TDuCSMA: Efficient Support for Triple-Play Services in Wireless Home Networks

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***Abstract*—The recently proposed Time-Division Unbalanced Carrier Sense Multiple Access (TDuCSMA) coordination function has been shown to cope very efficiently with many different data flows with widely different bitrates and packet lengths. This case may happen in a typical wireless home network setting where HD video, voice, videosurveillance and videoconference applications are used concurrently. This paper illustrates the advantages of TDuCSMA in such a scenario compared to the Enhanced Distributed Channel Access (EDCA), currently provided by the IEEE 802.11 standard, in terms of both performance from the end user's point of view and network resource utilization. The results show that TDuCSMA provides much better performance than EDCA, especially regarding video quality, delay and packet loss rate for voice streams, and stability over time. Moreover, the TDuCSMA performance is consistent as the network load increases while the EDCA performance significantly decreases.**

I. Introduction

The IEEE 802.11 standardizes the most widespread Medium Access Control (MAC) used to provide wireless capabilities to traditional computer networks. Given the tendency to move all digital services to wireless, it is reasonable to expect that the IEEE 802.11 technology will be the preferred choice to interconnect all digital devices in the home environment. In such wireless home networks, several types of traffic, with widely different requirements, will coexist. Traffic types range from the ones generated by traditional Internet applications, *e.g.*, web, e-mail, and file sharing, to the most demanding ones, *e.g.*, streaming and real-time multimedia applications. The network must be flexible enough to provide each application with the appropriate quality-of-service (QoS) to enhance the end user quality-of-experience (QoE), eventually enabling the so called *triple-play* service.

A lot of effort is currently carried out to increase the physical layer communication speed, such as the IEEE 802.11ac proposal, with the aim to use overprovisioning as a mean to provide a better service. However, the bandwidth demand of applications, especially video, will only increase in the near future due to the usage of high-definition and beyond resolutions. In this context, an interesting approach is the one proposed by the IEEE 802.11 High Efficiency WLAN (HEW) Study Group, started in July 2013, that focuses directly on improving the efficiency and performance of WLANs. Thus, the demand of solutions to provide QoS in the WLAN context will likely increase in the near future, and it is complementary to any other solution that will be devised to increase the speed at the physical layer.

Current IEEE 802.11 solutions, in fact, are not particularly efficient, since they rely on either the Distributed Coordination Function (DCF) or the Enhanced Distributed Channel Access (EDCA) [1]. However, both of them can only partially cope with the triple-play challenge, due to their fundamental limitation in ensuring the necessary QoS to all applications at the same time, especially when applications generate heterogeneous traffic which is widely different in bitrate and packet length. Moreover, network usage inefficiencies of a single high-priority application may negatively affect the performance of the whole network. Some work focused on optimizing the support of EDCA for multimedia applications, such as in [2] that studies the issue of how to tune the EDCA to provide good service differentiation in specific voice traffic scenarios. The particular case of IPTV streams has been addressed in [3], investigating the capacity, delay, jitter and video quality performance as a function of several physical data-rates.

A possible solution to the EDCA shortcomings, that also moves in the direction of providing HEW services, is represented by a novel coordination function called Time-Division Unbalanced Carrier Sense Multiple access (TDuCSMA) [4]. It relies on synchronized switching of contention parameters to provide a viable solution for bandwidth management and QoS provisioning while still retaining compatibility with the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) scheme which is the core of the original IEEE 802.11 standard. The TDuCSMA is flexible enough to provide the upper layers with a suitable interface to implement dynamic and distributed bandwidth and traffic management. This can be achieved by a signaling architecture as described in [5]. The TDuCSMA operating principles have been extensively investigated in single [4] and multi-hop [6] scenarios by simulations and theoretical models. Moreover, the work in [7] proved the full compliance with the IEEE 802.11 standard, including the possibility of coexistence of TDuCSMA nodes with legacy ones. Note also that the scheme can be implemented with very few adaptations of the software drivers of the wireless cards, and it does not require modifications in the firmware loaded into the hardware [8].

While the previous works assessed the TDuCSMA as a coordination function for broadband access strictly from the network point of view, to the best of our knowledge no works

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thoroughly investigated its performance in term of QoE in a wireless home networking scenario, especially when flows of different types (audio, video and data) coexist at the same time, as it is expected when a triple-play service is requested. The authors partially investigate a similar issue in [9] but limited to video services and only in a low physical data-rate wireless network, and in [10] where some experiments with a real implementation are presented in a limited-size setting.

The main contribution of this work is to investigate the performance of provisioning triple-play services in a wireless home network scenario from the application layer perspective. Comparisons are presented with alternative solutions, such as DCF and EDCA, showing the advantages of TDuCSMA in terms of both absolute performance, efficiency and consistency over time as a function of various network loads.

The paper is organized as follows. Section II recalls the TDuCSMA operating principles. How to optimally configure it to effectively support voice, video and data services concurrently is addressed by Section III. After the simulation setup described in Section IV, quantitative results are presented in Section V in terms of network and application level performance metrics. Finally conclusions are drawn in Section VI.

II. Time-Division Unbalanced Carrier Sense Multiple Access

A. Operating Principles

TDuCSMA nodes are standard EDCA nodes synchronized with a common time reference (CTR) whose structure is shown in the lower part of Fig. 1. The CTR is a periodical time structure; the time-frame (TF) is the entity representing the time unit and a given number of TFs are grouped in a time-cycle (TC). The time-cycle length T_c, measured in TFs, provides the periodicity of the CTR structure. Both the time-frame duration T_f and T_c are configurable system parameters. The synchronization among nodes can be achieved using, for instance, a distributed leaderless solution [11], where nodes collaboratively reach a consensus on a common clock. Note that, although a typical TDMA structure is employed, the channel access decision process is distributed among all nodes according to the traditional CSMA/CA rules.

Two sets of EDCA parameters $(AIFS, CW_{min}, CW_{max})$ are predefined and used by all nodes accessing the network. These sets are referred to as *high-priority* set EDCA^H and *low-priority* set EDCA^L . They have the following characteristics:

$$AIFS^H < AIFS^L,$$

$$CW^H_{min} \leq CW^H_{max} < CW^L_{min} \leq CW^L_{max}$$

so that a node contending for channel access in accordance with EDCA^H has strict priority on any other node using EDCA^L settings.

The underlying idea of TDuCSMA is to synchronize the switching of EDCA parameter sets at each node so that: (1) only one node at a time contends for channel access using the EDCA^H parameter set; (2) each node i uses the EDCA^H parameter set for a predefined periodically repeated time interval, referred to as T^i_H, otherwise EDCA^L is used.

Fig. 1 shows the time-driven switching of EDCA parameters inside three nodes sharing the same collision domain. As depicted, only one node contends for channel access using the EDCA^H parameter set during one TF, whereas the time period T^i_H during which each node i operates using EDCA^H is different for each node.

The channel access relies only on the CSMA/CA scheme coupled with suitable values for the EDCA parameter sets, and it is not achieved by means of predefined channel access rights as in traditional TDMA solutions. This is the key idea which allows both backward compatibility and to retain the advantages of TDMA-based solutions. The transmission opportunity *TXOP* mechanism is not exploited in TDuCSMA because if a node were delayed in its channel access, *TXOP* would enforce this delay and propagate it with a disruptive effect on the underlying TDuCSMA operating principles.

In principle the EDCA parameter sets are switched over time on a per-node basis, so that each node handles QoS-demanding traffic as a single aggregate. Thus, bandwidth management is performed on a per-node basis (not per-flow) by assigning different T_H to nodes sharing the same collision domain. However, TFs inside the TC can also be left un-allocated to send background traffic. Fig. 1 shows that the time cycle can be divided into two periods, “Allocated” in which TFs are allocated to nodes and “Unallocated” during which TFs are left un-allocated.

In a possible design, two backoff entities are implemented within a node to support QoS and best-effort services in an integrated fashion [7]. The TDuCSMA backoff entity contends for channel access to send QoS-demanding traffic using EDCA^H . The CSMA/CA backoff entity contends for channel access to send background traffic using *EDCA*^{L} settings. Moreover, since TDuCSMA preserves backward compatibility, legacy IEEE 802.11 nodes can still join the network [7]. Notably, the service provided to background traffic can be further differentiated at each node. For instance, in [7] the background traffic is delivered by parallel backoff entities which are prioritized using AC specific contention parameters as in the current EDCA implementation [12].

Moreover, since TDuCSMA still relies on CSMA/CA, if a node i does not have enough traffic to send before the end of its T^i_H, any other node can gain access to the channel, following the CSMA/CA rules, and transmit. Hence, bandwidth reuse is easily and intrinsically implemented without additional complexity.

The work in [4] showed two important consequences of the TDuCSMA operating principles: (1) if a node i accesses the channel during its allocation time T^i_H, only the node itself gains access, thus the congestion window parameters in EDCA^H can be minimized to reduce back-off time between two consecutive transmissions; an important consequence is that bandwidth utilization is increased without affecting collision probability; (2) if node i uses its T^i_H with poor efficiency due to, *e.g.*, small packets, this does not affect the transmission

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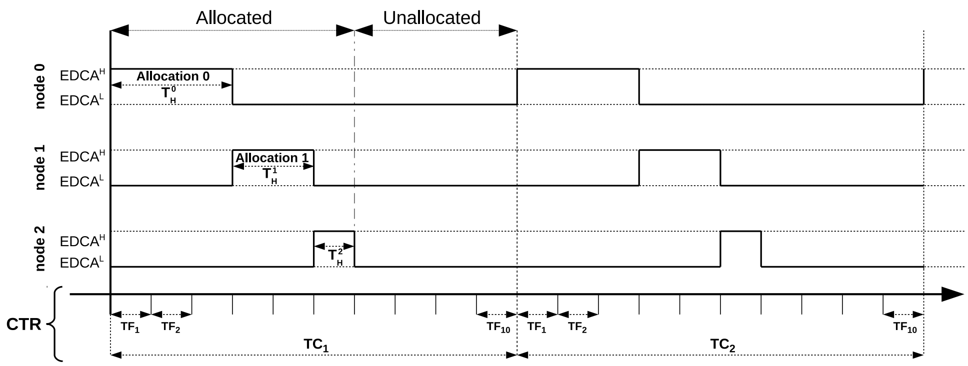

Fig. 1. Time-driven switching of EDCA parameters inside three nodes; $T_H^0 = 3$, $T_H^1 = 2$, $T_H^2 = 1$ over a time cycle of 10 TFs.

efficiency of the other nodes in their respective allocation periods.

B. Bandwidth Reservation Management

From the application point of view, the TDuCSMA provides an API to the upper layer to request and manage resource reservations. To implement this API, the TDuCSMA has been complemented by two control protocols to manage the signaling needs between the nodes as detailed in [5]. A Resource State Management Protocol (RSMP) has been designed to implement distributed bandwidth management with TDuCSMA, *i.e.*, to automatically distribute the allocation states of TFs and keep information consistent among nodes. The standard Resource Reservation Protocol with Traffic Engineering (RSVP-TE) has been properly extended to support the TDuCSMA signaling needs [5]. Both protocols follow the soft-state reservation approach and the overhead introduced, in terms of additional network traffic, is negligible compared to the bandwidth available.

III. Supporting Triple-Play Scenarios with TDuCSMA

In the very near future, it is expected that all digital services, in home scenarios, will take advantage of wireless communications. This poses the challenge to provide each application with the appropriate QoS at network level so that users experience an acceptable QoE.

From the multimedia communication point of view, the challenge basically translates into provisioning the applications with upper bounded delays and a bandwidth amount sufficient to transmit almost all the packets with no losses. Differently from generic data transfers, multimedia applications can, in fact, tolerate a few losses provided that their number is limited and appropriate concealment techniques are used at the receiver.

Note, however, that each multimedia application has peculiar characteristics that change the importance and the limits imposed on each QoS parameter, *e.g.*, delay and bandwidth. For instance, videoconference applications have strict constraints on the maximum tolerable delay, since this is an important factor to ensure interactivity between the users. For video streaming applications the delay constraint is more relaxed, even though it cannot be too high, especially in case of live streaming which simulates a traditional broadcast TV service; conversely, in streaming services the amount of lost packets should be very limited, especially for high quality video services, *e.g.*, HD video, where the user is less inclined to tolerate distortions.

Audio applications, stand alone or coupled with video communications, add complexity to the scenario. In wireless networks it is particularly difficult to cope with a high number of audio communications since they are generally characterized by a high number of very small packets which tend to decrease the network performance.

Moreover, while all the previous transmissions are active, the user often wants to be able to access Internet contents ubiquitously from different types of devices *e.g.*, laptops and tablets, inside the home area. Therefore, while other multimedia communications are taking place, the networking system should still be able to provide some bandwidth for elastic applications such as file transfers without starving them.

A system based on TDuCSMA, with its bandwidth management capabilities, can fulfill the different application requirements, while exploiting all the available wireless bandwidth. The available bandwidth should be reserved for audio and video applications while the remaining bandwidth can be left un-reserved for data applications, relying on their elastic nature. The bandwidth should be reserved considering all QoS-demanding flows together in each node. On the basis of this bandwidth requirement, each node computes the number of TFs which needs for the QoS-demanding traffic [7].

For the best results, it is desirable that the amount of bandwidth to reserve is known. This is usually not a problem for audio transmissions, since most voice and audio codecs know in advance the amount of bits that will be produced for a certain set of the encoding parameters. Video codecs, instead, can work ensuring almost constant quality while producing variable bitrate traffic or adjusting parameters, by means of rate control algorithms, to ensure that bitrate remains approximately constant while, of course, quality fluctuates depending on video content. The second approach is commonly used since communication channels and storage devices often poses strict constraints on the maximum amount of bits that can be transmitted or stored in a given time span. The following sections analyze how to suitably configure TDuCSMA to

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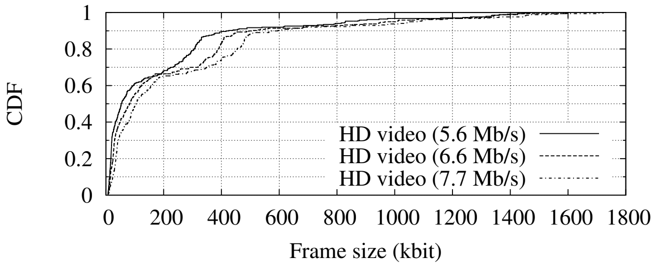

Fig. 2. CDF of the frame sizes for the HD video.

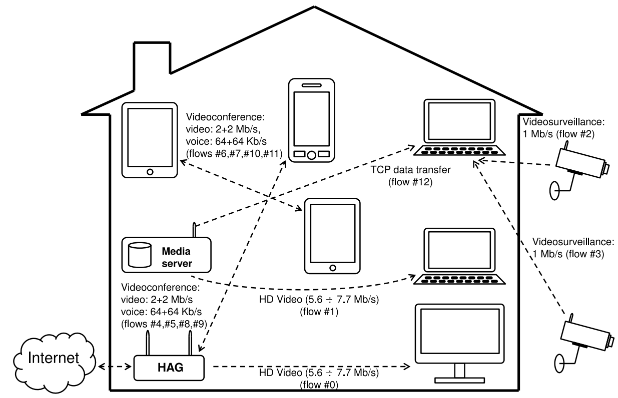

Fig. 3. Wireless home network scenario under simulation.

handle different types of traffic.

A. Voice

The bandwidth reservation process for audio application is extremely simple since an audio flow is characterized by strictly constant bitrate and packet length. A share of the available bandwidth equal to the bitrate is reserved by allocating a number of TFs. Moreover, the reservation is periodical with period T_c, hence the TFs can be chosen in order to match the voice flow period, which is generally standardized, with obvious benefits in terms of end-to-end delay.

B. Audio Video Services

Multimedia flows carrying synchronized audio/video streams can also be efficiently handled. The audio flow is treated as a voice flow while for the case of video traffic the estimation of bandwidth requirements is performed as follows. The bandwidth required by the video flow is considered as approximately equal to its average bitrate, although also for the rate-controlled case different types of frames (intra or inter-coded ones) have different sizes, which is the reason why the instantaneous bitrate value may be high. Fig. 2 show the cumulative distribution functions (CDF) of the sizes of the compressed video frames employed in the HD sequences used for the experiments described in Sec. IV. The curve sharply increases until about the 90% value, meaning that most of the video (90%) can be transmitted as high priority if the reservation corresponds to the bitrate of the 90% value. In this condition a satisfactory compromise between efficient resource utilization and video communication performance can be easily found [13].

This allows not to waste network resources while still achieving a good tradeoff between efficient resource utilization and video communication performance [6]. Moreover, TDuCSMA has been shown to be adaptive since it intrinsically allows bandwidth reuse. Hence the remaining part of the video can exploit the bandwidth unused by the other nodes, with obvious benefits in terms of QoE. This is a great advantage with respect to traditional TDMA-based solutions where reuse must be implemented with specific functions and indeed represents a cost in terms of complexity.

C. Data

Data traffic without particular QoS requirements can be handled using unallocated TFs. Therefore, management of QoS-demanding traffic (transmitted with the EDCA^H parameter set) and the T_c value implicitly defines how much bandwidth is left to the generic data traffic. Since all TDuCSMA nodes transmit using the same parameter set all data traffic gets equal chance to be transmitted. This also prevents starvation since high-priority traffic is either handled in their reserved time period or transmitted as normal traffic in the unallocated period.

In this phase, legacy IEEE 802.11 nodes also have the possibility to transmit in the traditional fashion [7]. Moreover, in the unallocated period, services could be further differentiated at each node using AC specific contention parameters as done in the traditional 802.11e architecture.

IV. Simulation Setup

Simulation were run in *ns-2* [14] to assess the performance of TDuCSMA in the home networking scenario depicted in Fig. 3. It comprises a Home Access Gateway (HAG), two video surveillance cameras, two laptops, two mobile computing devices such as tablets, a smartphone, a media server used to store multimedia contents and a digital TV set. The devices communicate wirelessly in accordance with the mesh paradigm, that is, they send data directly to the destinations without using an access point.

The devices operate in accordance with the IEEE 802.11a standard at the physical layer. The exact parameters are as follows: $SIFS = 16\mu s$, $slotTime = 9\mu s$, *PLCP preamble* is 96 bits, *header* is 24 bit. Moreover, the MAC header length is 34 bytes and the ACK length is 14 bytes. In all the simulations the physical data-rate R is set to 54 Mb/s, whereas the basic physical data-rate is set to 6 Mb/s. The auto-fallback mechanism has been disabled.

Concerning the TDuCSMA devices, the TDuCSMA back-off entity works in accordance with

$$AIFS^i = SIFS + AIFSN^i \cdot slotTime \ \forall i = H, L$$

where $AIFSN^H = 2$ and $AIFSN^L = 7$, $CW^H_{min} = CW^H_{max} = 1$ whereas $CW^L_{min} = 31$ and $CW^L_{max} = 1023$. The CSMA/CA back-off entity, within each TDuCSMA node, handles background traffic in accordance with EDCA^L.

Moreover, T_c=33 TFs and T_f=1 ms. In TDuCSMA simulations, for each device a bandwidth equal to the mean bitrate of each multimedia (voice and video) flow is reserved.

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TABLE I

MAC ACCESS PARAMETERS.

Coord. function	QoS	$AIFSN$	CW_{min}	CW_{max}
TDuCSMA	H	2	1	1
	L	7	31	1023
EDCA	AC_{VO}	2	3	7
	AC_{VI}	2	7	15
	AC_{BE}	3	15	1023
	AC_{BK}	7	15	1023
DCF	-	7	31	1023

All the simulations were also performed with legacy devices based either on EDCA or traditional DCF. When EDCA is deployed the contention parameters in the four ACs (AC_{VO}, AC_{VI}, AC_{BE} and AC_{BK}) are configured as in [12]. In DCF devices there is only one back-off entity that works in accordance with the default contention parameters. All the parameters are summarized in Table I for the three coordination functions.

In TDuCSMA simulations video and voice traffic is treated as QoS-demanding traffic, hence handled by TDuCSMA back-off entity, whereas data traffic is handled by the CSMA/CA back-off entity within the same TDuCSMA nodes. In EDCA simulations voice traffic is associated to AC_{VO}, video traffic to AC_{VI} and data to AC_{BE}. Finally when DCF is deployed all types of traffic are served equally.

The MAC/IP/UDP/RTP protocol stack is employed in all the simulations. The characteristics of the traffic flows are summarized in Table II. The bitrates of the multimedia flows are chosen in order to be suitable for the application envisioned in the home networking scenario. Videos are encoded using the H.264/AVC video coding standard. Standard video sequences at 30 frames per second (fps) from [15] are employed: video resolution varies between VGA (640×480) and 720p (1280×720 pixels) for HD video flows. Depending on the application, different GOP sizes and frame types have been used. For videoconference and videosurveillance applications, one frame every twelve has been encoded as an I-type frame while the other frames are coded as P-type. For the HD video application, the IBBPBBPBBPBB frame type scheme has been adopted. In case a slice is lost, the decoder applies a simple temporal concealment technique, *i.e.*, it replaces the missing data with the pixels in the same position in the previous frame. For the videoconference applications, the concurrent transmission of coded voice stream is simulated, splitting each stream into 20 ms voice frames which yields 50 packets per second. Moreover, for all multimedia communications, a play-out buffer is simulated to discard packets that arrive too late for playback.

The data traffic is simulated as a long lasting TCP flow in accordance with the *Reno* implementation within *ns-2*. All the simulations last for 302.4 seconds. This duration has been chosen so that it is a multiple of all sequence lengths. The sequences have different lengths and are repeated to make them longer.

V. SIMULATION RESULTS

The simulation results show the performance from the point of view of the multimedia applications. The video quality is

TABLE II

CHARACTERISTICS OF THE DATA FLOWS IN SIMULATION.

Flow #	Type	Video sequence	Avg. bitrate [Mb/s]	Mean packet size [bytes]
0	HD video	*flatirons*	5.6, 6.6, 7.7*	1400
1	HD video	*flatirons*	5.6, 6.6, 7.7*	1400
2	Videosurveillance	*coastguard*	1.0	1240
3	Videosurveillance	*coastguard*	1.0	1240
4	Videoconference	*spectrum*	2.0	1350
5	Videoconference	*spectrum*	2.0	1350
6	Videoconference	*spectrum*	2.0	1350
7	Videoconference	*spectrum*	2.0	1350
8	Voice	-	0.064	160
9	Voice	-	0.064	160
10	Voice	-	0.064	160
11	Voice	-	0.064	160
12	Data transfer (TCP)	-	-	1460

*This video sequence has been encoded at three different bitrates to simulate different network conditions.

evaluated by means of the Peak Signal-to-Noise Ratio (PSNR) which, despite its limitations, is a widely used measure in the multimedia research community. The Packet Loss Rate (PLR) value is reported for voice applications. As highlighted in the results, the PLR value for the three coordination functions is very different and this result is consistent also if simulation parameters are varied; hence, PLR is itself sufficient to get a glimpse on the application level quality. Finally, the goodput at MAC layer is used as the metric to evaluate the user experience with data applications. This is reasonable since the majority of data applications are based on the TCP protocol, whose main performance metric is indeed the goodput.

Fig. 4 shows the PSNR performance of the HD video applications as a function of the play-out buffer. The performance is shown as an aggregate value, that is, for each application of the same type (HD video, videoconference, videosurveillance) the points shown on the graphs are the average of the PSNR performance achieved by each one of them with one of the three coordination functions. Fig. 4 clearly shows that the performance of HD video applications with TDuCSMA is much better than with EDCA and DCF. A saturation effect is present when the play-out buffer is increased over a certain threshold. In this condition, the only factor influencing the performance is the PLR due to network losses. The performance with TDuCSMA saturates at about 500 ms play-out buffer size, with gains with respect

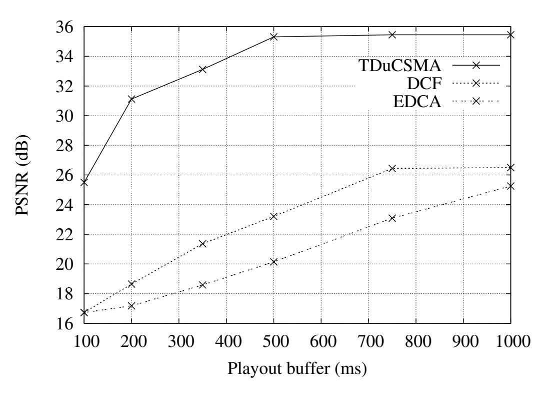

Fig. 4. PSNR as a function of the play-out buffer for the HD video application (aggregated).

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TABLE III
AGGREGATED PSNR PERFORMANCE. PLAYOUT BUFFER IS 200 MS.

Service	TDuCSMA	DCF	EDCA
Videoconference	39.49 dB	34.51 dB	30.95 dB
Videosurveillance	30.45 dB	33.46 dB	26.64 dB

to the other coordination functions in excess of 12 dB. Also considering a more relaxed application requirement, *i.e.*, play-out buffer equal to 1000 ms, the TDuCSMA still provides a gain of about 9 dB PSNR, with absolute value higher than 35 dB PSNR, while the other coordination functions provide unacceptable video quality. This is a clear indication that the TDuCSMA is able to bound the maximum delay of the communication to an amount which is compatible with the application requirements while this characteristic cannot be achieved with the other coordination functions.

Table III shows the performance of the videoconference and videosurveillance applications when the play-out buffer is 200 ms. The PSNR gains with TDuCSMA is about 8 dB PSNR with respect to EDCA and 4 dB with respect to DCF. For the case of the videosurveillance application the PSNR performance with DCF is 3 dB higher than with TDuCSMA. However, despite this fact, the PSNR performance is reasonable, over 30 dB, for the TDuCSMA case. This result is expected and it is a direct consequence of the CSMA/CA operating principles. In fact, DCF provides a fair share of the available bandwidth to each node in the collision domain. Thus a node willing to transmit a low bitrate flow, as the videosurveillance video flow (1 Mb/s), experiences a high QoS service, while the nodes transmitting high bitrate traffic flows are penalized by the fair share behavior of DCF. In the simulated scenario $N = 7$ devices compete for channel access to transmit traffic and the available bandwidth B, that can be estimated according to the analysis in [16], is about 28 Mb/s. Thus the devices transmitting videosurveillance flows (the two cameras) with bitrate 1 Mb/s $\ll B/N \approx 4$ Mb/s receive a very good service from DCF. However, when the whole scenario is considered, the performance decrease suffered by the videosurveillance application with TDuCSMA is counterbalanced by the much higher amount of time sensitive video traffic that can be reliably transmitted in the whole wireless network.

Fig. 5 shows the PLR of the voice part of the videoconference applications as a function of play-out buffer. The lowest PLR is provided by the TDuCSMA. It is about 1% when the play-out buffer is set to 200 ms, that is, the same application requirement considered for the corresponding video. Studies on compressed voice communications over networks indicate that the effect of 1% PLR is almost negligible regardless of the employed codec [17]. The PLR is, instead, too high to support quality audio-video-conference applications in the case of the EDCA (4%, barely intelligible) or, even worse, the DCF (25%, useless).

These PLR results also give useful indications to explain why the PSNR performance of video applications is better with DCF than with EDCA. First, the EDCA gives a priority to voice traffic in AC_{VO} which is strictly higher than any other type of traffic. Moreover, voice applications create many

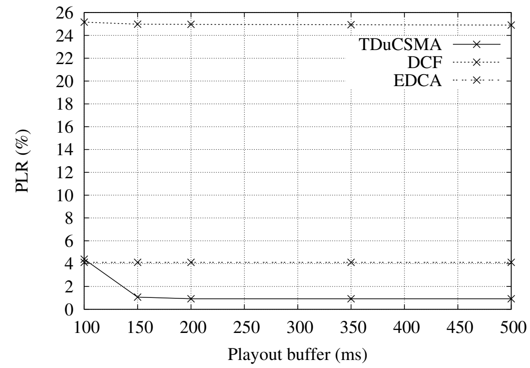

Fig. 5. PLR as a function of the play-out buffer for the voice part of the videoconference application (aggregated).

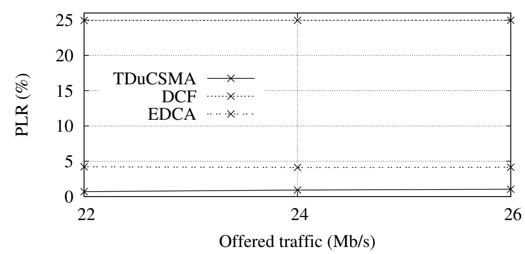

Fig. 6. PLR as a function of the offered traffic for the voice part of the videoconference application (aggregated). Play-out buffer is set to 200 ms.

and very small packets, hence devices transmitting voice flows utilize the available bandwidth with poor efficiency. As a result, the inefficiency affects the network service provided to traffic in lower priority classes (AC_{VI} and AC_{BE}) originating from all the devices and not only from the devices transmitting voice. Therefore, with EDCA, all video applications suffer from this condition. For the DCF case, all the nodes have the same chance to access the channel, but nodes which are requested to transmit voice traffic are more inefficient in using the channel due to the mentioned characteristics of voice traffic. The remaining nodes not involved in voice transmission experience the same efficiency in channel access which, on average, is higher than the one provided by EDCA, where voice traffic, being high priority, reduces the performance of all nodes participating in the network.

Finally, note that with TDuCSMA, instead, each node is isolated from the potential inefficiencies of other nodes. This is another important benefit of TDuCSMA which comes in addition to bandwidth management.

Fig. 7 and 8 shows the PSNR averaged on all the repetitions of the sequence, for the HD video application corresponding to Flow #1 when its bitrate is 5.6 or 6.6 Mb/s. In these conditions the total offered traffic is 22 or 24 Mb/s, respectively. In the simulations, the base sequence (*flatirons*) is repeated 36 times. Each point in the graph consider a specific frame in the base sequence and shows the average of 36 PSNR values, *i.e.*, the PSNR values for that frame in each repetition of the sequence. From the figure, it is clear that the performance with TDuCSMA is, on average, almost always higher than the one

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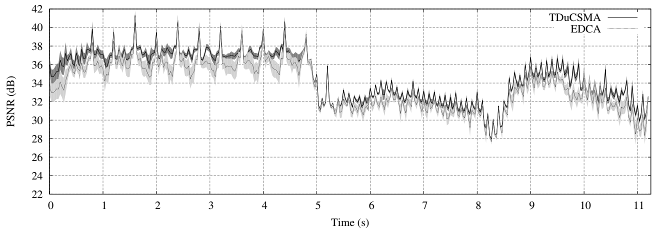

Fig. 7. PSNR as a function of time for the case of 22 Mb/s offered traffic. Flow #1 at 5.6 Mb/s. Play-out buffer is set to 1000 ms. The shaded areas represent the 95% confidence interval.

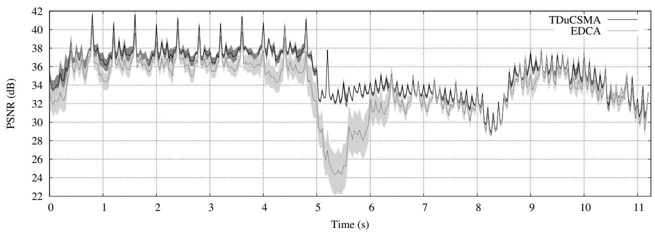

Fig. 8. PSNR as a function of time for the case of 24 Mb/s offered traffic. Flow #1 at 6.6 Mb/s. Play-out buffer is set to 1000 ms. The shaded areas represent the 95% confidence interval.

TABLE IV
GOODPUT AT MAC LAYER [MB/S] FOR THE TCP DATA APPLICATION.

Multimedia offered load [Mb/s]	Goodput [Mb/s] TDuCSMA	EDCA	DCF
22	3.15	1.60	2.19
24	2.10	1.26	1.11
26	0.97	1.23	0.14

achieved using the EDCA. The case of 7.7 Mb/s, *i.e.*, offered traffic equal to about 26 Mb/s, is not shown since the PSNR performance for the EDCA case strongly drops after about 2 seconds due to the excessive delay that packets experience in the transmission queue. Such a delay makes packets useless at the decoder which discards them because they are too late.

Moreover, the shaded areas in the figures show the size of the 95% confidence interval. The comparison between the confidence intervals of the TDuCSMA and the EDCA case shows that, for nearly all the frames in the sequence, the PSNR performance with TDuCSMA is much more stable. Therefore, it can be concluded that not only the average video quality performance with TDuCSMA is higher, but it is also less subject to variations compared to the EDCA case.

Finally, the goodput, at MAC layer, achieved by TCP data application is summarized in Table IV. The performance in the case of EDCA slowly decreases as a function of the offered load whereas they drastically decreases with DCF. The TDuCSMA outperforms the other coordination functions in all conditions, providing a very good service also to the data application, despite the amount of bandwidth decreases as the multimedia traffic share increases.

VI. CONCLUSION AND FUTURE WORK

This work investigated the efficiency of the IEEE 802.11-compatible TDuCSMA coordination function in providing efficient triple-play services in a wireless home networking scenario. It is shown that TDuCSMA can efficiently support scenarios where traffic flows with widely different characteristics coexist, *e.g.*, HD video with voice, videosurveillance and videoconference flows. The quality achieved through the TDuCSMA coordination function is shown to be superior in terms of overall application level quality experienced by the user. The performance is generally higher, in terms of better video quality (up to 9 dB PSNR higher for HD video), lower delay and packet loss rate, especially for the voice case, and lower variability. Moreover, as the network load increases, the TDuCSMA makes more efficient use of the bandwidth, being able to provide good quality while the EDCA performance drops significantly. Future work will address the potential integration with HEW initiatives.

REFERENCES

[1] ISO/IEC 8802-11, ANSI/IEEE Std 802.11, “Wireless LAN medium access control and physical layer specifications amendment 8: Medium access control quality of service enhancements,” IEEE, 2005.

[2] C. Casetti and C.F. Chiasserini, “Improving fairness and throughput for voice traffic in 802.11e EDCA,” in *Proc. of IEEE Symposium on Personal, Indoor and Mobile Radio Communications (PIMRC 2004)*, vol. 1, Barcelona, Spain, Sep. 2004, pp. 525–530.

[3] T. Guo, C.H. Foh, J. Cai, D. Niyato, and E.W.M. Wong, “Performance evaluation of IPTV over wireless home networks,” *IEEE Trans. Multimedia*, vol. 13, no. 5, pp. 1116–1126, Oct. 2011.

[4] A. Vesco and R. Scopigno, “Time-division access priority in CSMA/CA,” in *Proc. of IEEE Symposium on Personal, Indoor and Mob. Radio Comm. (PIMRC 2009)*, Tokyo, Japan, Sep. 2009, pp. 2162–2166.

[5] ——, “A weakly synchronous and distributed coordination function for QoS management in CSMA/CA wireless access networks,” in *Developments in Wireless Network Prototyping, Design and Deployment: Future Generation*, M.A. Matin, Ed. Hershey, Pennsylvania: IGI Global, 2011.

[6] ——, “A distributed bandwidth management scheme for multi-hop wireless access networks,” in *Proc. of Wireless Comm. and Mob. Comp. Conf. (IWCMC)*, Istanbul, Turkey, Jul. 2011, pp. 534–539.

[7] ——, “Advances on time-division unbalanced carrier sense multiple access,” in *Proc. of IEEE Conference on Computer Communications and Networks (ICCCN 2011)*, Maui, Hawaii, Aug. 2011, pp. 1–6.

[8] A. Vesco, G. M. Toso, and R. Scopigno, “Prototyping of Time-Division Unbalanced Carrier Sense Multiple Access and First Experiments,” in *Proceedings of IEEE Global Communications Conference (GLOBECOM 2012)*, Anaheim, California, USA, Dec. 2012.

[9] A. Vesco, E. Masala, and R. Scopigno, “Efficient support for video communications in wireless home networks,” in *Proc. Intl. Conf. on Comp., Netw. and Commun. (ICNC)*, San Diego, CA, USA, Jan. 2013.

[10] A. Vesco, R. Scopigno, and E. Masala, “Supporting Triple-Play Communications with TDuCSMA and First Experiments,” in *Proc. IEEE Wireless Comm. and Netw. Conf. (WCNC)*, Apr. 2014, pp. 3302–3307.

[11] A. Vesco, F. Abrate, and R. Scopigno, “Convergence and performance analysis of leaderless synchronization in Wi-Fi networks,” in *Proc. of ACM Workshop PM2HW2N*, Miami, FL, Nov. 2011, pp. 49–58.

[12] S. Mangold, S. Choi, G.R. Hiertz, O. Klein, and B. Walke, “Analysis of IEEE 802.11e for QoS support in wireless LANs,” *IEEE Wireless Communications*, vol. 10, no. 6, pp. 40–50, Dec. 2003.

[13] E. Masala, A. Vesco, M. Baldi, and J.C. De Martin, “Optimized H.264 video encoding and packetization for video transmission over pipeline forwarding networks,” *IEEE Trans. Multimedia*, vol. 11, no. 5, pp. 972–985, Aug. 2009.

[14] “UCB/LBNL/VINT network simulator—ns (version 2.28),” 2005. [Online]. Available: http://www.isi.edu/nsnam/ns

[15] CDVL Technical Committee, “The consumer digital video library,” May 2012. [Online]. Available: http://www.cdvl.org

[16] M. Heusse, F. Rousseau, G. Berger-Sabbatel, and A. Duda, “Performance Anomaly of 802.11b,” in *Proc. IEEE Infocom ’03*, vol. 2, Mar. 2003, pp. 836–843.

[17] A. Raake, “Short- and long-term packet loss behavior: Towards speech quality prediction for arbitrary loss distributions,” *IEEE Trans. on Audio, Speech, and Lang. Proc.*, vol. 14, no. 6, pp. 1957–1968, Nov. 2006.

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