind that HIPERLAN/2 will have to provide support to various types of applications. In this way, Type 1 is an excellent candidate for CBR applications requiring a fixed capacity to fulfill their QoS requirements.

In turn, Type 2 connections are well adapted for VBR applications, such as video streaming and videoconference, among others. The use of the polling mechanism guarantees that the MTs will be able to periodically gain access to the channel to place their requests.

The Type 3 connections have been designed to accommodate a best-effort type of service. The use of a contention-based process responds to the fact that this type of service does not require any guarantees in terms of delays or jitter. However, by allowing the MT to receive a SCH channel as soon as it has finished to exhaust its previous resource booking, the MT is able to place its next resource request shortly. Finally, Type 4 connections offer the lowest access level. This service has been designed to provide support to traffic that does not require any service guarantee, such as background traffic, while limiting the use of resources for signaling purposes at the lowest level, i.e., minimum overhead.

4.3 Bandwidth Allocation Schemes

One of the main roles of the AP is to define the actual allocation of the channels that compose the frame. In general terms, two groups of channels can be distinguished, data (LCH) and control (SCH) channels. As we shall see, the amount of overhead introduced into the frame will heavily depend on the way the channels are assigned to the various connections. In particular, the overhead can be reduced by contiguously placing the channels associated to a given connection. In order to analyze this important issue, we consider the use of the following three bandwidth allocation schemes:

- FIFO (*First-In-First-Out*): each resource request is served following a FIFO discipline. Given that the there is an explicit classification based on the resource request mechanisms, all the requests are stored in a single queue upon their arrival.
- RR (*Round Robin*): under this policy, a queue is assigned to each connection. The queues are served following a round-robin discipline and are allowed to make use of only one (LCH or SCH) channel per visit. A queue will be visited only once again after all the other queues have been visited. It is important to note that a work conserving strategy is used, i.e., whenever a queue being visited is empty, the next non-empty queue can make use of the available bandwidth. The requests for control channels (SCHs) are assigned a higher priority over the requests for data channels (LCHs).
- MORR (*Minimum Overhead Round Robin*): this scheme is similar in operation to the RR scheme. The main difference lies on the fact that whenever a queue is visited, all the requests present in it are served up to available bandwidth. The main aim of this scheme is to limit the amount of overhead to be introduced in the frame by contiguously allocating the channels pertaining to a given connection.

It should be clear that these bandwidth allocation schemes come to supplement the resource request mechanisms. While the role of the resource request mechanisms is the classification of the various applications, the role of the allocation mechanisms is the distribution of the channels among the requesting MTs based on this classification.

5 Performance Evaluation

5.1 Scenario

In our study we use one HIPERLAN/2 cell operating in centralized mode, which has been implemented in OPNET 10.0 [11]. We assume that the connections have already been established, i.e., the only control messages being sent over the channel are those used by the resource requests mechanisms previously described. In the composition of the frame we use short preambles, guard times of $2\,\mu s$, three RCHs channel in the RA phase and the physical mode for the SCH and LCH channels are QPSK3/4 (18 Mbit/s) and 16QAM9/16 (27 Mbit/s), respectively.

Throughout our study, we have considered four main traffic types: video, voice, best-effort and background. The video traffic has been characterized by MPEG-4 [12] video traffic traces. Each video application begins its transmission within a random period given by the expression $t = uniform(0, \frac{12}{f})$ being f the frame rate. In this way, the peak periods of the source rates are randomly distributed along a GOP (Group of Picture) period. The transmission of a video frame is uniformly distributed along the interval of duration of a frame ($\frac{1}{f}$). We use the MPEG-4 sequence *funny* encoded on CIF format at 25 frames/sec.

We assume the use of constant bit-rate voice sources encoded at a rate of 16 Kbit/s according to the G728 standard [13]. Similarly to the video applications, the voice sources are randomly activated within the first 24 ms of the simulation. The best-effort traffic is generated using the traffic model for Web surfing applications described in [14]. The background traffic generated by each source is a combination of ftp, e-mail and Napster according to the model described in [15]. The traffic sources of these two latter traffic types are initiated at the beginning of the simulation run.

In order to limit the delay experienced by the video and voice applications, an essential condition to guarantee the QoS required by both applications, the maximum time that a unit of video and voice referred from now on as packet may remain in the transmission buffer has been set to 100 ms and 10 ms, respectively. These time limits are on-line with the values specified by the standards and in the literature [16]. A packet exceeding this upper bound is dropped.

Each point in our plots is an average over twenty five simulation runs, and the error bars indicate the 90% confidence interval.

Our study has focused on evaluating the performance of the proposed QoS framework, when supporting various types of applications: video, voice, best-effort and background (this classification is on line with the IEEE 802.1p standard specifications), all of them in both directions, i.e., uplink and downlink. In order to carry out this study, we have considered a scenario where a third of the MTs are running voice/video applications. Other third of MTs generate best-effort traffic and finally all other MTs generate background traffic.

Given that one of the main objectives of the study is to evaluate the performance and effectiveness of the proposed resource request mechanisms, we have carried two set of simulations corresponding to two different scenarios. Under the first scenario, namely Scenario 1, all applications have to go through a contention-based process when attempting to transmit each and every resource request packet. Under the second scenario, Scenario 2, each of the applications makes use of a different type of mechanisms. The following has been used: voice services make use of the Type 1 mechanism with an LCH channel reserved every 12 frames (this corresponds to a guaranteed data rate of 16 Kbit/s). Video services make use of the Type 2 mechanism with a timer period of 40 ms; the value of this parameter has been derived based on the results obtained in [17]. The best-effort and background traffic make use of the Type 3 and Type 4 mechanisms, respectively. Table 1 summarizes the parameters used in the second set of simulations.

<table>
<tr><th>Service</th><th>Scenario 1</th><th>Scenario 2</th></tr>
<tr><td>Voice</td><td rowspan="4">contention-
-based</td><td>Type 1, #LCH/#frame = 1/12</td></tr>
<tr><td>Video</td><td>Type 2, Polling Thr. = 0.04s</td></tr>
<tr><td>Best-effort</td><td>Type 3</td></tr>
<tr><td>Background</td><td>Type 4</td></tr>
</table>

Table 1. Parameters for Simulation Scenarios.

The second objective of the work is to evaluate the performance of the bandwidth allocation schemes. In particular, we have been interested in studying the frame occupancy in terms of the overhead introduced to properly identify the channels allocated to each MT. In this second part of our study, we will be making use of all the three bandwidth allocation mechanisms introduced in Section 4.3.

5.1 Metrics

For this part of our study, we have been interested in assessing the performance in terms of the following metrics: total normalized throughput, overhead, mean end-to-end delay, jitter and the cumulative distribution functions for the end-to-end delay and jitter.

The total normalized throughput is calculated as the percentage of the total offered data (the traffic from the all sources) that is actually delivered to the destination. It should be clear that this metric lies within the interval $[0, 1]$. Whenever the reported value for this metric is less than 1, this fact indicates that there are packet losses.

The overhead is a relative measure and it is simply defined as the ratio between the control bits and the total number of bits (data plus control) being sent, i.e., composing the frame. This gives us a clear indication of how the capacity of the channel is being used. It should be clear that at low loads, there may be some spare capacity, i.e., the frame is not completely filled up.

The packet loss rate is evaluated for voice and video communications. Recall that in these two types of communications, packets experiencing long delays while waiting to be transmitted are dropped.

The packet end-to-end delay is evaluated for each type of connection. This metric is simply defined as the time elapsed between the packet generation time at the MT and the packet arrival time at its destination. This time includes both the waiting time in the transmitting buffer and the actual transmission time.

The jitter is an important metric for services requiring time guarantees. In our case, this metric is important to be evaluated in the case of voice and video communications. In the case of voice communications, jitter is defined as the inter-arrival time between two consecutive packets. In the case of video communications, the jitter is evaluated as the inter-arrival time between the last packets pertaining to two consecutive video frames. This last definition is on line with the timing requirements of video communications: the video frames have to be received at a constant pace.

Because delay-sensitive voice and video applications have a end-to-end delay and jitter bound after which the data is useless, it is equally important to study the CDF of these two metrics. The CDF is evaluated at a given network load.

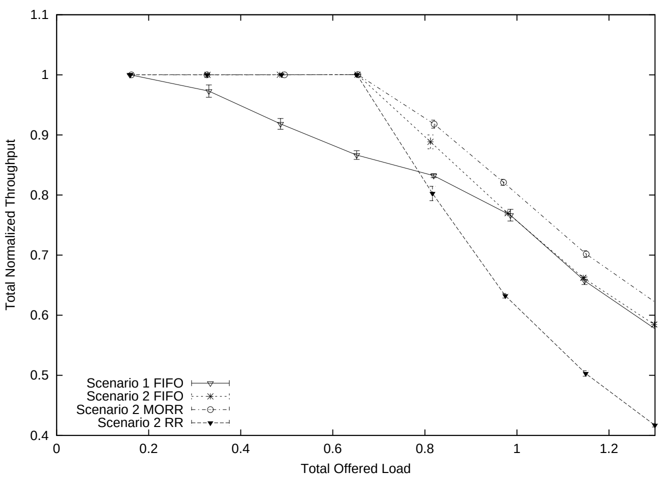


Fig. 1. Traffic Granted.

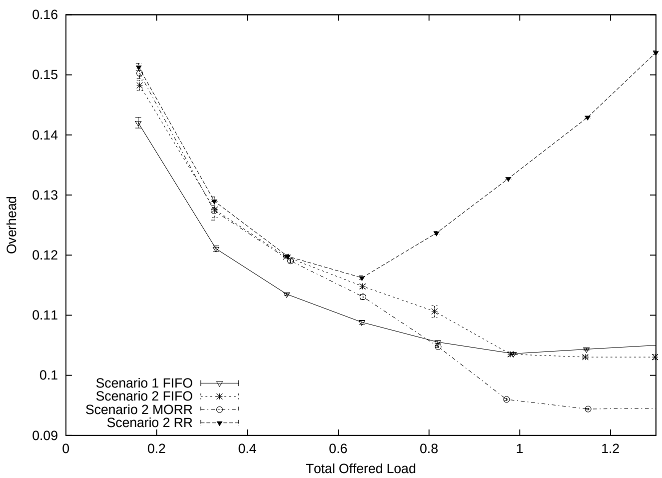
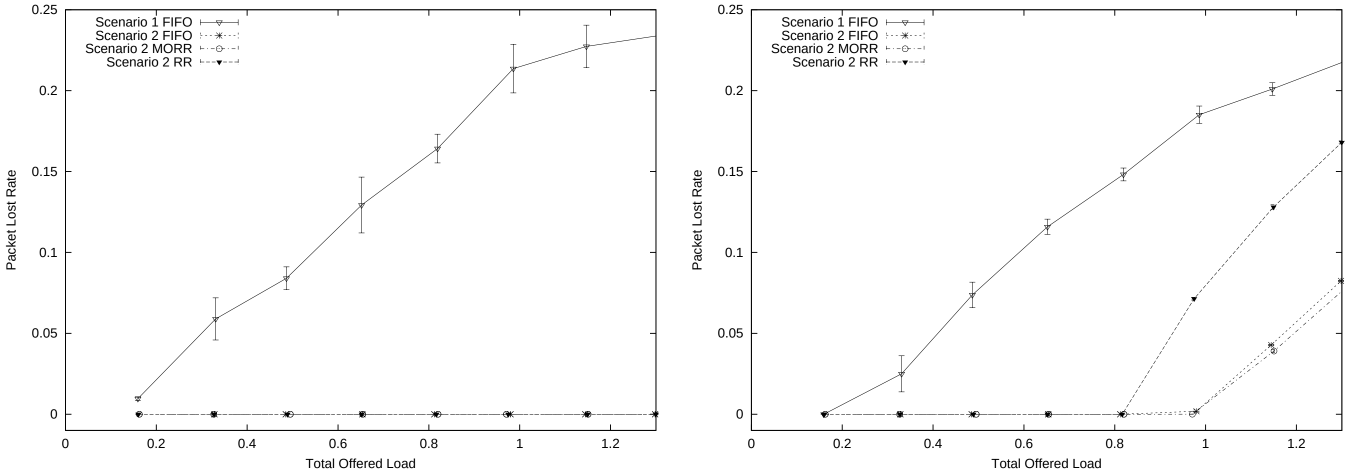


Fig. 2. Overhead.



(a) Voice PLR. (b) Video PLR.

Fig. 3. PLR for Voice and Video Connections.

5.2 Simulation Results

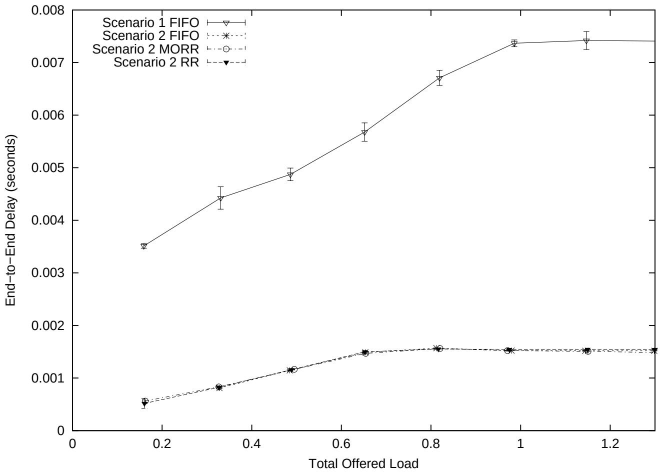
Figure 1 represents the normalized (carried) throughput as a function of the offered load for both scenarios and all three bandwidth allocation mechanisms. As seen from the figure, as the load increases, the performance of Scenario 1 badly degrades. This situation can be simply explained as follows. Since the MTs have to go through a contention mechanism to place their requests, as the load increases the number of collision in the RA phase increase dramatically. Furthermore, the fact that RR bandwidth allocation scheme exhibits the worst results under heavy load conditions is due to the need of dedicating more channel for control purposes. This problem is partially solved by making use of the MORR at the expense of penalizing the multiplexing gain.

Figure 2 depicts the overhead as a function of the offered load for all the three bandwidth allocation mechanisms under study. As seen from the figure, the overhead decreases as the load is increased for all three mechanisms and for loads up to 50%. However, as the load increases, the overhead introduced by the RR starts to steadily increase. This is due to the fact that by allocating the bandwidth to a larger number of MTs, the number of channels dedicated to convey control information increases. Under heavy load conditions, this behavior penalizes the system performance by limiting the available bandwidth for actual data transfer.

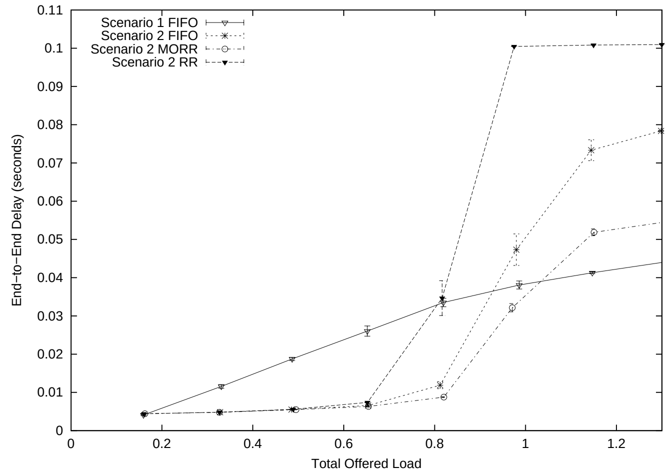
For the case of the FIFO mechanism, the overhead introduced in the frame is lower under the Scenario 1 than in the Scenario 2. This difference is due to the mechanism used to place the requests and the policy used to serve the requests. Recall that under the Scenario 1, the MTs make use of a contention-based process to place their requests. As the load increases, the MTs spend more time attempting to place their requests. As the number of channels requested is being updated during this period of time, a larger number of channels will be requested. Furthermore, since the requests are served following a FIFO policy, the overhead decreases as the number of actual channels used to convey user data is increased.

However, in the case of the Scenario 1 making use of a FIFO discipline, the amount of overhead is initially lower than for the Scenario 2 and a MORR discipline. This is due to the delay encountered to place the requests as previously explained. This trend is observed up as the load increases and converging as the network saturates.

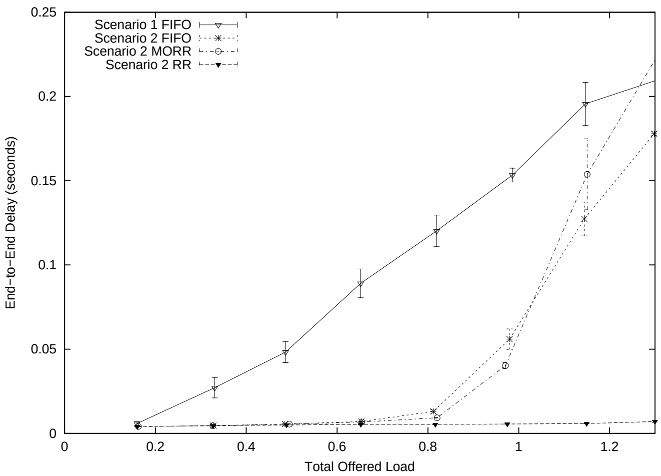
Figure 3 shows the packet loss rates for voice and video connections. These losses correspond to the packets dropped as soon as the exceed the maximum allowable queuing delay. In the case of voice connections, Figure 3.(a) shows that the losses are completely avoided by statically allocating an LCH every 12 frames and independently of the bandwidth allocation scheme being used. In the case of video connections, the bandwidth allocation scheme plays a major role on their performance. Figure 3.(b) shows that for the case when the RR scheme is used, the PLR steadily increases beyond the 70% of the network capacity. Once again, this can be explained by the overhead introduced by this scheme that attempts to multiplex a larger number of connections than the other two bandwidth allocation schemes, namely FIFO and MORR. The use of these two last schemes limits the PLR to less than 1% even when the network operates under very heavy load conditions ($\approxeq 1$).



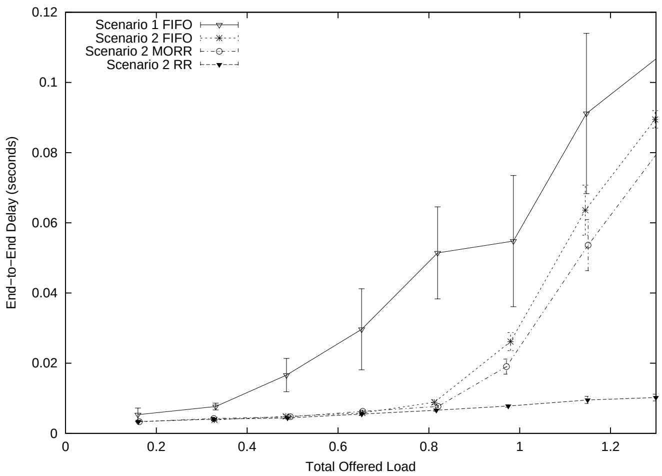
(a) Voice End-to-End Delay.



(b) Video End-to-End Delay.



(c) Best-effort End-to-End Delay.



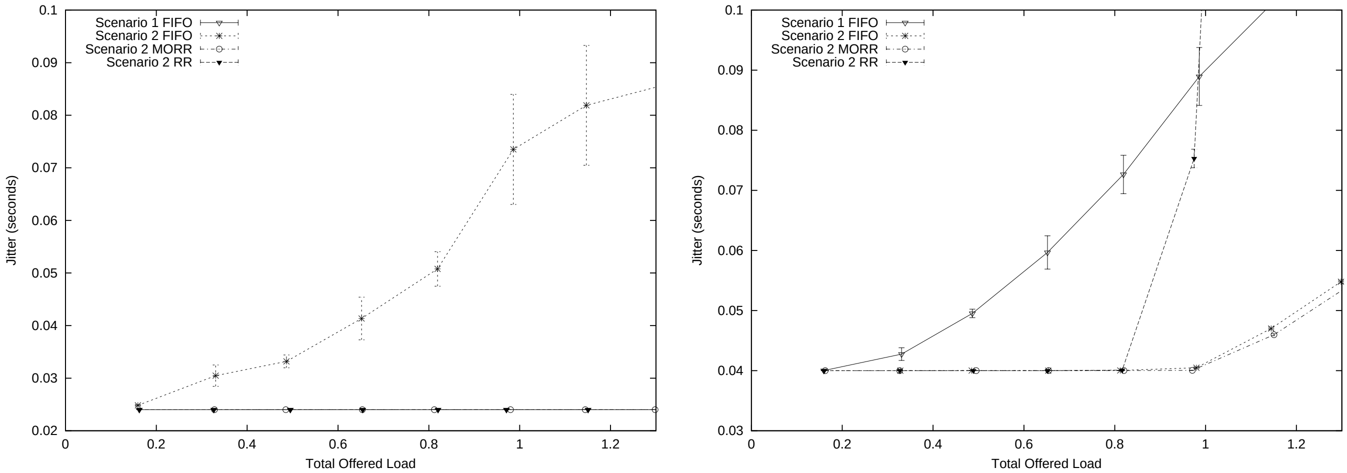
(d) Background End-to-End Delay.

Fig. 4. End-to-End Delay for all Connection Types.

Regarding the packet end-to-end delay shown in Figure 4, it is clear that the packets experienced longer delays under the Scenario 1 than in the Scenario 2. This is once again due to the contention-based mechanism used to place the requests. The only case for which both scenarios exhibit similar results are for the video communications and network loads exceeding 70%. However, this can be explained by realizing that Figure 4(b) only shows the end-to-end delay for the video packets actually having been transmitted.

For the case of voice communications, since under the Scenario 2, we make use of a static allocation of all the required resources, the voice packets are transmitted with no other delay by the time required for their actual transmission within a time frame (2 *ms*). The slight increase as the load increases is due to the fact that as more connections become active, the associated LCHs are spread across the frame.

Figures 4.(c) and (d) show that the RR bandwidth allocation mechanism exhibits the best results for best-effort and background connections. On the contrary, this mechanism shows the worst results for the video traffic (Figure 4.(b)). This is because these connections require a large number of LCH per visit.



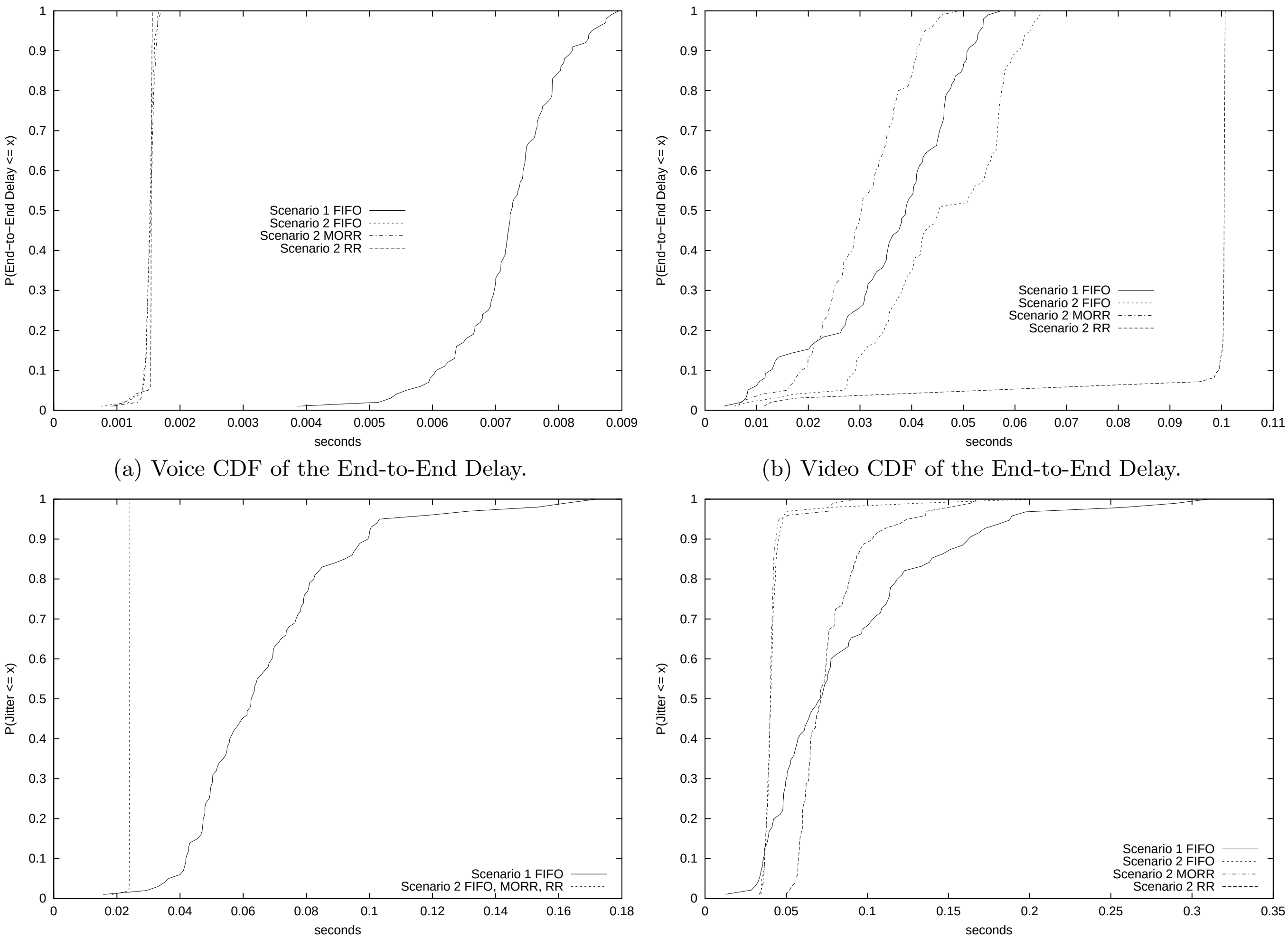
(a) Voice Jitter.

(b) Video Jitter.

Fig. 5. Jitter for Voice and Video Connections.

Regarding the jitter, Figure 5.(a) shows that voice communications do not suffer any deviation since a static allocation of the LCHs ensure the isochronous transmission of the voice packets, one voice packet every 24 *ms*, independently of the network load conditions. In the case of the video traffic (Figure 5.(b)), the jitter remains constant for the Scenario 2 when using the FIFO and MORR mechanisms and load conditions up to 1.

Even though, the mean end-to-end delay is an important metric when assessing the performance of a network, the CDF is a key metric in a multiservice network supporting time-constrained applications. Figure 6 shows the CDF of the end-to-end delay and the jitter for a system operating at full load ($\approxeq$ 98%). Figures 6.(a) and (c) shows that the voice communications are unaffected since the networks guarantees them the required capacity (Scenario 2). In the case of the video traffic, our results show that the MORR mechanism guarantees an end-to-end delay of less than 50 *ms* to all packets. In the case of the RR mechanism, the video packets can experience up to the maximum allowable end-to-end delay, i.e., 100 *ms*. For the jitter, Figure 6.(d) shows that 95% of the inter-arrival times between video frames are 40 *ms* when MORR or FIFO are used in the Scenario 2. This corresponds to the sampling rate of 25 frames/*s*, i.e., a frame every 40 *ms*. In other words, 95% of the video frames arrive to their destination on an isochronous manner. This is an excellent result that indicates clearly the effectiveness of the proposed mechanism.



(a) Voice CDF of the End-to-End Delay. (b) Video CDF of the End-to-End Delay.

(c) Voice CDF of Jitter. (d) Video CDF of Jitter.

Fig. 6. CDF for the End-to-End Delay & the Jitter for Voice & Video Connections (Offered Load $\approxeq 0.98$).

6 Conclusions

In this work, we have proposed a complete set of control mechanisms and evaluated their performance in terms of various metrics of interest. Our main aim has been to construct a structured set mechanisms aiming to provide the QoS guarantees required by time constrained applications when coexisting with other services (applications) in a TDMA/TDD wireless network.

We have presented a QoS framework to support a multi-service environment in a wireless network. The proposed framework has been used to design various QoS provisioning mechanisms. As part of this QoS framework, we have defined a service taxonomy. Under this definition, we have defined a two level hierarchy making use of the mechanism proposed by the HIPERLAN/2 standards. We have evaluated the proposed framework and conducted a comparative performance evaluation study, concluding that the various types of resource request mechanisms are very effective in meeting the applications needs of time constrained applications, voice and video, as well as on improving the service provided to the other two types of traffic: best-effort and background traffics.

We have come to the conclusion that the use of resource request mechanisms adapted to the requirements of various types proves effective as an initial step towards the provisioning of QoS guarantees. These mechanisms can be supplemented by resource (bandwidth) allocation mechanism to effectively guarantee the requirements of each one of the applications.

We have also evaluated various bandwidth allocation mechanisms. In particular, we have been interested in looking into scheme allowing us to better use the available bandwidth. We have shown that there is possible to make use of simple scheme to reduce the amount of overhead to be introduced in to the frame.

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