UNIVERSITY OF CASTILLA-LA MANCHA

Computing Systems Department

Improvements to Multimedia Content Delivery over IEEE 802.11 Networks

A dissertation for the degree of Doctor of Philosophy in Computer Science to be presented with due permission of the Computing Systems Department for public examination and debate

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UNIVERSIDAD DE CASTILLA-LA MANCHA

Departamento de Sistemas Informáticos

Improvements to Multimedia Content Delivery over IEEE 802.11 Networks

Tesis Doctoral presentada al Departamento de Sistemas Informáticos de la Universidad de Castilla-La Mancha para la obtención del título de Doctor en Tecnologías Informáticas Avanzadas

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A Gabriel,
por estar a mi lado en cada paso del camino.

A mis padres,
por enseñarme lo que soy.

Declaration

I declare that this Doctoral Thesis entitled "***Improvements to Multimedia Content Delivery over IEEE 802.11 Networks***" is a presentation of original work and is my own personal effort. This work has not previously been presented for examination at this or any other university.

The interpretation put forth is based on my reading and understanding and has not been taken from other sources, except where this is specifically indicated. All sources are properly acknowledged within the text as references.

Furthermore, I declare myself to be one of the main authors of the work used in this Thesis. In this regard, the following works published in JCR journals have been considered in the presentation of this Doctoral Thesis by compendium of publications:

- E. Coronado, J. Villalón, and A. Garrido, "An Adaptive Medium Access Parameter Prediction Scheme for IEEE 802.11 Real-Time Applications". *Wireless Communications and Mobile Computing*, 2017 [1]. (IF:1.899, Q2).
- E. Coronado, R. Riggio, J. Villalón, and A. Garrido, "Efficient Real-time Content Distribution for Multiple Multicast Groups in SDN-based WLANs". *IEEE Transactions on Network and Service Management*, 2017 [2]. (IF:3.134, Q1).
- E. Coronado, R. Riggio, J. Villalón, and A. Garrido, "Joint Mobility Management and Multicast Rate Adaptation in Software-Defined Enterprise WLANs". *IEEE Transactions on Network and Service Management*, 2018 [3]. (IF:3.134, Q1).

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Since I was a child I have been a whirlwind and a very headstrong person, and nobody knows it better than my parents, Antonio and Mª del Pilar. Sometimes I was so tired that I did not pay enough attention to you, even getting angry with you because of the stress and the frustration I felt. However, you have always been there for me whenever I have needed it and have encouraged me to follow my dreams. You always trusted me and were sure that I could achieve all my goals. I feel very grateful for your support and for taking care of all the things I could not during the Ph.D. Thank you for not getting angry (or not very angry) when I turned our house into a lab and filled it with cables, devices and laptops, or for all the mess I made when going to Italy and to conferences. It cannot be easy to be my parents, but I am very proud of being your daughter. A family is not always linked by blood ties and for that reason I would also like to thank one person that has treated me like her daughter, Llanos. Thank you for welcoming me into your home, for your calls and words of encouragement, and also for all the times you helped me when I did not even have time to cook.

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Summary

The breakthrough in wireless technologies and the tendency to be permanently connected have made wireless communications a fundamental aspect of daily life. The past few years have witnessed an explosion in high quality multimedia contents and a change in the consumption patterns of end users, who are increasingly using mobile devices equipped with IEEE 802.11 interfaces [4]. This fact, combined with the simplicity, low cost and multimedia support of the IEEE 802.11 standard, have led Wi-Fi networks to prevail on the market and have created new challenges in network performance and user experience.

The emergence of platforms such as Netflix or Youtube has increased the popularity of multimedia content distribution. However, the original IEEE 802.11 standard was not designed to support voice and video services. These services have a large impact in terms of throughput, latency and jitter on the quality perceived by users if not handled adequately. Therefore, effective Quality of Service (QoS) mechanisms are required. The IEEE 802.11e amendment [5] presented traffic differentiation capabilities by introducing Enhanced Distributed Channel Access (EDCA) as a new channel access method capable of classifying and prioritizing the traffic. Despite the benefits provided by this amendment, the performance of voice and video transmissions is not meeting expected standards, especially as the network load increases, and mobility and scalability aspects are involved.

QoS issues in Wi-Fi networks also arise due to interference between the access points. Although the use of the unlicensed spectrum greatly simplifies the deployment of this type of networks, it makes them extremely vulnerable to interference, collision problems and performance drops. Moreover, especially in large scale deployments, the inappropriate distribution of the users across the Access Points (APs) contributes to overloading some parts of the network while others are idle. As a result, inefficient resource assignment causes collision problems and greatly impairs the user experience.

The difficulty of providing QoS support increases in the case of multicast traffic. This type of transmission provides an efficient communication mode when the content must be simultaneously delivered to several users, as is the case of scenarios such as social and sports events, conferences and educational courses. Nevertheless, multicast transmissions in 802.11 networks face serious reliability and scalability problems due to the lack of acknowledgments and retransmissions. For this reason, the standard recommends the use of basic data rates, hence making the channel occupied for longer periods and affecting other network services. With the purpose of solving these problems, the IEEE 802.11aa amendment [6] introduces a set of retransmission policies to provide robust audio and video services in multicast mode. Despite these efforts, multicast rate selection and scalability problems are still important issues which remain to be solved.

Addressing the underlying shortcomings requires the introduction of major improvements. Nevertheless, the rigidity of the traditional network architecture, the increase in the size of the deployments and the coexistence of network devices from different manufacturers hinder this task. Software Defined Networking (SDN) has emerged as a new paradigm that is able

to reshape the network infrastructure by decoupling the control from the data plane. In this regard, the hardware abstraction provided for applications and network services makes network control fully programmable and allows it to shift network intelligence to a logically centralized controller that maintains a global view of the network. As a result, network management and the introduction of new solutions are greatly simplified.

In this Doctoral Thesis we explore multimedia content distribution over IEEE 802.11 networks. On the one hand, data mining and machine learning techniques are crucial in finding traffic patterns, which makes it possible to identify the most important determinants in the performance and QoS level of voice and video services. On the other hand, SDN enables a global vision of the network status, which is exploited to perform flexible and efficient resource allocation and to improve the performance of multicast communications.

In order to tackle the existing difficulties, this Doctoral Thesis presents three main approaches with the aim of improving the aspects affecting the transmission of multimedia applications. First, we endeavour to enhance the QoS level in 802.11 by improving the capabilities of EDCA. In this context, machine learning models are used to design a dynamic prediction scheme for the medium access parameters in EDCA. However, despite the gains achieved, and due to the difficulty of introducing new standard-compliant solutions in the MAC layer, there is still room for improvement in terms of throughput and collision reduction. In view of this, the second approach of the Thesis relies on the SDN paradigm to provide efficient radio resource management through adaptive channel assignment and traffic distribution methods. The last part of the Thesis seeks to address the shortcomings found in multicast multimedia transmissions. To this end, similarly to the previous case, an integral SDN-based solution is proposed to adapt the multicast rate transmission. Building on the rate selection algorithm, the inherent problems of the multicast transmissions are further addressed. First, a new scheme is introduced to enable seamless user mobility between the access points and select accordingly the transmission rate. Finally, an approach for efficiently managing multiple simultaneous multicast services is presented.

Resumen

El avance de las tecnologías inalámbricas y la tendencia a estar permanentemente conectados han convertido las comunicaciones inalámbricas en un aspecto fundamental en nuestra vida diaria. En los últimos años se ha producido una explosión de contenidos multimedia en alta definición y un cambio en los patrones de consumo de los usuarios, que cada vez más tienden al uso de dispositivos móviles, habitualmente equipados con interfaces IEEE 802.11 [4]. Este hecho, unido a la simplicidad, bajo coste y soporte multimedia del estándar IEEE 802.11, han llevado a las redes Wi-Fi a imponerse en el mercado y han abierto nuevos desafíos para mejorar el rendimiento y la experiencia del usuario.

El éxito de plataformas como Netflix y Youtube ha aumentado la popularidad de la distribución de contenidos multimedia. Sin embargo, el estándar IEEE 802.11 no fue originalmente diseñado para soportar servicios de voz y vídeo. Estos servicios poseen importantes requisitos en términos de rendimiento, latencia y retraso, ya que su tratamiento inadecuado puede degradar en gran medida la calidad percibida por los usuarios. Por ello, es necesario disponer de mecanismos de calidad de servicio o *Quality of Service* (QoS). La enmienda IEEE 802.11e [5] introdujo características de diferenciación de tipos de tráfico mediante la función de acceso al canal conocida como *Enhanced Distributed Channel Access* (EDCA), capaz de clasificar y priorizar el tráfico de la red. A pesar de los beneficios introducidos por esta enmienda, el rendimiento de las transmisiones en tiempo real no es el esperado, especialmente a medida que la carga de la red aumenta y entran en juego aspectos de movilidad y escalabilidad.

Los problemas de QoS en redes Wi-Fi aparecen igualmente debido la interferencia producida entre distintos puntos de acceso. A pesar de que el uso del espectro inalámbrico sin licencia facilita en gran medida la instalación de estas redes, este hecho las hace especialmente vulnerables a sufrir interferencias, un gran número de colisiones e importantes pérdidas de rendimiento. Además, especialmente en despliegues a gran escala, la distribución inapropiada de los usuarios sobre los puntos de acceso contribuye a sobrecargar algunas partes de la red mientras que otras se encuentran ociosas. Como consecuencia, la asignación ineficiente de recursos genera problemas de colisiones y afecta en gran medida la experiencia de los usuarios.

Las dificultades para proporcionar soporte de QoS se intensifican en el caso del tráfico multicast. Este tipo de transmisión proporciona un método de comunicación eficiente cuando los contenidos deben distribuirse de forma simultánea a varios usuarios, como es el caso de eventos sociales y deportivos, conferencias y cursos educativos. No obstante, el tráfico multicast en redes 802.11 posee serios problemas de fiabilidad y escalabilidad debido a la falta de confirmaciones y retransmisiones de paquetes. Por ello, el estándar recomienda el uso de la tasa básica de transmisión, aumentando así el tiempo que el medio inalámbrico permanece ocupado. Para solventar estos problemas, la enmienda IEEE 802.11aa [6] presenta un grupo de políticas de retransmisión con el objetivo de aportar una mayor robustez a las transmisiones de voz y vídeo en modo multicast. A pesar de ello, la adaptación de la velocidad de transmisión y los problemas de escalabilidad continúan siendo cuestiones sin resolver.

Solucionar las deficiencias presentadas anteriormente requiere la introducción de nuevos esquemas de mejora. Sin embargo, la rigidez de la arquitectura tradicional de red, el aumento del tamaño de las mismas y la integración de dispositivos de distintos fabricantes dificultan esta tarea. El concepto de *Software Defined Networking* (SDN) ha surgido recientemente como un nuevo paradigma capaz de proporcionar una abstracción del hardware de la red y de separar su funcionamiento entre el plano de control y el plano de datos. En este sentido, la introducción de abstracciones de programación de alto nivel permite desplazar gran parte de la lógica de la red hacia un controlador central y simplificar la introducción de nuevos esquemas de mejora.

Esta Tesis Doctoral propone mejorar el rendimiento de la distribución de contenidos multimedia sobre redes IEEE 802.11. Por un lado, las técnicas de minería de datos y *machine learning* resultan clave en la búsqueda de patrones de tráfico para identificar los parámetros más determinantes en el rendimiento y el nivel de QoS de las aplicaciones de voz y vídeo. Por otro lado, el enfoque basado en SDN permite disponer de una visión global de la red a través del controlador central, y que se emplea para realizar una gestión eficiente de los recursos y mejorar el rendimiento de los servicios multicast.

Para solventar las problemáticas anteriores, en esta Tesis Doctoral se emplean tres enfoques destinados a mejorar distintos aspectos de la transmisión de las aplicaciones multimedia. En primer lugar, se pretende aumentar el nivel de QoS mediante la mejora de las capacidades ofrecidas por EDCA. Con este fin, se emplean técnicas de *machine learning* para diseñar un esquema de predicción dinámico de los parámetros de acceso al medio. No obstante, a pesar de los resultados alcanzados y debido a la dificultad para introducir nuevos mecanismos en la capa MAC, todavía existe margen de mejora en términos de rendimiento y reducción de colisiones. El segundo pilar de esta Tesis se basa en el paradigma SDN para mejorar el uso de los recursos de la red mediante la asignación inteligente de los canales inalámbricos y la distribución dinámica del tráfico entre los puntos de acceso de la red. La última parte pretende solventar las deficiencias encontradas en las transmisiones multimedia en modo multicast. Para ello, y análogamente al caso anterior, se propone un nuevo enfoque basado en SDN que permite adaptar el caudal de transmisión en multicast. Tomando esto como base, se propone un nuevo esquema capaz de garantizar la movilidad transparente de los usuarios entre los puntos de acceso y adaptar el caudal de transmisión en modo multicast. Por último, se plantea un algoritmo que permite asegurar la gestión eficiente de aplicaciones multicast simultáneas.

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List of Acronyms

3GPP	3rd Generation Partnership Project
AC	Access Category
ACK	Acknowledgement
ADRR	Airtime Deficit Round Robin
AI	Artificial Intelligence
AIFS	Arbitration Interframe Space
AIFSN	Arbitration Interframe Space Number
AP	Access Point
BACK	Block ACK
BE	Best Effort
BK	Background
BSSID	Basic Service Set Identifier
CSA	Channel Switch Announcement
CTS	Clear To Send
CW	Contention Window
DCF	Distributed Coordination Function
DIFS	DCF Interframe Space
DMS	Directed Multicast Service
EDCA	Enhanced Distributed Channel Access
FEC	Foward Error Correction
GATS	Group Addressed Transmission Service
GCR	Groupcast with Retries
HCF	Hybrid Coordination Function

HD	High Definition
IEEE	Institute of Electrical and Electronic Engineers
IGMP	Internet Group Management Protocol
IP	Internet Protocol
IPTV	Internet Protocol Television
KDD	Knowledge Discovery from Data
LBP	Leader-Based Protocol
LVAP	Light Virtual Access Point
MAC	Medium Access Control
MBMS	Multimedia Broadcast Multicast Service
MCS	Modulation and Coding Scheme
MINLP	Mixed Integer Non Linear Programming
NFV	Network Function Virtualization
PHY	Physical layer
PSNR	Peak Signal to Noise Ratio
QoE	Quality of Experience
QoS	Quality of Service
RSSI	Received Signal Strength Indicator
RTS	Request To Send
SDN	Software Defined Networking
SIFS	Short Interframe Space
SNR	Signal-to-Noise Ratio
TXOP	Transmit Opportunity
UR	Unsolicited Retries
VI	Video
VO	Voice
WBA	Wireless Broadband Alliance
WLAN	Wireless Local Area Network

CHAPTER 1

Introduction

Wireless Local Area Network (WLAN) technology was initially designed to provide local radio access to a limited number of users. However, the ongoing proliferation of 802.11 Wi-Fi networks [4], the explosion in the number of connected devices and the increasing consumption of high quality multimedia content are driving an insatiable demand to improve. More specifically, this leads to a search for a scalable end-to-end infrastructure capable of supporting the exponential traffic growth from millions of users and delivering high Quality of Service (QoS) to ensure a satisfactory user experience.

In this chapter we first describe the landscape of multimedia content delivery over IEEE 802.11 networks in order to set out the main motivations behind this Doctoral Thesis. Second, we introduce the main goal pursued in this Thesis and the methodology defined to achieve this objective. Finally, we present the contributions of the Thesis and the results obtained from this work.

The rest of the Thesis is organized as follows. In Chapter 2 an overview of the publications that support this work is provided. Moreover, the full citations, and the information concerning the relevance and the current status of the articles are also given. In Chapter 3 we draw our conclusions and summarize the main contributions of this dissertation. Finally, we provide some future lines of work and present the research challenges envisioned for WLANs.

1.1. Motivation

1.1.1. Overview

The past few years have witnessed a marked increase in the demand for wireless technologies due to the explosion in the number of Wi-Fi enabled mobile end-user devices such as smartphones and tablets. Due to their simple deployment, low operational cost and multimedia content support, IEEE 802.11 networks are used in daily operations in both professional and personal spheres, including online gaming, video streaming, live television and social media. Furthermore, Wi-Fi traffic is expected to continue growing over the next few years and, according to the Cisco Forecast, Wi-Fi and mobile traffic will account for more than 63% of total IP traffic by 2021 [8].

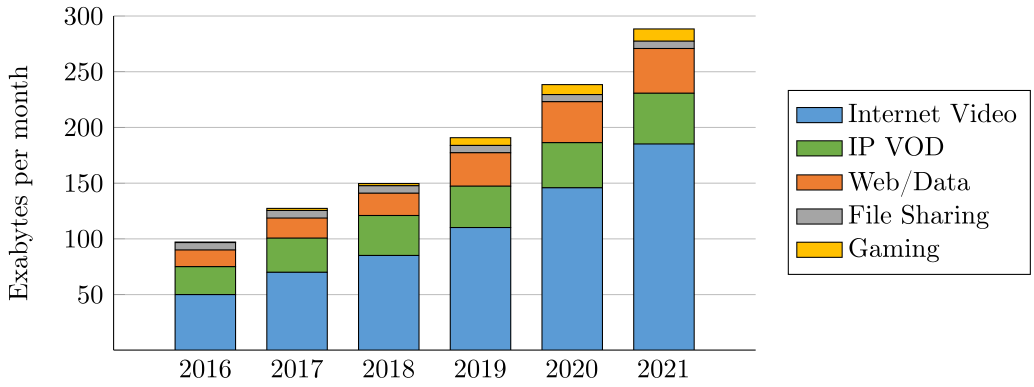


Figure 1.1: Global IP traffic by application category. *Source*: Cisco Visual Networking Index (VNI) 2016-2021 [7].

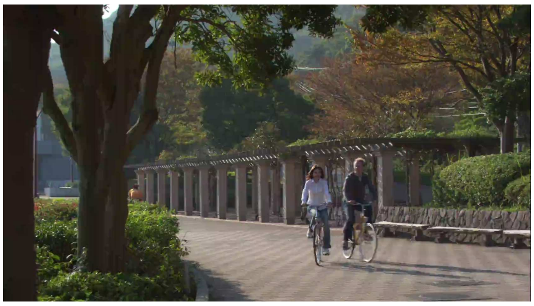
Wi-Fi is one of the most widespread technologies for the deployment of WLANs in residential and business areas, and they have increased in popularity due to their maturity, their high performance and the use of unlicensed frequency bands. As a matter of fact, according to the Wireless Broadband Alliance (WBA), the number of Wi-Fi hotspots (including public and home WLANs) will increase 6-fold from 2016 to 2021 and the number of setups will amount to 541.6 million by 2021 [9]. With the advent of cellular networks such as 5G, 802.11 networks are currently becoming an interesting solution for traffic offloading [10]. Usually, Wi-Fi rates are on average higher than cellular speeds, which creates user preferences in favor of Wi-Fi. In this sense, the amount of traffic offloaded from 3G and 4G networks will reach 55% and 66%, respectively, by 2021. Moreover, although the offload percentage of 5G networks will be less than 50%, this figure is expected to grow as 5G technology matures [8].

1.1.2. Challenges

Recently, applications such as interactive gaming and video and audio streaming have become a significant portion of IP traffic and, as illustrated in Figure 1.1, video services are expected to represent 82% of all consumer Internet traffic by 2021 [7]. Most of these services are delay sensitive and, although increasing the overall throughput is a significant matter, providing end-to-end QoS support acquires even greater importance. Moreover, as shown in Figure 1.2, for a video application, the Quality of Experience (QoE) perceived by the users can be severely compromised. However, these capabilities are not provided by the original IEEE 802.11 standard.

Challenges in QoS Provisioning at the MAC Layer

In the original version of the IEEE 802.11 standard, the Distributed Coordination Function (DCF) controls medium access. In this mode, the stations must sense the channel as free for a period given by the DCF Interframe Space (DIFS) to start a transmission. If after this period the channel is busy, the stations must start the backoff algorithm, a contention method for collision avoidance. To this end, a random value in the interval $[0, \mathrm{CW}-1]$ is chosen, where



(a) Video stream transmitted without errors.

(b) Video stream transmitted with errors.

Figure 1.2: Differences in the visual perception between a video sequence with and without transmission errors.

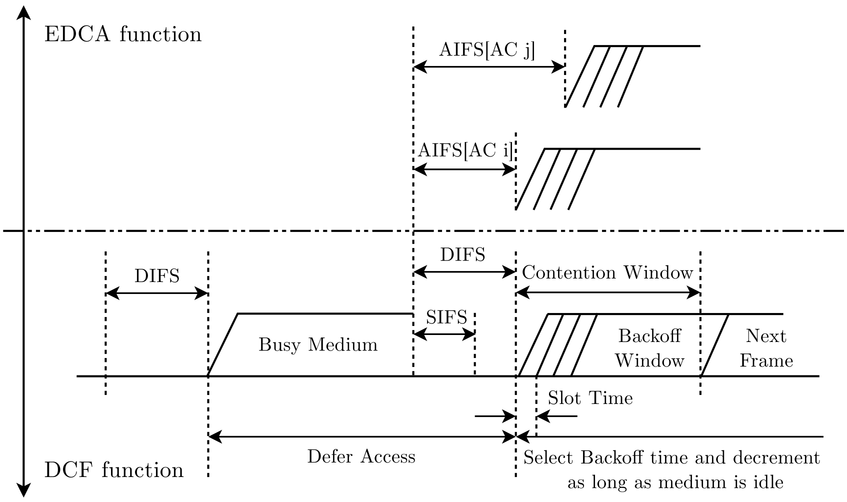


Figure 1.3: Interframe spaces in the DCF and EDCA mechanisms.

the value of the Contention Window (CW) is given by the physical layer and is represented by its minimum and maximum limits, CW_{min} and CW_{max}. Initially, the size of CW is set to CW_{min}. After n erroneous transmissions, CW is increased following a sequence of 2^n until CW_{max} is reached. This procedure is carried out to set the period of time that a station must wait before scheduling a retransmission. In case of reaching the maximum limit, the frame is silently dropped. In this regard, the selected value for the backoff algorithm is decreased by one unit every time the medium is sensed as idle for a DIFS period. In this way, the transmission begins when the counter reaches zero. As a result, the same priority is assigned to all the users and applications, thus rendering the traffic differentiation. This process is shown in Figure 1.3, in which the Short Interframe Space (SIFS) represents the amount of time used by high priority actions such as the ACKs.

In order to guarantee robust communications, the 802.11 standard specifies a two-way handshake protocol for unicast data transmission in which each client must confirm frame reception. However, this protocol is not effective in the presence of hidden nodes in the network. To address this problem, the standard introduces the RTS/CTS protocol. This

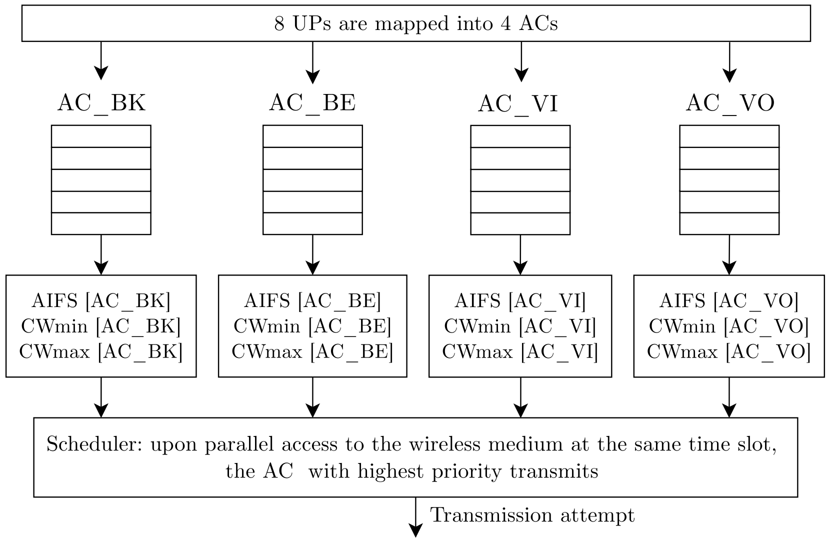


Figure 1.4: EDCA access categories and priorities in IEEE 802.11e.

method allows a station to indicate the intention to begin a transmission by sending an RTS frame to the Access Point (AP). Upon its reception and if the medium is idle, the AP broadcasts a CTS frame to all the users in the network. Therefore, regardless of their location, this protocol makes it possible to inform all the stations about the beginning of a new transmission. As a result, collisions are reduced and the performance of the network is improved if hidden stations are present.

The IEEE 802.11e amendment [5] introduces Enhanced Distributed Channel Access (EDCA) as a medium access function capable of classifying and prioritizing traffic streams. EDCA extends the capabilities of DCF and introduces four traffic Access Categories (ACs), as depicted in Figure 1.4. In EDCA each AC, named Voice (VO), Video (VI), Best Effort (BE) and Background (BK) from highest to lowest priority, makes use of its own traffic queue and has its own set of medium access parameters: Arbitration Interframe Space (AIFS), Transmit Opportunity (TXOP) and CW. In contrast to DCF, EDCA specifies different waiting times for each type of traffic, as seen in Figure 1.3. Given that the legacy stations cannot modify their medium access parameters, the standard establishes a fixed set of values for EDCA with the aim of ensuring interoperability with these stations while improving the performance of the delay sensitive traffic. Even though 802.11e allows it to perform changes in the EDCA parameters to adapt them to network conditions, this feature is not used in commercial APs due to the complexity involved and the necessity of carrying out this operation in real-time.

In EDCA a station must sense the channel idle for a period equal to AIFS before attempting a transmission. Similarly to DCF, the backoff algorithm is started in case of finding the medium busy and a random value in the interval $[0, \mathrm{CW}-1]$ is chosen. Nevertheless, using the same AIFS and short periods for the CW size for the voice and video ACs causes a great number of erroneous transmissions and has important negative effects on QoS.

Although EDCA delivers some QoS improvements, it has been demonstrated that the features provided by 802.11e do not accurately handle voice and video traffic [11, 12, 13].

Especially in scenarios with high traffic load or holding a considerable number of users, this issue causes performance drops, packet losses, high delay and transmission errors. In this regard, different studies have shown that the appropriate selection of the EDCA parameters leads to efficiency enhancements and contributes to reducing the delay and the collisions in the network. However, only the CW and the AIFS parameters are associated with the highest improvements [14, 15, 16]. With the purpose of providing QoS guarantees, some research works focus on adapting the EDCA parameters [17, 18, 19], designing analytical models [20, 21], and modeling their own medium access function [19]. Nevertheless, in spite of outperforming EDCA, most of these proposals are not compatible with the standard, make unreal assumptions or introduce unnecessary traffic overheads. For this reason, this Doctoral Thesis aims to introduce a predictive scheme capable of dynamically tuning the EDCA parameters in real-time without modifying the 802.11 standard.

Challenges in Channel Selection and User Association

QoS in delay sensitive services is also determined by significant collision probability and interference, especially when the number of devices increases. In addition to these pitfalls, in WLANs users are in charge of selecting the AP for the association given that the standard does not specify a procedure to perform this operation, and it is usually left as a free decision for the vendor. As a design simplification, clients perform active scans and normally select the AP that offers the strongest Received Signal Strength Indicator (RSSI) without taking into consideration the traffic load of such an AP [22, 23]. However, this fact may lead to an inefficient distribution of the clients across the APs and to an increase in the collisions. Furthermore, a limited number of radio channels are available in Wi-Fi networks, which may result in overlaps in the coverage area of several APs.

Resource scheduling can bring significant benefits in terms of QoS. Specifically, in scenarios where low latency requirements must be simultaneously guaranteed for multiple users, an effective collision domain isolation and channel assignment strategy acquires particular importance. With the aim of addressing this situation, significant research has been conducted. Some proposals aim to minimize the number of clients per AP [24, 25], while others consider the average network workload to redistribute the traffic [26, 27], or use a group of parameters (e.g. user location and link quality) to perform the association [28]. The problem has also been formulated using Mixed Integer Non Linear Programming (MINLP) to compute the difference in terms of the bandwidth demand of the users [29], and has been modeled through graph theory by considering channel utilization and interference of the APs [30]. Some other approaches require the APs to operate on non-overlapping channels to reduce the interference, thus limiting the size of the deployment [31, 32]. However, the required changes to the 802.11 standard of some of these proposals show that the traditional architecture is often inflexible and ignores the specific requirements of users and applications.

Since the original 802.11 standard was released, the IEEE has worked to improve the performance of Wi-Fi networks by increasing transmission rates and addressing the limitations found in the MAC layer. Nevertheless, this process is very slow and, together with network rigidity, hampers the quick introduction of efficiency improvements. Moreover, the need to ensure compatibility between the different amendments to the standard prevents WLANs from

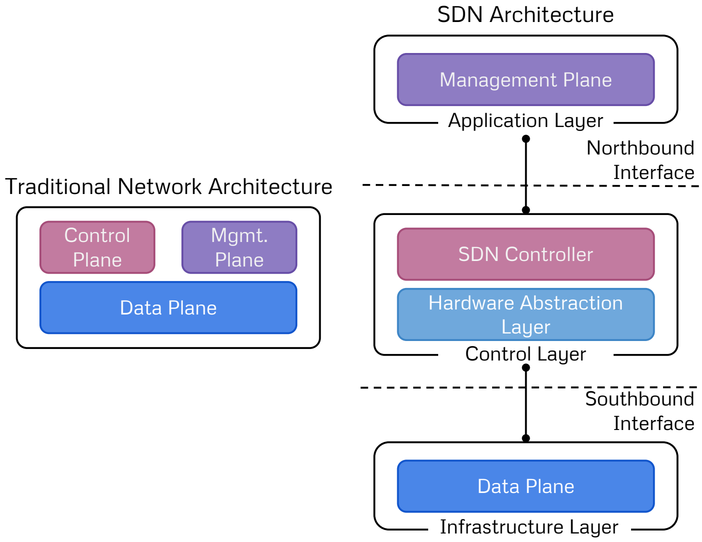


Figure 1.5: Overview of the traditional and the SDN network architectures.

reaching their pick theoretical performance and exploiting all their potential. Furthermore, Wi-Fi APs represent the last wireless hop for the users, which is usually the critical point in terms of network performance, and delay and bandwidth issues may arise. This situation creates new challenges for 802.11 networks to provide QoS guarantees and simplify network management. Intelligent radio resource management strategies can notably improve network performance. Nevertheless, the complexity involved in introducing new mechanisms due to network ossification constitutes a bottleneck.

Recent research efforts in the wired domain have focused on the Software Defined Networking (SDN) paradigm, a promising technology to enable network programmability and overcome the rigidity of the traditional network architecture. SDN has emerged as a novel way of refactoring network functions by effectively decoupling the data-plane from the control-plane, and by outsourcing the network intelligence to a logically centralized controller. This novel network architecture is presented in Figure 1.5. OpenFlow [33] is normally used as the de-facto standard to implement the protocol between the control and the data planes. However, its features are not able to tackle the complexities of wireless networks, which comprise design aspects such as association and rate control. As a result, equivalent solutions in wireless and mobile networks have recently started to appear [34, 35, 36].

SDN is a good solution to tackle the challenges created by the lack of coordination in the traditional deployments. User distribution techniques inevitably involve reassociation processes that can degrade network performance due to the time required to reconnect to the new AP. Especially when delivering multimedia content, this concern shows the need for supporting seamless migration mechanisms given that interruptions in the transmission severely impair the QoS level and the satisfaction of the end-user. Nonetheless, the rigid network architecture is not capable of supporting these features. As a result, the problem has recently been studied from the SDN perspective in order to explore the reconfiguration of the transmission power of the APs [37] and load balancing over different connection nodes (i.e. 2.4 and 5 GHz Wi-Fi, and Ethernet interfaces) [38, 39, 40]. In this sense, the studies have shown the usefulness of the SDN paradigm when it comes to introducing capabilities that can optimize resource management in Wi-Fi networks, and which would dramatically improve user experience. On this basis, this Doctoral Thesis explores the options available in wireless SDN to improve user association and channel assignment in WLANs.

Challenges in Multicast Transmissions

Although web applications such as Youtube constitute a significant proportion of the video traffic, scenarios such as live events, conference meetings and Internet Protocol Television (IPTV) have witnessed unprecedented growth in recent years. They are good examples of use cases in which multicast is an efficient transmission mode to simultaneously deliver the same data to several users [41]. In fact, the use of unicast frames would greatly increase the network load, especially as the number of receivers rises. In view of this, multicast applications are suitable for business and entertainment purposes as is shown, for instance, in the deployment of campus network IPTV systems [42]. Multicast services are also experiencing a considerable growth in other radio access networks such as the emerging 5G networks, and this has led to the release of the Multimedia Broadcast Multicast Service (MBMS) by 3GPP [43]. This fact, coupled with the high bandwidth and low latency requirements of the delay sensitive applications, especially in mobile environments, shows the importance of ensuring reliable transmissions and transparent user mobility in multicast environments.

The two-way handshake protocol in unicast makes it possible to gather feedback information from the users and select the transmission rate from among different Modulation and Coding Scheme (MCS) indexes. However, multicast communications in 802.11 lack frame acknowledgments and retransmissions. In the IEEE 802.11 standard, multicast is specified as a simple broadcast mechanism and does not include error control or recovery mechanisms. Since the acknowledgments would inevitably collide at the transmitter, this design decision aims to prevent the frame implosion that would happen as a result. Accordingly, the absence of feedback brings several problems. On the one hand, it prevents the data from being retransmitted in the case of errors. Moreover, the basic access scheme must be utilized, which precludes the use of the RTS/CTS procedure to protect the frame transmission. On the other hand, it makes it impossible to adapt the data rate to the user conditions. For this reason, the standard recommends the use of the lowest data rate in order to increase the probability of all the users receiving the data. This becomes particularly important for the receivers under bad channel conditions because they are more likely to suffer interference and collision problems. As a consequence, the wireless medium is occupied for longer periods, which increases the radio resource consumption and impairs the capacity available for other network services. In this context, there is a clear necessity for mechanisms capable of ensuring robust multicast transmissions in order to meet the QoS requirements in Wi-Fi networks.

The IEEE 802.11aa amendment [6] aims to support robust audio and video transport streaming and address the limitations caused by the lack of feedback. To this end, it introduces Group Addressed Transmission Service (GATS) as a new mechanism that encompasses two new services named Directed Multicast Service (DMS) and Groupcast with Retries (GCR). DMS offers the same reliability level as the unicast transmissions since each frame is converted into as many unicast frames as the number of receivers in the multicast group. Hence,

Table 1.1: Overview of the highlights and limitations of the GATS service in 802.11aa.

Scheme	Complexity	Efficiency	Reliability	Scalability	Suitable scenarios
Legacy	Low	Low	Low	High	Low bitrate services, good channel conditions
DMS	Moderate	Variable	High	Low	Small groups, average channel conditions
UR	High	Moderate[1]	Moderate[1]	High	Large groups, good or moderate channel conditions
BACK	Moderate	Moderate[2]	High	Moderate	Moderate number of receivers, average channel conditions

[1] Depending on the number of retries.
[2] Depending on the number of receivers per group.

this solution is not valid for large groups. Conversely, GATS consists of three retransmission methods: legacy multicast, Unsolicited Retries (UR) and Block ACK (BACK). Legacy multicast refers to the basic multicast scheme defined in the original standard. UR retransmits the same packet N times without waiting for the ACKs in order to increase the probability of success. Finally, BACK provides higher reliability by allowing each user in the group to confirm the reception of several frames at the same time. Nevertheless, these two policies are still inefficient: UR does not adapt the data rate or the number of retransmissions to the channel conditions and introduces an unnecessary traffic overhead, while BACK continues facing scalability problems derived from the number of acknowledgments.

The main highlights and limitations of 802.11aa have been widely examined in the literature [44, 45, 46], and are summarized in Table 1.1. Despite the improvements achieved, it has been shown that each of these mechanisms is only suitable for a specific type of scenario. Nonetheless, the decision on the policy used is left to the implementer. Moreover, the standard does not provide any procedure to adapt the transmission data rate in multicast. Motivated by these drawbacks, some works propose several solutions to address the reliability and rate selection problems. Foward Error Correction (FEC) techniques [47, 48, 49, 50] encode the information in a redundant manner for error recovery over unreliable channels. By contrast, Leader-Based Protocols (LBPs) select a group leader (usually the receiver exhibiting the worst signal quality) that is responsible for confirming frame reception on behalf of the group members [51, 52, 53]. Finally, QoE measurements have also driven rate selection in multimedia applications [54, 55]. Additionaly, recent SDN-based solutions steer the transmission rate of the multicast applications by taking advantage of the global view of the SDN controller [56, 57, 58].

In view of the increasing number of mobile devices and the QoS requirements of streaming services, multimedia content distribution must provide support for roaming users, whose channel conditions constantly change. In this context, streaming services are especially sensitive since, in addition to the rate selection problem, they must include efficient mobility management mechanisms. This target has been pursued by different proposals in the field with the aim of reducing the handover delay and guaranteeing the experience of the end-user. Even though traditionally RSSI measurements drive this function, other approaches are based on QoS and QoE requirements [59], or pursue energy saving goals [60, 61]. This problem has also been analyzed from the perspective of the wired backbone interconnecting the APs [62, 63]. Nevertheless, these proposals usually neglect the specific challenges that arise in the radio access segment. Taking all this into account, it is important to mention that efficient operations across multiple APs require global visibility, which is not usually provided by traditional network architectures. Although the concept of seamless mobility has motivated the emergence of SDN-based approaches [61, 64], nowadays they are targeted at unicast traffic and do not address the special necessities of multicast services.

The demand for multimedia content has become more pronounced because of the introduction of High Definition (HD) and 4K video streaming. This latter technology has a bitrate that is more than twice that of HD and nine times higher than that of standard video definition. Therefore, this fact illustrates that bandwidth is a critical factor in streaming services. Moreover, the expanding ecosystem of mobile devices and the increasing availability of these services in diverse events such as conferences and social events show the need to support simultaneous streaming applications in a scalable fashion. In this regard, network functions must adequately manage several multicast sessions and the resources assigned to each of them. Nevertheless, despite the research efforts made [65, 66], very little progress has been made in practice in integrating rate adaptation features while ensuring high scalability, and this fact is still an open issue. As with the challenges described above, this Doctoral Thesis aims to design a scalable scheme for multicast data rate selection that enables seamless user mobility and the management of several multicast sessions.

In view of this, it is easy to see that voice and video services represent an important part of IP and Internet traffic. Moreover, the versatility of WLANs, their multimedia content support and the high performance of the 802.11 standard are generating renewed interest in the technology. However, the rigidity of the network stack means that QoS provisioning in 802.11 is not sufficient to support the diverse requirements that characterize these services. Since most of the operations are performed at the APs, they rarely consider the individual demands of the applications, such as packet loss or jitter. In this regard, SDN introduces new opportunities for fine-grained application control and QoS guarantees. In fact, although wireless networks virtualization has just started to appear, the potential of the paradigm has shown the benefits that can be brought to future Wi-Fi networks.

1.2. Objectives

The main objective of this Doctoral Thesis is to design an integrated framework for providing QoS support for delay sensitive applications over IEEE 802.11-based WLANs. As we will explain in further detail in the following section, this objective rests on two fundamental pillars: machine learning techniques and network programmability offered by the Software Defined Networking paradigm.

The approaches presented in this dissertation aim not only to ensure a high-quality user experience for voice and video applications, but also to guarantee the performance of other services in the network. To achieve this objective, the set of goals defined for this Thesis is the following:

- **Goal 1**. To review the state of the art, to acquire a deeper knowledge of wireless communications and, specifically, to explore the details of the IEEE 802.11 standard. Furthermore, the main related work in the literature focused on achieving optimal QoS support in unicast and multicast modes should also be reviewed.
- **Goal 2.** To assess the performance of multimedia transmissions, to understand the complexity involved in managing and coordinating multiple network devices and wireless clients, and to identify the most determinant QoS factors involved in the transmission of voice and video traffic.
- **Goal 3.** To improve channel access in IEEE 802.11e. After identifying the main QoS factors in voice and video transmissions, they are used to dynamically adapt the medium access parameters defined by the standard in the EDCA function.
- **Goal 4.** To study the Software Defined Networking paradigm, and in particular, to explore how the main benefits of abstracting the insights into the network architecture are applied to the wireless domain.
- **Goal 5.** To enable efficient channel selection and user association in WLANs, to mitigate interference impact and to coordinate the APs with the purpose of preventing them from simultaneously using the same resources.
- **Goal 6.** To enable robust multimedia transmissions and seamless user mobility in multicast environments, in order to improve the performance of live applications targeted at multiple users in Wi-Fi networks.
- **Goal 7.** To orchestrate and schedule multiple simultaneous multicast services, in order to enable the efficient transmission of several delay sensitive applications to different types of users at the same time.

1.3. Methodology and Work Plan

In order to achieve the objective introduced in the previous section, we apply a research methodology that divides each specific goal into a set of tasks. Establishing a relationship between all these tasks allows it to address the formulated problem from different points of view. As a consequence, this enables the design of a solution that improves the QoS delivered in multimedia content distribution over WLANs using different strategies.

The research work in this Thesis is based on 802.11 networks, whose setups can be separated into two groups. The experiments in the first part of the dissertation, covering *Goals 2-3*, are conducted via simulation[1]. Despite the modeling work, the proposals in this part of the

[1] We use Riverbed Modeler [67] to implement the proposals via simulation.

Thesis are standard compliant and the design decisions aim to maintain compatibility with current commercial devices. The second part of the work, encompassing *Goals 5-7*, is performed on a real-world testbed based on SDN principles to assess the efficiency of the proposals in real environments[2].

The tasks performed to achieve the main objective of this Doctoral Thesis are described below, and grouped by goal:

- **Goal 1**. *To review the state of the art.*
  - To study the IEEE 802.11 standard in order to know its functionality and performance limitations, with particular emphasis on analyzing the working mode of the medium access functions.
  - To study the IEEE 802.11e amendment, which is aimed at providing QoS facilities for bandwidth-sensitive applications such as voice and video. Specifically, particular attention is paid to EDCA, a contention-based channel access method that delivers traffic based on differentiated ACs.
  - To review the main research work dealing with QoS provisioning in WLANs from various points of view such as link layer solutions related to medium access parameters, schemes targeted at other layers (IP and application levels), and analytical models, among others.
  - To explore the features of data mining and machine learning techniques and how they may be applicable to the specific problem of QoS aware service delivery.
  - To analyze the behavior of the multicast transmission mode in Wi-Fi networks, as well as the reliability features and retransmission policies introduced in the IEEE 802.11aa amendment.
  - To review the related research work concerning rate adaptation mechanisms and reliability improvements for multicast in WLANs.
- **Goal 2.** *To assess the performance of multimedia transmissions.*
  - To determine the importance of the requirements of voice and video applications and the QoS indicators such as minimum amount of packet loss, delay and jitter.
  - To identify the parameters that have a significant impact on the performance of these services and on the overall user experience.
- **Goal 3.** *To improve channel access in IEEE 802.11e.*
  - To study network behavior when using alternative values for the EDCA medium access parameters.
  - To perform a set of tests considering various traffic conditions such as varying the number of stations, several types of applications and different EDCA values.
  - To conduct a Knowledge Discovery from Data (KDD) process in order to clean, reduce and integrate the data obtained from the previous tests. This process aims

[2]We run the SDN-based proposals in real scenarios on the 5G-EmPOWER platform [34].

to discover the traffic patterns that deliver the highest throughput for voice and video applications and to identify the most relevant factors involved.

- To implement and test two predictive algorithms (*M5* regression model and *J48* classifier tree) for adapting the AIFS combination in EDCA.
- To build on these algorithms to adapt the CW parameter in EDCA. The performance of the algorithms considering both factors must also be assessed.
- To design and test a unified dynamic algorithm for the EDCA parameters that is capable of discerning the use of each of the designed predictive models according to the network conditions.

- **Goal 4.** *To study the Software Defined Networking paradigm.*
  - To analyze the principles behind the SDN paradigm and its basic architecture, including the separation in the logic of the network between the control and data planes.
  - To review the existing SDN solutions available for radio access networks and, in particular, for Wi-Fi networks.
  - To understand the programmability of the IEEE 802.11 data path. This is composed of the upper-level MAC functionality, encompassing aspects related to each wireless client state such as association and authentication, and the lower-level MAC characteristics, such as transmission rate and power control.

- **Goal 5.** *To enable efficient channel selection and user association in WLANs.*
  - To design an SDN-based channel assignment algorithm to reduce the number of APs in the same collision domain.
  - To implement the signaling protocol between an SDN controller and the APs to transparently readjust the working channel based on the medium conditions.
  - To understand and implement the Channel Switch Announcement (CSA) control defined in the IEEE 802.11 standard in order to allow the wireless clients to be aware of channel changes in the serving AP.
  - To design and test an SDN-based load balancing algorithm to efficiently manage user mobility and enable seamless handover between APs.

- **Goal 6.** *To enable robust multimedia transmissions and seamless user mobility in multicast environments.*
  - To evaluate the performance of the standard multicast schemes in IEEE 802.11aa for multimedia applications in terms of QoS and QoE.
  - To design a hardware abstraction layer capable of enabling multicast transmissions in SDN-based WLANs and implement the multicast retransmission policies introduced in the IEEE 802.11aa amendment.
  - To implement the signaling required for the southbound communications between the SDN controller and the APs to configure specific retransmission policies for each multicast group.

  - To design and evaluate an SDN-based algorithm for the dynamic data rate control in multicast transmissions in real-world scenarios.
  - To design and evaluate an SDN-based algorithm for managing user mobility in multicast, in such a way that it enables seamless handover and reduces the radio resource utilization.

- **Goal 7.** *To orchestrate multiple simultaneous multicast services.*
  - To evaluate the performance and channel availability requirements to support and manage multiple multicast groups at the same time.
  - To implement the Internet Group Management Protocol (IGMP) with the aim of identifying and reporting to the SDN network controller on the formation of new multicast groups.
  - To design a protocol between the APs and the SDN controller to allow the latter to be aware of the changes in group membership.
  - To implement and evaluate an SDN-based algorithm in charge of computing the data rate for each multicast group and efficiently scheduling their transmissions based on the effective coordination of multicast retransmission policies.

1.4. General Discussion and Main Contributions

In the above sections we have presented the reasons that have motivated this Doctoral Thesis. As can be deduced from the objectives, the aforementioned reasons demonstrate that there is an incessant need for QoS guarantees in several aspects of a WLAN. However, especially when delay sensitive services are involved, several factors can limit network performance and the problem must be tackled from different perspectives.

1.4.1. Contributions to QoS Provisioning at the MAC Layer

The IEEE 802.11e extension establishes a baseline in QoS provisioning that attracts great research efforts at the MAC layer to provide reliability and access coordination to the wireless medium. In this respect, several studies demonstrate the importance of traffic prioritization and the limitations of the EDCA function for this purpose. As stated in Section 1.1, although IEEE 802.11e allows the use of dynamic values for the EDCA parameters, it recommends a specific set of values to guarantee compatibility with all Wi-Fi compliant devices (including those without QoS support). Accordingly, the values assigned to the AIFS, CW and TXOP parameters in real networks are set by the standard and are not updated during the transmissions. These values are shown for each AC in Table 1.2. In this table, the AIFS parameter is derived as $AIFS[AC] = AIFSN[AC] \cdot Slot\,Time + SIFS$, where the Arbitration Interframe Space Number (AIFSN) ensures that each AC has a different priority for accessing the channel, and the *Slot Time* denotes the duration of a slot according to the physical layer.

Motivated by this conclusion, we first analyze via simulation the performance of the standard EDCA combination. This study shows that the number of collisions in the network

Table 1.2: Default EDCA Parameter set for each AC in IEEE 802.11e.

AC	CW_{min}	CW_{max}	AIFSN	TXOP
AC_BK	aCW_{min}	aCW_{max}	7	-
AC_BE	aCW_{min}	aCW_{max}	3	-
AC_VI	$(aCW_{min}+1)/2-1$	aCW_{min}	2	3.008 ms
AC_VO	$(aCW_{min}+1)/4-1$	$(aCW_{min}+1)/2-1$	2	1.504 ms

arises from a medium traffic load level, especially for voice and video applications. This is due to the fact that these ACs use the same AIFSN and a small value for the CW parameter. The 802.11e amendment establishes these values with the aim of ensuring a higher priority for the voice and video applications with regard to the stations that use DCF, i.e. with regard to the stations that only support the original standard. In view of this, the collision ratio can be decreased in two ways. On the one hand, the collisions between voice and video services can be lowered by shifting the AIFSN values of the different ACs. Therefore, voice applications would have an exclusive period to access the channel. On the other hand, the CW size of the voice and video ACs can be adjusted. However, despite the collision reduction, this approach may impair the performance of the voice and video applications when the network is partially composed of stations using DCF. For this reason, it is very important to distinguish between scenarios with and without the presence of legacy stations.

In this context, to dynamically adapt the medium access parameters in EDCA it is essential to identify the presence of stations without QoS support and the changes in the network conditions over time. To this end, the most relevant factors in determining the network status must be identified. In this regard, we have analysed the impact of the number of active transmissions of each type of traffic, the bitrate of the applications, the transmission rate, the presence of legacy DCF stations and the channel utilization. Nevertheless, the EDCA parameters must be updated in real-time, which requires the implementation of a scheme of low computational complexity and involving as few variables as possible. In this sense, the channel utilization of each AC in EDCA together with that of DCF are able to provide an accurate approximation of the network status and summarize the information derived from the remaining parameters. Therefore, these factors are taken as a reference for tuning the medium access parameters.

Artificial Intelligence (AI) techniques are a good way to identify traffic patterns that can be used to make network management decisions. Specifically, supervised learning algorithms have been considered in this work. These algorithms use a data set composed of n input features and an output variable, Y. In this way, a pair (X, Y) whose outputs are currently known allows the prediction of other unlabeled data (y) when only the input features (x) are known. In this context, X refers to the medium access parameters in EDCA, whereas the output Y represents the performance achieved by the multimedia applications. In this scenario, it is necessary to select predictive models that are capable of being used in real-time and whose results can be easily interpreted. On this basis, after analysing several options, we have made use of a *J48* classifier tree [68] and an *M5* regression model [69].

Table 1.3: AIFSN values analysed for the medium access parameters prediction scheme.

AC	**S0**	**S1**	**S2**	**S3**	**S4**	**S5**	**S6**	**S7**	**S8**	**S9**
BK	7	8	9	8	9	12	10	12	14	14
BE	3	4	5	4	5	6	6	8	10	12
VI	2	2	2	3	3	3	4	5	6	7
VO	2	2	2	2	2	2	2	2	2	2

Table 1.4: Alternative values for CW for the medium access parameters prediction scheme.

AC	**CW 1**		**CW 2**	
	$\mathbf{CW_{min}}$	$\mathbf{CW_{max}}$	$\mathbf{CW_{min}}$	$\mathbf{CW_{max}}$
BK	$2 \cdot \text{aCW}_{\text{min}} + 1$	aCW_{max}	$2 \cdot \text{aCW}_{\text{min}} + 1$	aCW_{max}
BE	$2 \cdot \text{aCW}_{\text{min}} + 1$	aCW_{max}	$2 \cdot \text{aCW}_{\text{min}} + 1$	aCW_{max}
VI	aCW_{min}	$2 \cdot \text{aCW}_{\text{min}} + 1$	$2 \cdot \text{aCW}_{\text{min}} + 1$	$2 \cdot \text{aCW}_{\text{min}} + 1$
VO	$(\text{aCW}_{\text{min}}+1)/2\text{-}1$	aCW_{min}	aCW_{min}	aCW_{min}

In order to achieve this purpose, a three-phase predictive scheme is proposed. The first phase is targeted at adapting the AIFSN combination that performs best for the voice and video traffic while ensuring the aggregated network throughput. Taking this output as a basis, the second phase aims to compute the length of CW that makes it possible to maintain or increase the performance achieved during the first phase. Conversely, the third phase comprises the previously designed algorithms in a single scheme which is capable of selecting the appropriate model according to the network conditions.

The first phase of the EDCA predictive model is in charge of AIFSN adaptation. This operation is performed by using the aforementioned *J48* and *M5* algorithms in a deep training step considering a wide range of network scenarios. To this end, a group of 10 AIFSN configurations, shown in Table 1.3, has been selected by taking into consideration the performance and collision requirements discussed above. The training and validation steps consider a variable number of both legacy and QoS stations that deliver constant and intermittent traffic, and involves different application types and bitrates.

The second phase tunes the CW size to address the performance issues found by only steering the AIFSN combination. When the length of CW is increased, the legacy stations have higher priority for the channel access than some of the ones using EDCA. For this reason, making modifications to this parameter is of particular interest in scenarios having low or null DCF traffic. The values proposed for CW are shown in Table 1.4. On the one hand, the first possible combination, *CW 1*, increases the CW size by a power of two for each AC with respect to EDCA, with the exception of the CW_{max} limits for BE and BK traffic. On the other hand, the second group of values,*CW 2*, focuses on the voice and video traffic, and only the duration of CW_{min} of these ACs is increased, again by a power of two.

Table 1.5: CW-AIFSN configurations for the medium access parameters prediction scheme.

AC	Configuration 1	Configuration 2	Configuration 3	Configuration 4
AIFSN	Predicted	Predicted	Prev. combination	2nd Prev. combination
CW	Default	CW 1	CW 1	CW 2

Algorithm 1 Medium access parameters prediction scheme

Input:

be_util: channel utilization of the BE traffic.
bk_util: channel utilization of the BK traffic.
dcf_util: channel utilization of the DCF traffic.

procedure EDCA PARAMETER PREDICTOR($be\_util, bk\_util, dcf\_util$)
  if ($dcf\_util == 0$) and ($be\_util == 0$) and ($bk\_util == 0$) **then**
    return $M5 - Configuration\,2$
  else if ($dcf\_util == 0$) **then**
    return $J48 - Configuration\,3$
  else if ($dcf\_util 15\%$) **then**
    return $M5 - Configuration\,3$
  else
    return $J48 - Configuration\,1$

Some AIFSN-CW combinations may increase the priority of the stations using DCF with respect to those that use EDCA. To address these pitfalls, Table 1.5 presents four approaches that combine the increase in CW with the use of lower values for AIFSN than those predicted in the first phase. The first configuration, named *Configuration 1*, refers to the schemes designed in the first phase. Hence, CW maintains the values defined in EDCA. In the remaining configurations, this parameter is adapted according to the values shown in Table 1.4. In this regard, *Configuration 2* and *Configuration 3* consider the values defined in *CW 1*. By contrast, *Configuration 4* takes the values related to *CW 2*. Furthermore, the AIFSN combination can take different values. First, in *Configuration 1* and *Configuration 2* the prediction made in the first phase is used. Moreover, in *Configuration 3* we select the AIFSN combination immediately below the one predicted by the *J48* and *M5* models. Finally, *Configuration 4* selects the second AIFSN combination immediately below the one predicted in the first phase.

The third phase of the EDCA prediction scheme combines the results of the two previous phases. As can be observed in Table 1.5, eight predictive schemes are designed, four of them for each initial model (*M5* and *J48*). In this sense, an exhaustive analysis makes it possible to identify a group of traffic patterns to distinguish the models that achieve the highest performance based on network conditions. The structure of the dynamic prediction scheme is shown in Algorithm 1. In order to simplify the final model, the combinations achieving similar results have been omitted. In this way, the algorithm is able to dynamically adapt the EDCA parameters according to network conditions and, as a consequence, to enhance the performance of the delay sensitive services and the QoS level. Moreover, the scheme is standard compliant and guarantees compatibility with the stations without QoS support.

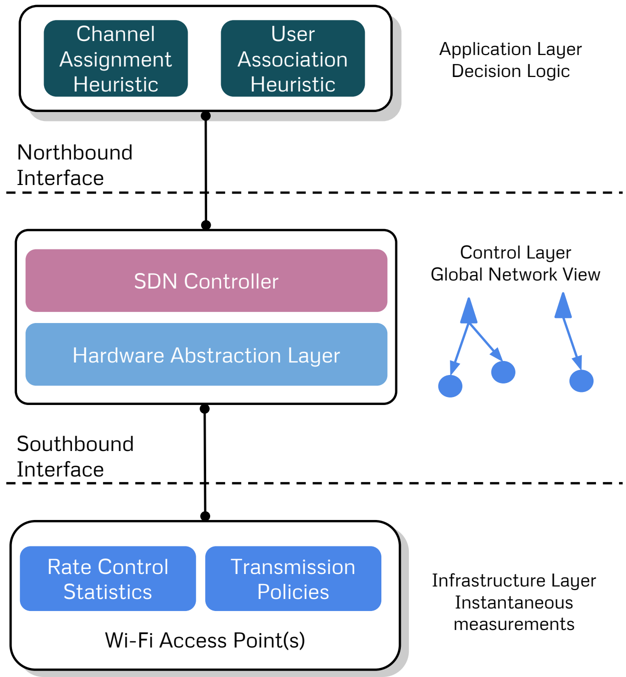


Figure 1.6: *Wi-Balance* system architecture.

1.4.2. Contributions to Channel Selection and User Association

Collisions usually lead to selecting lower data rates, an increase in the channel utilization and QoS impairment. Although the above approach aims for a considerable collision reduction, enhancements at the MAC layer are limited without modifying the network stack. This is particularly relevant in scenarios composed of several APs and involving mobile users, where having a global view of the network would enable more accurate management decisions. Moreover, the models presented above are evaluated via simulation, which may make some aspects of real environments pass unnoticed. With the aim of tackling with these problems, we take advantage of the flexibility provided by the Software-Defined WLANs to design a joint user association and channel assignment solution named *Wi-Balance*. As shown in Figure 1.6, we implement these algorithms as network applications. Furthermore, the APs are located in the infrastructure layer and just follow the operations from the SDN controller.

In scenarios composed of several APs, the first issue to be faced is the appropriate assignment of the wireless channels for the APs. A channel assignment procedure must consider the number of available channels and the APs sharing the same collision domain. To solve this problem, in this Thesis we propose a recursive constraint programming algorithm. This algorithm firstly assigns a channel to the APs with the lowest number of available channels. In this context, available channels have been considered to be those that are not assigned to other surrounding APs and do not overlap with the ones already assigned (co-channel interference). From among this group of APs, it first selects the ones with the highest number of neighbors already assigned. Furthermore, if all the channels have already been selected for the neighboring APs, the one that is less used is chosen. Moreover, if various channels match

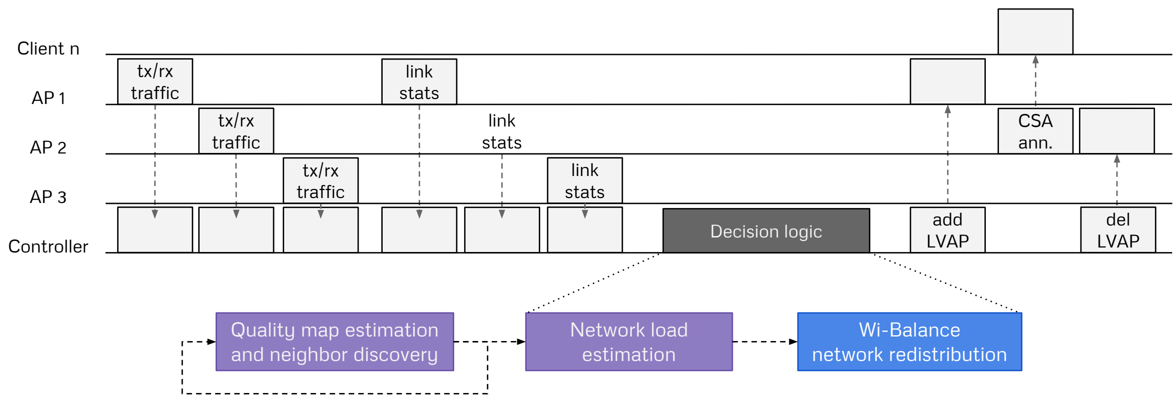


Figure 1.7: Overview of the working mode of *Wi-Balance*.

this restriction, the one less utilized is chosen. The algorithm converges when the distribution that minimizes the number of APs in the same collision domain is found. Nevertheless, this assignment is not static and the process is executed upon changes in the network.

Although it contributes to reducing network interference, a channel assignment algorithm is not sufficient and, in addition to this, an effective user association solution must be introduced. As stated in Section 1.1, in Wi-Fi networks users normally associate with the AP that provides the highest RSSI. However, considering only this factor may lead to situations in which some APs handle a high traffic load while others are idle. As a result, a user attached to the first group of APs may experience poor performance. For this reason, *Wi-Balance* follows the working mode illustrated in Figure 1.7 and, after channel assignment, it creates a channel quality map including the signal quality perceived from all the APs for each station. This quality map is built from the information that the SDN controller retrieves from the APs. Furthermore, *Wi-Balance* needs to estimate the traffic load of each AP and each channel. For this purpose, *Wi-Balance* gathers the link delivery statistics from the APs, and for each supported MCS and each user these provide the delivery probability and the expected throughput in the last observation window. This information is updated by the rate adaptation algorithm running on each AP. The algorithm also gathers the amount of data transmitted and received by all the users from the APs.

Based on the information described above, and considering U as the set of users, M as the set of APs and $\Omega(u) \subseteq M$ as the set of APs within the coverage area of the user $u \in U$, *Wi-Balance* computes the channel utilization $\mu(n)$ of each AP $n \in M$ and the average utilization across the network, $\overline{\mu}$. If a significant difference is found, a reassociation process is triggered for that specific AP. This process aims to balance the network load by migrating some of the users involved to other APs that are less heavily loaded. To this end, the algorithm, for each user u attached to the AP n, needs to compute the channel utilization of the APs in its coverage area, $\Omega(u)$, as well as the RSSI level R_u^m perceived by each AP $m \in \Omega(u)$. Then, for the handover *Wi-Balance* selects the user u with the lowest result for the product between the current channel utilization of the AP n, $\Omega(u)$, and the RSSI R_u^m. Finally, the average channel utilization $\mu(n)$ is recalculated to verify whether the process was efficient. Otherwise, the handover is reverted to ensure an effective resource allocation.

Handovers in WLANs usually require a substantial length of time for the reassociation process, which causes the connection to be temporarily interrupted. In this regard, the SDN approach allows it to transparently shift the client information between APs without performing a new association. This is possible through the *Light Virtual Access Point (LVAP)* abstraction, a per-client virtual AP that abstracts all the client state complexities such as association and authentication, and introduces seamless mobility support. In contrast to traditional Wi-Fi networks, each *LVAP* is identified with its own Basic Service Set Identifier (BSSID). Therefore, if an *LVAP* is migrated to another AP, the BSSID is preserved and, from the point of view of the user, it is still connected to the same physical AP.

The load balancing process may involve APs operating on different channels. In such cases, the CSA procedure must be used to inform the user about the channel change. In 802.11, the CSA data is transmitted via beacon frames. Nevertheless, the *LVAP* abstraction allows beacon frames to be transmitted in unicast mode to each client. In following this approach, when a handover is performed between different channels, the controller first creates an inactive *LVAP* for the client on the target AP and instructs the source AP to start a CSA procedure. After this process, the *LVAP* is removed from the source AP. In the meantime, the user will have switched the operating channel and will have found its *LVAP* on the target AP. In this way, users are not affected by the handover since the connection is never interrupted. This fact shows that *Wi-Balance* is capable of: (i) performing an efficient channel assignment for the APs to isolate possible collision domains; and (ii) conducting a transparent user reassociation process to balance the traffic load across the APs in the network.

1.4.3. Contributions to Multicast Transmissions

We have shown that SDN is able to address serious constraints in Wi-Fi networks. One of the most challenging issues is the delivery of streaming services in multicast mode and, in particular, the difficulty of adapting the transmission rate. To this end, the *Transmission Policy* abstraction is introduced, which enables the configuration of rate control policies on a layer 2 address basis for each AP. This abstraction allows the SDN controller to specify the MCS selected to transmit a frame and the retransmission policy used in case of multicast destinations. In other words, this latter attribute defines the policy from 802.11aa that can be used, i.e. legacy multicast, DMS or UR. This abstraction lays the foundation for implementing *SDN@Play*, an algorithm capable of intelligently steering the data rate for multicast services and improving transmission reliability. *SDN@Play* is introduced as an SDN application, as can be seen in the high-level reference system depicted in Figure 1.8.

SDN@Play makes use of the link delivery statistics obtained by the SDN controller from the rate control algorithm implemented at the AP to compute the MCS used for the multicast transmissions. For this reason, *SDN@Play* is presented as a two-phase scheme, as illustrated in Figure 1.9. During the first phase, the controller sets DMS as multicast policy for a certain multicast address with the aim of retrieving the rate control statistics of all the users in a multicast group. This information is used to calculate the MCS with the highest delivery probability for all the receivers. Then, the controller sets legacy multicast as the policy to be used during the second phase and instructs the AP to use the MCS calculated before instead

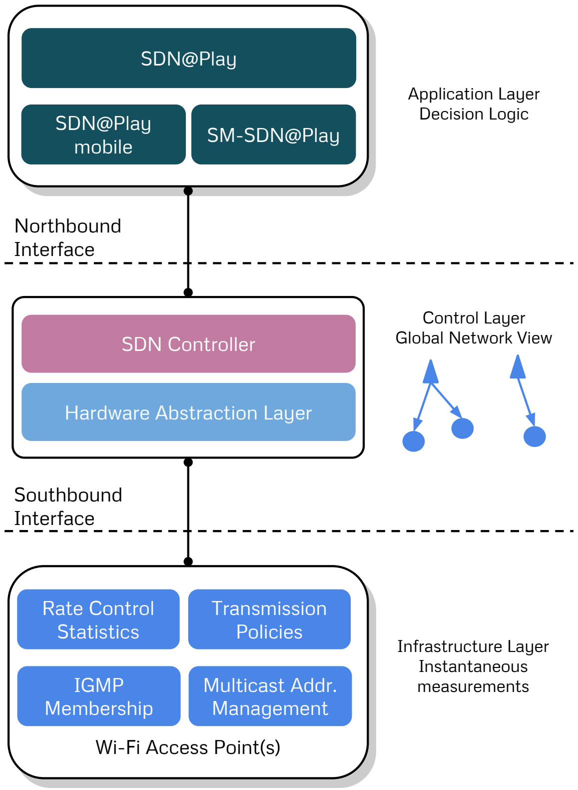


Figure 1.8: *SDN@Play* system architecture.

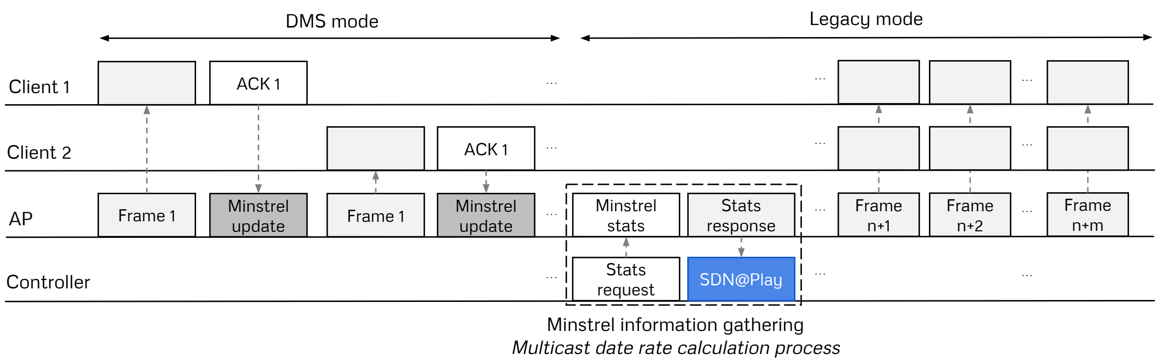


Figure 1.9: Overview of the working mode of *SDN@Play*.

of the basic rate specified by the IEEE 802.11 standard. This process is repeated until the end of the transmission with a configurable ratio between the DMS and the legacy periods.

Considering M to be the set of receivers in a multicast group, R the set of supported MCS, U the set of APs and $P_r^{n,n'}$ the delivery probability between the AP $u \in U$ and the receiver $n' \in M$, *SDN@Play* firstly calculates the valid MCS indexes, R_{valid}^n, for the receivers attached to an AP $n \in U$, as shown in Equation 1.1. The threshold r_{th} sets a relation between reliability and channel utilization. The use of higher values for this threshold leads to a reliability increase given that the delivery probability required for selecting a certain

MCS is higher. However, this results in an increase in the time that the channel is busy due to the use of less efficient data rates.

$$R^n_{valid} = \bigcap_{n' \in M} \left\{ r \in R | P_r^{n,n'} > r_{th} \right\} \quad \forall n \in U \tag{1.1}$$

Based on this information, Equation 1.2 presents the worst receiver approach used by *SDN@Play* to estimate the multicast transmission rate R^n_{tx}. The algorithm aims to ensure that the information is properly received by as many users as possible. For this reason, if none of the MCS indexes of a receiver has a delivery probability higher than the input threshold, for each of them the algorithm takes the MCS with the highest delivery probability and, from this set, it selects the lowest MCS as the new data rate.

$$R^n_{tx} = \begin{cases} max\left(R^n_{valid}\right) & if\ R^n_{val} \neq \emptyset \\ min\left(\bigcup_{n' \in M} \left\{ \underset{r \in R}{\text{argmax}}(P_r^{n'}) \right\} \right) & otherwise \end{cases} \quad \forall n \in U \tag{1.2}$$

SDN@Play is able to compute the data rate that delivers the expected QoS and performance in a multicast transmission. However, it does not consider the behavior of roaming users. Accordingly, *SDN@Play Mobile* builds on the capabilities of *SDN@Play* to account for mobile receivers and associate them to the AP that also reduces the network resource utilization. This algorithm is also introduced at the application layer, as shown in Figure 1.8, and makes use of the *LVAP* and *Transmission Policy* abstractions presented above.

With the purpose of improving client association in multicast environments, users must periodically report to their serving AP on the set of APs in their coverage area as well as the perceived signal quality (RSSI). This operation is performed via the *Beacon Reports* available in the IEEE 802.11k amendment [70]. Then, the information is forwarded to the SDN controller in order to build a channel quality map. *SDN@Play Mobile* periodically checks this quaility map with the aim of finding possible signal drops between a multicast receiver and its serving AP and checking whether another AP can deliver significant improvements in terms of channel quality for a certain user. The fulfillment of any of these conditions highlights the need to perform a handover process towards a more efficient network distribution. In view of this, considering $S(n)$ to be the set of receivers served by the AP $n \in U$, with $S(n) \subset M$, and $\rho^n_{n'}$ the channel quality between the AP $n \in U$ and the multicast receiver $n' \in M$, this process requires the computation of the average channel quality $\rho(n)$ and the standard deviation $\sigma(n)$ for all the APs, as shown in Equation 1.3 and Equation 1.4, respectively.

$$\rho(n) = \frac{\sum_{n' \in S(n)} \rho^n_{n'}}{|S(n)|} \tag{1.3}$$

$$\sigma(n) = \sqrt{\frac{1}{|S(n)|} \sum_{n' \in S(n)} (\rho^n_{n'} - \rho(n))^2} \tag{1.4}$$

There can be cases in which a certain AP is not serving any users. If this AP is not in the coverage area of the receiver n', the quantities above are simply undefined. Conversely, if this AP falls within the range of the receiver n', these quantities are set as $|S(n)| = 1$ and $\sigma(n) = 0$. Taking this information as a basis, *SDN@Play Mobile* defines the set $\Omega(n')$ of suitable APs for the user n' as presented in Equation 1.5 to ensure that the handover is not performed to an AP serving users under significantly worse channel conditions. Finally, the algorithm selects the candidate AP that would use the highest MCS to deliver the information to the receiver n' and performs the handover. After that, the data rate of the APs involved in the process is recomputed. Nevertheless, the handover must be network-wide efficient in terms of channel occupancy. Therefore, channel utilization is recalculated, and the handover is reverted in the case of higher resource consumption. This approach ensures the satisfaction of the end user and provides high QoS even in changing environments when the multimedia services are transmitted in multicast mode.

$$\Omega(n') = \{n \in U | \rho(n) - \sigma(n) \leq \rho^{n}_{n'}\} \tag{1.5}$$

SDN@Play Mobile optimizes the network radio resources and provides QoS support for mobile users in multicast communications. However, it does not consider the situations in which multiple multicast sessions are simultaneously held in a network. In fact, in this case, one instance of *SDN@Play* would be needed for each multicast group, thus greatly impacting network performance, especially due to the overlap of the DMS phase of each group. In this context, the *Scalable Multigroup SDN@Play* (*SM-SDN@Play*) algorithm is presented in the form of an SDN application, as depicted in Figure 1.8, and extends the capabilities of *SDN@Play* to effectively coordinate the actions of multiple multicast transmissions. With this purpose, the algorithm must recognize the formation of new multicast groups and handle the requests from the users to join or leave a certain multicast sesion. In this regard, the functionality of the IGMP protocol is extended to allow the SDN controller to maintain the information of each multicast group in each AP. This information comprises the receivers registered in the group and the *Transmission Policy* defined for that specific transmission.

To support several multicast transmissions in a scalable fashion, *SM-SDN@Play* divides the total duration L of the two phases defined in *SDN@Play* (legacy and DMS) into n small parts, whose length is determined by the duration of the DMS period, dms_d. On this basis, we define the length of each subphase as $d_i = \frac{L}{dms_d}$. Accordingly, the legacy period consists of $n-1$ consecutive phases, whose duration can be defined as $leg_d = (n-1) \cdot d_i$. This division is performed to prevent the unicast transmissions of all the multicast groups in the DMS mode from taking place at the same time. In this way, when a new multicast group is created, the algorithm schedules its DMS period in a different subphase i.

Figure 1.10 illustrates how *SDN@Play* is able to schedule multiple groups. In this example, 500 ms and 2500 ms have been set for the duration of the DMS and legacy phases, respectively. Specifically, it can be observed how after registering a multicast address request in *Step 1* and *Step 2*, the algorithm must find an available phase for the DMS period. In case of free slots (*Step 3.1*), this phase is simply scheduled. By contrast, if all the slots are occupied, the duration of each subphase, d_i, must be recalculated according to the minimum time required to obtain the link delivery statistics in DMS, dms_{min}, and the maximum duration dms_{max} of the DMS phase to avoid performance degradation (*Step 3.2*). As a consequence, the length of the DMS and legacy periods can be expressed as shown in Equation 1.6 and

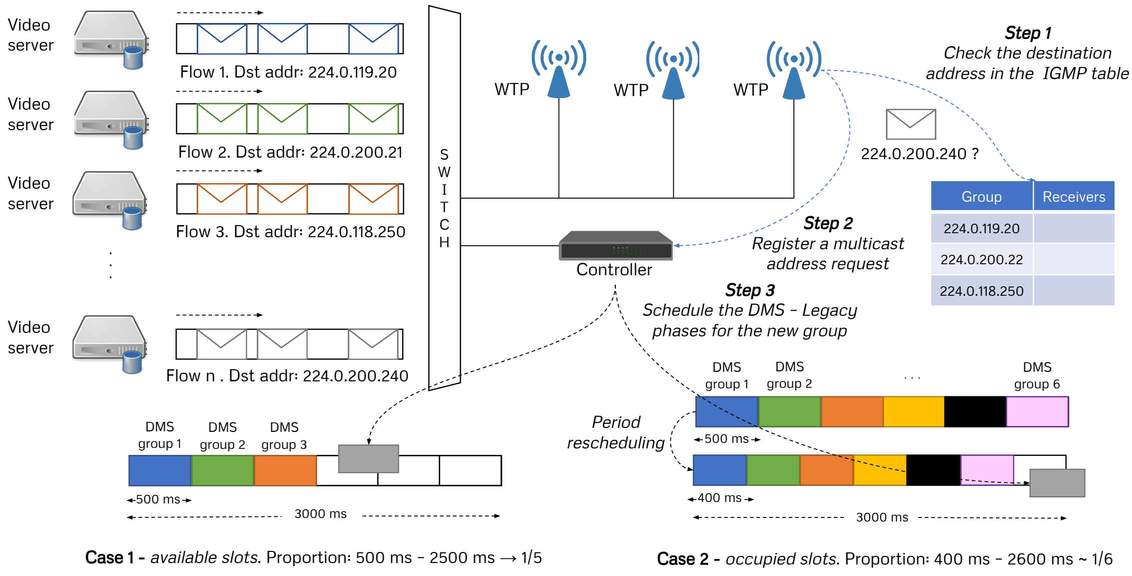


Figure 1.10: *SDN@Play* phases management and transmission policies coordination.

Equation 1.7, respectively. Furthermore, if the proportion between phase duration is not exact, the algorithm will approximate this quantity and maintain the phases ratio. After this process, the data rate for each multicast group in each AP is recomputed. However, only this information is reported to the APs, while the logic complexity lies in the SDN controller.

$$dms_d = max(dms_{min}, min(dms_{max}, \lceil \frac{L}{S} \rceil)) \tag{1.6}$$

$$leg_d = max(L - dms_d, dms_{min}) \tag{1.7}$$

The description presented above shows how *SDN@Play* introduces an integral multicast solution that is able to: (i) improve the reliability of the multicast transmissions provided by the IEEE 802.11 standard; (ii) dynamically adapt the data rate of the multicast applications to the network conditions; (iii) provide transparent mobility management in multicast scenarios; and (iv) schedule several multicast services transmitted at the same time.

In summary, the proposal of this Doctoral Thesis demonstrates the huge variety of factors and challenges that an integrated framework must consider in order to ensure QoS provisioning in multimedia content distribution over IEEE 802.11 networks. In particular, the main issues and a wide range of scenarios involving voice and video transmissions have been addressed from the point of view of the traditional network architecture at the MAC layer, as well as from the high-level vision provided by the network programmable capabilities of the SDN paradigm. In this regard, as will be further described in the next section, these proposals lead to an increase in user satisfaction and to an improvement in radio resource utilization. All the solutions presented are standard compliant which, together with their low complexity, make them an attractive proposition that can be applied directly to commercial devices that are available on the market and in the industry.

1.5. Results

After describing the main proposal of this Doctoral Thesis, in this section we report the results obtained from achieving the research goals presented in Section 1.3. Furthermore, we present the articles published in the course of this Ph.D. as a result of completing each goal.

This Thesis started, as described in **Goal 1**, by reviewing the state of the art in IEEE 802.11 networks. In this review, particular attention was paid to QoS capabilities, as well as to the unicast and multicast transmission modes in Wi-Fi networks. Moreover, the main aspects of the machine learning algorithms were also reviewed.

After completing the first goal, the tasks associated with **Goal 2** were undertaken. As a result, it was possible to assess the performance of the multimedia applications in 802.11. The evaluation conducted allowed us to simulate a wide range of scenarios and identify the main constraints determining user experience and the QoS level in the IEEE 802.11e amendment.

The work in the first steps of this Thesis provided the basis for the improvements pursued in **Goal 3**. This goal corresponded to addressing the pitfalls found in 802.11e and, specifically, in the EDCA function by means of machine learning techniques. Given the difficulty involved in this process, the final algorithm was designed in several phases. The first phase aimed at dynamically estimating the most appropriate values for the AIFSN parameter in EDCA according to the traffic conditions. This part of the scheme was composed of two predictive algorithms: a *J48* classifier tree and an *M5* regression model. The design of the *M5* model was presented in the paper entitled *"An AIFSN prediction scheme for multimedia wireless communications"*, published in the *"International Conference on Computer Communications and Networks (ICCCN 2015)"* [71], while the main aspects of the development of the *J48* classifier were discussed in the work *"Dynamic AIFSN tuning for improving the QoS over IEEE 802.11"*, published in the *"International Wireless Communications and Mobile Computing Conference (IWCMC 2015)"* [72]. From the results obtained in these works we observed that the standard EDCA parameters achieve the highest performance in both scenarios with a low traffic load and those with a high traffic load from stations without QoS support. However, regardless of the traffic load, the gradual separation between the AIFSN values for each AC led to a decrease in the amount of collisions and retransmissions, which contributed not only to improving the performance of the voice and video applications, but also the overall quality of the network. More specifically, the results showed an enhancement in the performance of the multimedia traffic by an average of 8% with regard to EDCA and a significant collision reduction of around 12%.

A comparison between the capabilities offered by these algorithms was presented in the chapter *"Ensuring QoS for IEEE 802.11 Real-Time Communications Using an AIFSN Prediction Scheme"*, published in the book entitled *"Security, Privacy and Reliability in Computer Communications and Networks"* [73]. Even though the nature of the predictive models was similar, their distinctive features made them especially suitable depending on the network conditions. This comparison demonstrated that the *M5* model ensured the highest performance for multimedia services in scenarios with a low traffic load from legacy stations, while the *J48* classifier performed better in the opposite situations. Furthermore, the *M5* model was able to outperform the global network performance with respect to the *J48* classifier.

The results obtained from the third goal showed that there was still room for improvement. In this regard, the second phase estimated the values assigned to the CW parameter in EDCA, and combined its capabilities with the functionality of the schemes of the first phase. The tests conducted in this part demonstrated that CW tuning in scenarios with a low traffic load from stations without QoS support contributed to outperforming the results achieved by the *J48* and *M5* initial models. Given the wide variety of schemes and results, the third phase unified them into a single dynamic algorithm capable of properly choosing the prediction scheme that selected the most suitable parameter configuration at every moment. The results at this point showed a boost in the voice and video throughput by an average of 20%, it being 30% in the absence of legacy stations. These results were described in the paper *"An Adaptive Medium Access Parameter Prediction Scheme for IEEE 802.11 Real-Time Applications"*, published in the journal *"Wireless Communications and Mobile Computing (2017)"* [1].

Given the difficulty of introducing new standard compliant solutions at the MAC layer, the tasks defined in **Goal 4** were aimed at exploring other innovative ways to address the QoS provisioning problem in Wi-Fi. To accomplish this goal, the principles of the Software Defined Networking paradigm were reviewed, as well as the existing solutions for implementing these features in radio access networks. In particular, the 802.11 data-path implementation and the communication between the SDN controller and the APs were analysed in depth.

Once we corroborated the adequacy of the SDN paradigm for the main objective of this Thesis, in **Goal 5** we leveraged the SDN hardware abstractions to investigate the effects of the network resource allocation on the delivery of multimedia contents. In view of this, a joint channel assignment and load balancing algorithm was designed to accomplish this goal. The proposal was assessed in multi-channel environments in a real-world scenario and compared with RSSI-based user association schemes. The results showed a reduction in channel utilization by up to 30% by means of a more efficient user distribution and a decrease in channel contention. Consequently, the throughput increased by up to 25% without penalizing network fairness. The results of this work were discussed in the paper *"Wi-Balance: Channel-Aware User Association in Software-Defined Wi-Fi Networks"*, published in the proceedings of the *"IEEE Network Operations and Management Symposium (NOMS 2018)"* [74]. The initial capabilities of the algorithm were also shown in a demonstration paper entitled *"Wi-Balance: SDN-based load-balancing in Enterprise WLANs"* in the conference *"IEEE Conference on Network Softwarization (Netsoft 2017)"* [75].

After achieving QoS enhancements in unicast transmissions through SDN, we studied whether we could obtain similar benefits in the services delivered in multicast mode in **Goal 6**. In this regard, we first evaluated the performance of the multicast policies included in the IEEE 802.11aa amendment in terms of QoE and QoS to identify the most appropriate conditions for applying each policy. The results of this work were discussed in the paper *"QoE evaluation of the IEEE 802.11aa Group Addressed Service for Robust Audio-Video Streaming"*, submitted to the journal *"Sensors"* [76], in which it is shown that even a very low number of transmission errors has a big negative impact on the quality perceived by the end-user. For this reason, a new hardware abstraction layer (*Transmission Policy* abstraction) was designed to enable multicast transmissions in SDN-based WLANs and to allow the SDN controller to configure the retransmission policy (defined in IEEE 802.11aa) specified for each multicast group at each AP. This abstraction became the basis for a data rate adaptation algorithm for multicast, initially presented in the work *“SDN@Play: A Multicast Rate Adaptation Mechanism for IEEE 802.11 WLANs”*, published in the proceedings of the *“IEEE Consumer Communications and Networking Conference (CCNC 2017)”* [77].

The rate selection algorithm is further improved in terms of multicast performance and coexistence with other unicast streams. These novel capabilities and a comparison with the standard multicast schemes were discussed in the paper entitled *“Programming Abstractions for Wireless Multicasting in Software-Defined Enterprise WLANs”*, published in the *“IEEE International Symposium on Integrated Network Management (IM 2017)”* [78]. The results showed how the algorithm was able to increase the data rate of the transmission and achieve a reduction in channel utilization of up to 80% without compromising the performance, and how it could easily scale regardless of the number of receivers. Moreover, a live demonstration of the functionality of the scheme was presented in the work *“Demo: SDN@Play as a strategy to enhance the multicast delivery rate in WLANs”*, published in the proceedings of the *“IEEE Consumer Communications and Networking Conference (CCNC 2017)”* [79].

As part of this goal we aimed to guarantee the experience of roaming users when consuming multimedia contents. In this regard, we extended the capabilities of the multicast rate selection algorithm to account for mobile users and manage the operations of various APs involved in a transmission. Moreover, this mechanism enabled seamless handover, which prevented the connection from being interrupted, thus increasing the QoS level. Based on the experimental measurements, we demonstrated that the algorithm performed better than both our previous proposal and the multicast standard schemes in terms of quality and channel utilization, also ensuring higher robustness and reliability levels. In fact, we observed that the delivery ratio for mobile receivers was improved to the extent that it practically reached the same performance as for stationary ones. As a result of completing these tasks, the work *“Joint Mobility Management and Multicast Rate Adaptation in Software-Defined Enterprise WLANs”* was accepted for publication in the journal *“IEEE Transactions on Network and Service Management (2018)”* [3].

In the course of the above evaluation we realized that the presence of several multicast sessions may saturate the network and affect the performance of the rate selection algorithm. For this reason, we extended the functionality of the IGMP protocol to make the SDN controller aware of changes in the multicast groups. To achieve **Goal 7**, the IGMP capabilities were taken as a basis to implement a new SDN-based algorithm capable of orchestrating multiple simultaneous multicast services and dynamically selecting the transmission rate of each multicast group. In this respect, a scalability (considering up to 20 users) and a multiple groups analysis were conducted in a real-world scenario, establishing a comparison with the multicast schemes defined in 802.11. The results showed that, due to the low channel utilization, the algorithm was able to improve the efficiency of the multicast video transmissions and simultaneously deliver a greater amount of information. As a matter of fact, for a considerable number of concurrent multicast transmissions, our algorithm achieves a normalized throughput of around $96 - 100\%$, while the two standard mechanisms evaluated offered a performance of around 40% and 60%, respectively. These results were presented in the paper entitled *“Efficient Real-time Content Distribution for Multiple Multicast Groups in SDN-based WLANs”*, accepted for publication in the journal *“IEEE Transactions on Network and Service Management (2017)”* [2].

Summary of Results

To conclude, once the tasks defined in Section 1.3 had been performed, we were able to achieve the goals presented in Section 1.2. As a result, the following list of articles have been published or accepted for publication:

- **Goal 3.** *To improve channel access in IEEE 802.11e.*
  - *"An AIFSN prediction scheme for multimedia wireless communications"*, published in *"Proceedings of the International Conference on Computer Communications and Networks 2015"* [71]. The AIFSN prediction scheme based on the use of an *M5* regression model is introduced.
  - *"Dynamic AIFSN tuning for improving the QoS over IEEE 802.11"*, published in *"Proceedings of the International Wireless Communications and Mobile Computing Conference 2015"* [72]. The feasibility of the *J48* classifier for the AIFSN estimation problem is assessed.
  - *"Ensuring QoS for IEEE 802.11 Real-Time Communications Using an AIFSN Prediction Scheme"*, published in the book *"Security, Privacy and Reliability in Computer Communications and Networks (2016)"* [73]. A comparison between the previous AIFSN prediction schemes is performed.
  - *"An Adaptive Medium Access Parameter Prediction Scheme for IEEE 802.11 Real-Time Applications"*, published in the journal *"Wireless Communications and Mobile Computing (2017)"* [1]. The complete EDCA tuning scheme based on the network conditions is defined and evaluated via simulation. This paper is included in the Thesis on page 31.
- **Goal 5.** *To enable efficient channel selection and user association in WLANs.*
  - *"Wi-Balance: Channel-Aware User Association in Software-Defined Wi-Fi Networks"*, published in *"Proceedings of the IEEE Network Operations and Management Symposium 2018"* [74]. A joint SDN-based channel assignment and load balancing algorithm is implemented and evaluated on real devices. This paper is included in the Thesis on page 53.
  - *"Wi-Balance: SDN-based load-balancing in Enterprise WLANs"*, published in *"Proceedings of the IEEE Conference on Network Softwarization 2017"* [75]. The capabilities of the channel assignment and load balancing algorithm are shown in a live demonstration involving current commercial devices.
- **Goal 6.** *To enable robust multimedia transmissions and seamless user mobility in multicast environments.*
  - *"QoE evaluation of the IEEE 802.11aa Group Addressed Service for Robust Audio-Video Streaming"*, submitted to the journal *"Sensors (2018)"* [76]. It presents a review of the multicast mechanisms introduced in IEEE 802.11aa and a quantitative performance evaluation in terms of QoS and end-user QoE.

- *"SDN@Play: A Multicast Rate Adaptation Mechanism for IEEE 802.11 WLANs"*, published in *"Proceedings of the IEEE Consumer Communications and Networking Conference 2017"* [77]. It presents an SDN abstraction for dynamically selecting the multicast policy used for each multicast group and a draft design for a rate selection algorithm in multicast scenarios.
- *"Demo: SDN@Play as a strategy to enhance the multicast delivery rate in WLANs"*, published in *"Proceedings of the IEEE Consumer Communications and Networking Conference 2017"* [79]. The functionality of the data rate adaptation algorithm for multicast applications is initially shown on a real-world testbed.
- *"Programming Abstractions for Wireless Multicasting in Software-Defined Enterprise WLANs"*, published in *"Proceedings of the IEEE International Symposium on Integrated Network Management 2017"* [78]. The complete multicast rate selection algorithm that ensures coexistence with other unicast streams is implemented and evaluated in real environments. This paper is included in the Thesis on page 65.
- *"Joint Mobility Management and Multicast Rate Adaptation in Software-Defined Enterprise WLANs"*, accepted for publication in the journal *"IEEE Transactions on Network and Service Management (2018)"* [3] . The capabilities of the transmission rate adaptation algorithm are extended to ensure seamless mobility and high performance for roaming users. This paper is included in the Thesis on page 77.

- **Goal 7.** *To orchestrate multiple simultaneous multicast services.*
  - *"Efficient Real-time Content Distribution for Multiple Multicast Groups in SDN-based WLANs"*, accepted for publication in the journal *"IEEE Transactions on Network and Service Management (2017)"* [2]. The designed algorithm is capable of scheduling several multimedia multicast services that are simultaneously transmitted in real scenarios. This paper is included in the Thesis on page 93.

1.6. Summary of Publications

In the preceding sections we have presented the research goals that have motivated this Doctoral Thesis. Furthermore, we have described the main proposal of this dissertation and the outcomes obtained from different works. In this regard, the articles published in the course of this Ph.D. are shown below, and classified by publication type, including those that are not directly related to the main objective of this Thesis but arose as a result of other works and collaborations.

- **JCR Journals**
  - "An Adaptive Medium Access Parameter Prediction Scheme for IEEE 802.11 Real-Time Applications". *Wireless Communications and Mobile Computing.* 2017 [1]. (IF:1.899, Q2. Published).
  - "Efficient Real-time Content Distribution for Multiple Multicast Groups in SDN-based WLANs". *IEEE Transactions on Network and Service Management.* 2017 [2]. (IF:3.134, Q1. Accepted).

- "Joint Mobility Management and Multicast Rate Adaptation in Software-Defined Enterprise WLANs". *IEEE Transactions on Network and Service Management.* 2018 [3]. (IF:3.134,Q1. Accepted).
- "QoE evaluation of the IEEE 802.11aa Group Addressed Service for Robust Audio-Video Streaming". *Sensors.* 2018 [76]. (IF:2.677, Q1. Under review).

■ **International Conferences**

- "An AIFSN prediction scheme for multimedia wireless communications". *Proc. of IEEE ICCCN 2015* [71]. (CORE A. Conference Rating class 2).
- "Dynamic AIFSN tuning for improving the QoS over IEEE 802.11". *Proc. of IEEE IWCMC 2015* [72]. (CORE B. Conference Rating class 3).
- "SDN@Play: A Multicast Rate Adaptation Mechanism for IEEE 802.11 WLANs". *Proc. of IEEE CCNC 2017* [77]. (CORE B. Conference Rating class 3).
- "Programming Abstractions for Wireless Multicasting in Software-Defined Enterprise WLANs". *Proc. of IEEE IM 2017* [78]. (CORE A. Conference Rating class 3).
- "Wi-Balance: Channel-Aware User Association in Software-Defined Wi-Fi Networks". *Proc. of IEEE NOMS 2018* [74]. (CORE B. Conference Rating class 3).
- "Lasagna: Programming Abstractions for End-to-End Slicing in Software-Defined WLANs". *Proc. of IEEE WoWMoM 2018* [80]. (Core A. Conference Rating class 3. Under review).

■ **Demonstrations and Posters in International Conferences**

- "Demo: SDN@Play as a strategy to enhance the multicast delivery rate in WLANs". *Proc. of IEEE CCNC 2017* [79]. (CORE B. Conference Rating class 3).
- "Wi-Balance: SDN-based load-balancing in Enterprise WLANs". *Proc. of IEEE Netsoft 2017* [75].
- "Wi-Not: Exploiting Radio Diversity in Software-Defined 802.11-based WLANs". *Proc. of IEEE NOMS 2018* [81]. (CORE B. Conference Rating class 3).

■ **Book Chapters**

- "Ensuring QoS for IEEE 802.11 Real-Time Communications Using an AIFSN Prediction Scheme". *Security, Privacy and Reliability in Computer Communications and Networks.* 2016 [73].

■ **National Conferences**

- "Esquema de predicción dinámica de AIFSN para mejorar la QoS en redes IEEE 802.11". *Jornadas de Paralelismo 2015* [82].
- "Un nuevo enfoque para la adaptación de caudal en transmisiones multicast en SDWN". *Jornadas de Paralelismo 2016* [83].
- "Gestión de la movilidad de usuarios basada en SDN para aplicaciones multicast en WLANs". *Jornadas de Paralelismo 2017* [84].

CHAPTER 2

Publications

2.1. An Adaptive Medium Access Parameter Prediction Scheme for IEEE 802.11 Real-Time Applications

- **Title:** An Adaptive Medium Access Parameter Prediction Scheme for IEEE 802.11 Real-Time Applications
- **Authors:** Estefanía Coronado, José Villalón, and Antonio Garrido
- **Type:** Journal
- **Journal:** Wireless Communications and Mobile Computing
- **Publisher:** Hindawi
- **ISSN:** 1530-8677
- **State:** Published
- **Year:** 2017
- **DOI:** 10.1155/2017/5719594
- **URL:** https://www.hindawi.com/journals/wcmc/2017/5719594/
- **Category:** Computer Science, Engineering, Telecommunications
- **Impact Factor:** 1.899
- **JCR Ranking:** Q2

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Wireless Communications and Mobile Computing
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Research Article

An Adaptive Medium Access Parameter Prediction Scheme for IEEE 802.11 Real-Time Applications

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Multimedia communications have experienced an unprecedented growth due mainly to the increase in the content quality and the emergence of smart devices. The demand for these contents is tending towards wireless technologies. However, these transmissions are quite sensitive to network delays. Therefore, ensuring an optimum QoS level becomes of great importance. The IEEE 802.11e amendment was released to address the lack of QoS capabilities in the original IEEE 802.11 standard. Accordingly, the Enhanced Distributed Channel Access (EDCA) function was introduced, allowing it to differentiate traffic streams through a group of Medium Access Control (MAC) parameters. Although EDCA recommends a default configuration for these parameters, it has been proved that it is not optimum in many scenarios. In this work a dynamic prediction scheme for these parameters is presented. This approach ensures an appropriate traffic differentiation while maintaining compatibility with the stations without QoS support. As the APs are the only devices that use this algorithm, no changes are required to current network cards. The results show improvements in both voice and video transmissions, as well as in the QoS level of the network that the proposal achieves with regard to EDCA.

1. Introduction

Wireless technologies have experienced a marked increase in popularity over the past few years. As a result, the trend towards the use of wireless networks has been noticeable, and nowadays it is possible to find them in many different scenarios such as hospitals, airports, and universities. Due to their simplicity of deployment, low cost, and multimedia content support, IEEE 802.11 [1] networks have become essential and have reached a leading position in the market.

Access mode to the Internet and consumption patterns are also changing, especially those related to multimedia applications. In fact, the development of new video coding standards has led to an enhancement in the quality of the contents and a growth in the number of High Definition (HD) video streaming services. However, this improvement results in an increase in the volume of data that must be transmitted over the network. This new trend comes from the emergence of intelligent devices such as smartphones, tablets, and new generation game consoles. All these devices share a common feature, namely, being equipped with IEEE 802.11 interfaces.

Real-time applications have high Quality of Service (QoS) requirements, which are not provided by the original IEEE 802.11 standard. This drawback leads to developing the IEEE 802.11e amendment [2], where the QoS level is improved by introducing EDCA (Enhanced Distributed Channel Access) as a medium access function. EDCA is able to classify and prioritize the traffic by defining a group of medium access parameters. Nevertheless, some research proves that there are still some limitations in the QoS field that must be overcome, particularly with respect to voice and video transmissions.

In addition to stations with QoS support, IEEE 802.11e networks can be also composed of legacy stations. Legacy stations do not offer QoS capabilities and cannot make modifications to the medium access parameters. Accordingly, and with the aim of maintaining the interoperability between both types of stations, EDCA recommends the use of a group of values for these parameters. In spite of improving the performance of real-time applications, these values are not optimum for voice and video communications in a huge number of scenarios. For that reason, making an appropriate adjustment to these parameters becomes a key issue.

Artificial Intelligence (AI) techniques are developed to solve complex problems that usually require human reasoning. In this context, the use of such methods may be useful given the diverse conditions that can be found in a wireless network. In particular, the number of collisions of the network is one of the factors that determines the network status in a more significant way. Therefore, the application of AI techniques could make it possible to find traffic patterns and contribute to enhancing the QoS level and the performance of the network.

In this paper, we introduce a dynamic prediction scheme for the medium access parameters in EDCA to improve the QoS level over IEEE 802.11 WLANs. The suitable selection of the waiting time periods for every Access Category (AC) to access the channel leads to a reduction in the collisions in the network, mainly between voice and video frames. This strategy achieves, therefore, a double objective: on the one hand, the quality of the multimedia communications is considerably enhanced; on the other hand, it allows for a channel usage optimization. The major contribution of this paper is the capacity to adapt the EDCA configuration dynamically based on the traffic conditions by using a group of simple rules at the Access Point (AP). Thus, it only requires a few small modifications to the AP firmware, maintaining full compatibility with current commercial network cards.

The remainder of this paper is organized as follows. Section 2 reviews the IEEE 802.11e amendment and some proposals that aim to enhance the QoS level. Section 3 gives a summary of the Artificial Intelligence techniques used in this research. In Section 4 we introduce the proposed scheme and the most relevant design aspects, while in Section 5 the details of the implementation process are presented. Section 6 describes the final dynamic predictive model. As this proposal has been designed as an incremental process, the results of the performance evaluation and a comparison with EDCA are also discussed in this section. Finally, Section 7 provides some concluding remarks on our work.

2. QoS in IEEE 802.11 Networks

The original IEEE 802.11 standard introduces two functions to access the wireless medium, which cannot provide QoS capabilities. These functions are called Distributed Coordination Function (DCF) and Point Coordination Function (PCF). For this reason, the IEEE 802.11e amendment was developed.

2.1. IEEE 802.11e. The IEEE 802.11e amendment was released with the aim of providing QoS support to voice and video applications over IEEE 802.11 WLANs [2]. Actually, the main feature of this amendment is the capacity to differentiate traffic flows and services. As backward compatibility must be kept, a distinction is drawn between the stations that support QoS (QSTAs) and the stations that use DCF and do not offer such support (nQSTAs). For this purpose, the 802.11e amendment implements the Hybrid Coordination Function (HCF).

Table 1: Default EDCA parameter set for IEEE 802.11g PHY layer.

AC	CW_{min}	CW_{max}	AIFSN	TXOP
AC_BK	aCW_{min}	aCW_{max}	7	—
AC_BE	aCW_{min}	aCW_{max}	3	—
AC_VI	$(aCW_{min}+1)/2-1$	aCW_{min}	2	3.008 ms
AC_VO	$(aCW_{min}+1)/4-1$	$(aCW_{min}+1)/2-1$	2	1.504 ms

This function is composed of two channel access methods: HCF Controlled Channel Access (HCCA) and EDCA. As was the case with PCF, the first of them follows a centralized scheme to access the medium, while the second one works in a distributed way, as DCF does. To this end, the HCF implementation is mandatory for all the QSTAs. Nevertheless, only EDCA is supported by commercial network cards on current devices as a method for accessing the wireless medium.

EDCA improves the capabilities of DCF and distinguishes between eight User Priorities (UPs). Moreover, four ACs are defined, which are derived from the UPs and are in charge of classifying the traffic streams. In this way, in order to from highest to lowest priority, Voice (VO), Video (VI), Best Effort (BE), and Background (BK) Access Categories are considered, as sketched in Figure 1. Each AC works on its own transmission queue and is characterized by an EDCA parameter set. The EDCA parameter set specifies a priority level through an Arbitration Interframe Spacing Number (AIFSN) combination, a Transmission Opportunity interval (TXOP), and the duration of the Contention Window (CW). In order to provide a fair transmission for the DCF stations, the IEEE 802.11e amendment defines a standard combination of the medium access parameters, as shown in Table 1.

The Arbitration Interframe Spacing (AIFS) period determines the amount of time that a station must wait before beginning a new transmission. This is derived from the AIFS Number (AIFSN) value for each AC, as can be seen in (1) where the SlotTime denotes the duration of a slot according to the physical layer, and the Short Interframe Space (SIFS) refers to the amount of time used by high priority actions that require an immediate response.

$$\text{AIFS}[\text{AC}] = \text{AIFSN}[\text{AC}] \cdot \text{SlotTime} + \text{SIFS}. \quad (1)$$

The CW size sets the length of idle time; after that a transmission of a given station may occur. The CW values are assigned in the inverse order to that of the priority of the corresponding AC. Whenever an unsuccessful transmission takes place, the CW follows an increment sequence in powers of two minus one. In this way, the size of CW could be increased until it reaches at most the value of CW_{max}. It must keep this value until a frame is successfully transmitted, CW being reset to CW_{min}. This algorithm is not exactly the same as that used by DCF stations [3]. In this case, whenever the medium is sensed busy after an AIFSN period, the counter previously mentioned is decreased by one time slot. Due to this change, the use of an AIFSN value of 3 time units by the BE traffic provides a similar priority to that offered to the DCF traffic. Likewise, TXOPs allows the transmission of multiple streams without gaining the medium access every time that a

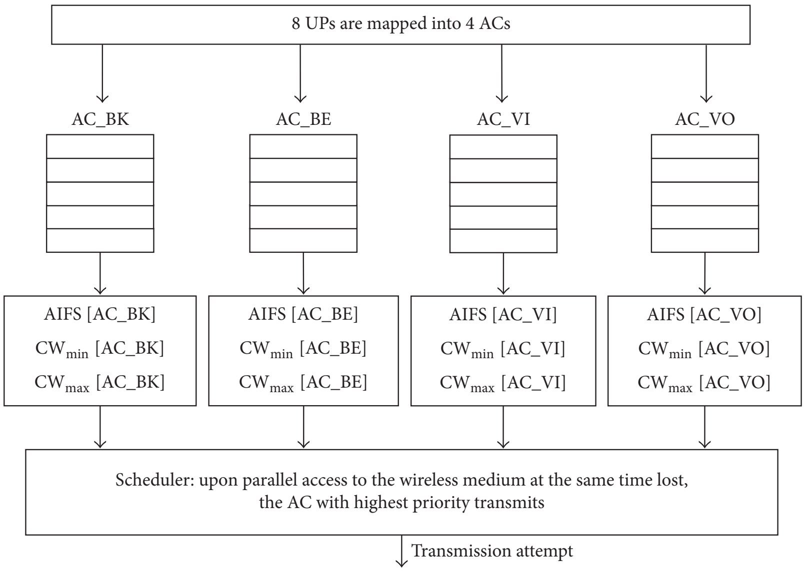


Figure 1: EDCA Access Categories mapping.

frame is transmitted. Therefore, they are usually used in real-time applications.

The AP of the network sends the EDCA parameter set through beacon frames to the stations of a Basic Service Set (BSS). The IEEE 802.11e amendment allows the APs to modify the values for this parameter set. However, no mechanism is considered in this amendment for carrying out this task and most commercial devices do not implement it.

2.2. Dynamic Adaptation in IEEE 802.11e. Wireless networks are conditioned by a huge set of factors that can change over time. For instance, the QoS level provided for all the services becomes worse as the congestion of the network and the number of collisions increase. In this regard, the usage of optimum values for the MAC parameters gains importance when trying to access the wireless medium in a more efficient way. In view of the above, several proposals have appeared with the aim of overcoming the QoS limitations that exist in IEEE 802.11 networks. They could be classified into three different categories: solutions designed at the link layer related to the medium access parameters, schemes designed at other layers such as the application level, and mathematical theories and formal models.

Recently, adapting the medium access parameters has taken increased importance. Depending on the selected values, the channel access can be more efficient, this being reflected in a significant reduction in the collisions and an increase in the QoS level. This aspect is of particular interest in the case of multimedia traffic transmitted in real-time due to their inherent temporal restrictions. This aspect has been studied in [4], in which it is shown the notable effect that using adequate priority parameters has on the network performance. After an analysis of these parameters using different priorities, it is shown that only the CW and the AIFS are associated with improvements, while the TXOP has no effect in the studied scenarios.

In this context, other authors have also examined this issue from a different point of view. In [5] a set of scenarios is presented in which several values for the AIFSN and CW are taken into account. Through this analysis, it is proved that the appropriate selection of the AIFSN combination contributes to reducing the collisions among the traffic of different ACs. Furthermore, the adequate tuning of the CW also leads to an enhancement of the QoS level. This conclusion is also reached in [6], where it is pointed out that a decrease in both the collisions and access media delay leads to an improvement in the efficiency of the network.

In [7] an adaptation scheme for the duration of CW is introduced, achieving better results than EDCA. However, compatibility with legacy DCF stations is not considered. To address this issue, another proposal presents a new way of offering backward compatibility with the DCF stations [8]. This algorithm prioritizes the voice and video traffic streams over the others. As the priority of the DCF stations cannot be modified by updating the EDCA parameter set, the CW size is increased by retransmitting packets that are properly received by the DCF stations. In this way, the priority of the stations that use EDCA decreases. Nevertheless, unnecessary traffic is introduced into the network. The tuning of CW is also taken into account in [9] for collision avoidance, outperforming the results of EDCA in terms of medium access delay.

An approach with this same goal is presented in [10], where three possible load levels are considered. This proposal achieves a reduction in the number of retransmissions and an enhancement in network performance. However, there is a drop in the voice and video traffic transmitted, which impairs its temporal restrictions. With the same aim, the approach proposed in [11] is based on the reservation and scheduling of multiple TXOPs, reducing in this way the number of collisions in the network. Nevertheless, using several consecutive TXOPs may penalize the remaining real-time applications, since they cannot access the channel for a certain amount of time.

Some other authors model their own medium access functions on the sidelines of EDCA or DCF. However, this type of models is usually incompatible with the operation mode of IEEE 802.11. In [12] a new function that combines the features of both DCF and EDCA functions is proposed. Indeed, it establishes a protocol that is able to determine the next transmitter according to a probability that depends on the priority of the traffic. Nevertheless, this approach cannot maintain interoperability with the current network cards.

Other approaches focus on ensuring an optimum level of QoS by carrying out any kind of optimization over other network layers. In [13] a middleware at the application level to improve the multimedia traffic performance is introduced. In particular, the main goal of this scheme is to maximize the provisioning quality to wireless clients in case of anomaly conditions in the network. Another scheme at this same level is proposed in [14]. iPAS is presented as a new way to adapt the priority of multimedia frames to establish an adequate bandwidth according to a group of QoS-related parameters. The architecture of this approach is composed of two main blocks: the iPAS server and the iPAS client. The iPAS client collects information about stream preferences, which is sent as feedback to the iPAS server. Then, based on these data, the iPAS server is responsible for managing bandwidth resources by means of a stereotype-based resource allocation mechanism and a bandwidth estimation scheme.

The design of an analytical model to improve network performance has also been considered. Nevertheless, most of these models make assumptions that may not be fulfilled in real transmissions. In [15] a model using Markov chains is defined. However, the same bit rate is considered for all the stations. By contrast, this issue is addressed by means of a bandwidth control scheme in [16]. In a similar way, the mathematical model presented in [17] is only tested under network saturation conditions. Finally, authors in [18] evaluate the model under an ideal channel scenario.

Although they are less common, besides Markovian models, p-persistent models have also been used. In [19] a time-domain analysis has been carried out in order to check the CSMA/CA performance and model the EDCA behavior from a different point of view. In spite of being properly validated via simulations, it is assumed that no transmission errors occur in the channel during the transmissions and all the stations always have a packet to send.

Although most of the described proposals outperform the results achieved by EDCA, it is not possible for all of them to maintain interoperability with legacy DCF stations. Actually, in many cases, these models experience compatibility problems with the IEEE 802.11 standard or introduce additional control traffic overhead in the network. In particular, despite this subject having been widely discussed, there are no other works that make use of Artificial Intelligence techniques to adapt the medium access parameters of EDCA.

3. Supervised Learning

Supervised learning refers to the task of defining a model, $h_\Theta(x)$, from supervised training data. The information relative to objects in this data is represented by a set of n input features, $X = (X_1, \ldots, X_n)$, and an output variable, Y. In supervised learning, data is defined as a pair, (X, Y), whose current outputs are already known (that is why it is called *supervised*). The process is in charge of analyzing the training data and using them to induce a model able to predict other unlabelled data (y) when only the values of their input features (x) are known.

The learning process may be of a different nature. In classification problems, the goal is to determine the class of an instance which is unknown ($Y \in \{c_1, \ldots, c_K\}$). These models can be represented in several ways such as decision trees or classification rules. Regression analysis refers to a statistical methodology used in cases of numeric prediction. It is expected to identify distribution patterns in the current data, obtaining $Y \in \mathbb{R}$, and therefore $h_\Theta(x) \in \mathbb{R}$.

In this work, both types of supervised learning described above are utilized. X represents the parameter configuration used for managing the voice and video traffic in a network, whereas the output Y represents the throughput achieved by this setting. More specifically, in the case of the regression model, $h_\Theta(x) \in \mathbb{R}$ returns the predicted throughput of the network, y, given the parameter configuration x. By contrast, using the same configuration x, the classification model provides a label Y for such an instance.

There are a large number of supervised learning models for regression, such as Linear Regression [20], Neural Networks [21], Support Vector Machines [22], or Regression Trees [23]. Selecting a certain model depends mainly on the aim of the application. Some of them are more powerful than others; in other words, they achieve greater precision and can detect more relevant patterns in data. However, the ease with which they can be interpreted can be an issue in some scenarios. Models such as Neural Networks are considered *Black Box* models due to the fact that the information related to underlying patterns in data cannot be extracted from them. In contrast, Regression Trees can be easily interpreted and provide useful information regarding the relation between input and output features. Another important issue concerns computational complexity. For instance, obtaining y from x with a Neural Network implies some matrix multiplications and can be too slow in some settings. However, processing a regression tree might only require a few comparisons. This becomes particularly important when using these models in real-time applications.

Due to the application domain of this work, it is important to select models that can quickly obtain the required result. The models must be used in real-time to determine the parameter setting that achieves a higher throughput. It is also important that the obtained models can be interpreted, analyzed, and even modified after having been learned. In the context of this work, a *J48* classifier tree and a *M5* regression model have been selected. They are further described in the sections below.

3.1. J48 Classifier Tree. The *J48* classifier tree is based on the *C4.5* algorithm, which is the successor to the ID3 algorithm [24]. This tree can be found in the *weka* package for machine learning [25]. This model aims to design a decision tree that is as short as possible. The algorithm follows a recursive procedure by means of a heuristic greedy search to obtain the final model. In this way, it selects every attribute according to its gain ratio (see (2)). This guideline expressly refers to the information gain obtained as a result of the classification made and the entropy of the predictive variable, X_i.

$$\text{Gain ratio} = \frac{I(C, X_i)}{H(X_i)} = \frac{H(C) - H(C/X_i)}{H(X_i)}. \quad (2)$$

The information gain is given by the expression $I(C, X_i)$, which obtains the mutual information between X_i and C; that is, the algorithm evaluates the potential uncertainty when classifying an attribute X_i on a set C. This is calculated as the difference between the entropy of the different outputs of the set C, $H(C)$, and the entropy obtained after using a certain attribute X_i, $H(C, X_i)$. In order to prevent the variables with a wider range of possible values from being the biggest beneficiaries in the classification, the information gain is weighted with the entropy of the predictive variable, $H(X_i)$.

The algorithm divides the training set into several subsets that are as pure as possible until a leaf node is reached. In this respect, the following internal node to be selected is the attribute which maximizes the aforementioned gain ratio. Once the tree has been modeled, this algorithm also incorporates a pruning technique to reduce its size and complexity. In the context of this work, an example of a subtree of the J48 classifier designed can be observed in Figure 2. In this way, it is possible to calculate the label for a certain combination which is situated at the leaf nodes. This label is obtained by using the rest of the parameters as an entry point for the tree.

3.2. M5Rules. The *M5* algorithm [23] represents $h_\Theta(x) \in \mathbb{R}$ as a regression tree and is very similar to its counterpart, *c4.5* [24], which is used for classification problems. This tree represents a division of the input space and each node defines a condition over some input attribute X_i. For instance, a node defined by the condition `[VI_channel_occupancy <= 0.341]` represents the branch which would be used to process all objects whose value for variable `VI_channel_occupancy` is smaller than 0.341. Meanwhile, the other branch would be used to process the remaining cases. Each leaf represents an input subspace and corresponds to the cases which fit the conditions represented by the path from the root of the tree to the leaf.

In *M5*, there are two possibilities to obtain the output values for the cases falling into a leaf of the tree. The first one, namely, *regression tree*, uses the mean output value of the training data falling into that leaf as default prediction. The second one, namely, *model tree*, learns a multivariate Linear Regression equation from the training data corresponding to the leaf and uses it to predict the output values.

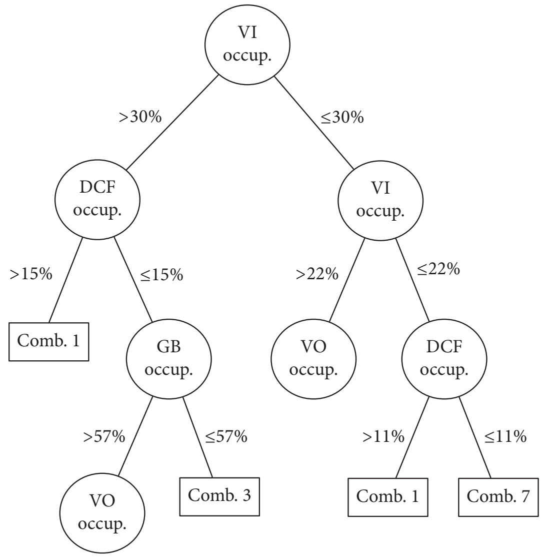


Figure 2: Example of subtree obtained by the *J48* classifier tree.

```
Rule: 1
IF
    VI_channel_occupancy <= 0.341
THEN
max_th[0] =
    −0.0449 ∗ global_channel_occupancy
    + 0.0701 ∗ DCF_channel_occupancy
    + 0.1152 ∗ BE_channel_occupancy
    − 0.0392 ∗ VI_channel_occupancy
    − 0.279 ∗ VO_channel_occupancy
    + 2.0059 [151/2.844\%]
```

Algorithm 1: Example of rule induced by *M5Rules*.

The algorithm *M5Rules* is also included in the *weka* package. It first learns a regression (or model) tree from training data by means of the implementation of *M5* included in this package, namely, *M5P*, and then extracts a set of rules. Algorithm 1 shows an example of the rules obtained. So that the value of all objects for variable `[VI_channel_occupancy <= 0.341]`; the output value is obtained as a linear expression from the remaining (5) variables. The information `[151/2.844\%]` indicates that 151 objects of the training dataset fall into that leaf and that the relative error obtained for those objects is 2.844%.

4. MAC Parameter Tuning Design

In recent times, the number of applications with high QoS level requirements has increased as a result of the improvement in the quality of the multimedia contents. Consequently, traffic patterns in wireless networks have also changed significantly.

Voice and video traffic transmitted in real-time is especially sensitive to latency and packet losses. Therefore, it must be given an even higher priority. The IEEE 802.11e amendment and the introduction of EDCA allow the definition of medium access parameters for each traffic type. However, due to the complexity involved in determining the traffic conditions at every moment, this amendment uses a default combination for these parameters instead of providing a mechanism for their adaptation. Moreover, the aforementioned parameters cannot be tuned in the case of legacy DCF stations. For this reason, compatibility with these stations becomes an important issue.

This paper presents a dynamic tuning scheme for the medium access parameters in EDCA to enhance QoS differentiation, targeted for real-time applications. The priority assigned to each AC is variable and depends on the traffic conditions. These priority levels are calculated by employing Artificial Intelligence techniques which consider both traffic flow patterns and network status. Furthermore, the scheme is fully compatible with EDCA, not being necessary to make changes to current network cards. Moreover, it seeks to ensure backward interoperability between the stations that use EDCA and those that use DCF.

4.1. Design Considerations. In order to optimize the performance of EDCA, it has been demonstrated that the AIFSN and the CW parameters are the most relevant factors [4–6]. Accordingly, the proposal focuses on determining the most suitable values for these parameters and adjusting them dynamically over time. In this way, it is possible to provide a more efficient mechanism to access the wireless medium. This improvement, in turn, allows it to reduce the collisions among the streams of both different and the same ACs. This leads to an enhancement in the QoS level, mainly for the voice and video traffic, due to its particular requirements for transmission in real-time.

Our scheme aims to improve the performance of real-time applications in Wi–Fi networks by reducing the number of collisions. One of the most serious difficulties in EDCA is precisely the huge amount of collisions that take place between voice and video transmissions, which are usually caused because they use the same AIFSN value and a very short length for the CW_{min} and CW_{max} parameters. In this context, a reduction of the number of collisions can be addressed in two ways: (i) increasing the separation between the AIFSN values for the voice and video applications and (ii) rising the length for the CW_{min} and CW_{max} parameters.

From the point of view of the AIFSN combination, the increase of the values must be performed on the video AC given that voice transmissions have stricter time constraints. The usage of AIFSN values higher than 2 slots for the video AC allows the voice applications to be provided with longer exclusive time periods to access the channel. Accordingly, collisions between voice and video frames would be reduced. Moreover, when voice packets collide and must be retransmitted, they could be long delayed or even discarded because of reaching the maximum deadline. As a consequence, the user experience would be highly damaged. On the other hand, if the AIFSN value for the video AC is increased, the values for the BE and BK ACs must be accordingly increased in an equal or greater proportion in order not to impair the video transmissions. However, if the network is partially composed of stations that use the original IEEE 802.11 standard (i.e., they use the DCF function), these legacy stations would have a higher priority to access the channel than the video applications, hence resulting in a noticeable decrease in the QoS performance.

Regarding the CW size, an increase of this parameter would directly lead to a reduction in the number of collisions in the network with respect to EDCA. This matter takes on particular importance for the voice and video traffic due to the short values defined for these ACs in EDCA. However, a significant enlargement of CW size would involve an increase in the waiting time to access the wireless medium. As a result, the real-time applications would be the most affected by this modification and the legacy stations would acquire even greater priority.

As can be seen, transmissions in wireless networks can be determined by several factors. For this reason, the complexity involved in determining the network conditions and tuning the medium access parameters according to them is considerably high. To perform this task, the most relevant factors are described below.

(i) Number of Active Applications of Each Type of Traffic. This parameter can be identified in a simple way by the AP. However, this is insufficient at a particular moment because it cannot provide further information about the current conditions of the network: that is, the scheme will not be allowed to obtain real information about the current occupancy of the wireless channel.

(ii) Application Bit Rate. Linked to the previous one, this factor provides more detailed information about the wireless medium status. Unfortunately, it is difficult to calculate in real-time. To identify these values it is necessary to introduce periodic control traffic in the network. Nevertheless, this feature is not typically used in IEEE 802.11e.

(iii) Transmission Rate. Each station may carry out its transmissions by using a different transmission rate. Therefore, the specific period of time that each of them keeps the channel busy is different. This parameter would be a good way of estimating the network conditions. Nonetheless, this value needs to be used jointly with the above factors.

(iv) Presence or Absence of Legacy DCF Stations. The existence of DCF applications restricts the use of priority parameters in EDCA, given that these values cannot be duly adjusted for these stations.

(v) Occupancy Level of the Wireless Medium. The amount of time that each traffic type keeps the medium busy allows us to obtain a good approximation of the network conditions. Indeed, the approximation that this only parameter can provide in a simple way is similar to that obtained by taking together all the previously described factors.

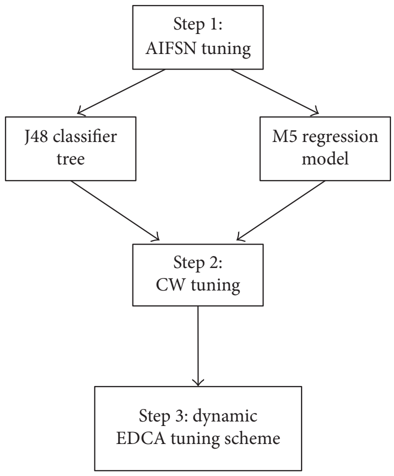


Figure 3: Dynamic EDCA prediction scheme's phases.

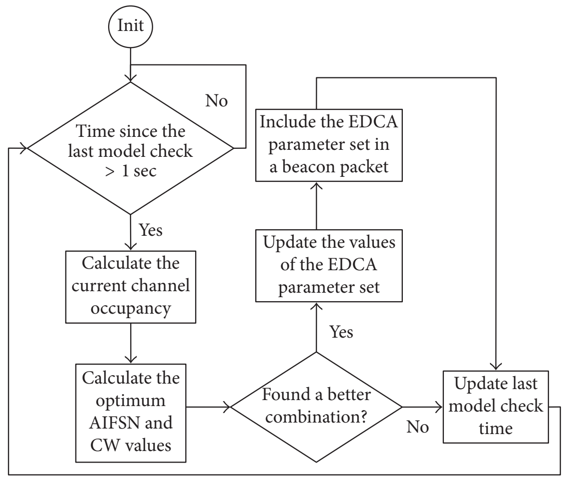


Figure 4: MAC parameters update process.

The proposed method requires to be used in real-time contexts without degrading the network performance, so the model should not be computationally complex. In this way, the medium access configuration can be recalculated at short intervals to achieve more accurate results. In this regard, AI techniques may be useful in identifying traffic patterns and determining optimum priority values for each AC. Furthermore, these techniques make it possible to simplify the huge amount of information that is involved in determining the network conditions. In this way, it is possible to handle only the information which plays a critical role in ascertaining the channel status.

4.2. Proposal Description. Given the presented constraints, our scheme takes into account the considerations described above. Firstly, the algorithm selects the optimum AIFSN combination depending on the current channel conditions. After that, and based on these values, the scheme calculates accordingly the most appropriate size for the CW parameter.

To ensure a better QoS differentiation, this proposal is divided into three phases, as depicted in Figure 3. The *first phase* involves the adaptation of the AIFSN combination through the design of two independent predictive models: an M5 regression model and a J48 classifier tree. To design these models, a deep training step must be performed. This training must consider a wide range of network situations to ensure that the models are built as close as possible to real environments. The prediction obtained from these models is used in the *second phase*, when the optimization of the CW size is carried out. In the *third phase*, all the designed models are unified into a single scheme which is able to discern the use of every one of them according to the network conditions. The appropriate selection of these priority values makes it possible to reduce both the internal and external collisions. This result is especially suitable for the multimedia traffic enhancement. Moreover, the usage of the wireless channel is optimized and, as a consequence, the network performance is also improved.

The proposed scheme is loaded in the APs of the network by introducing a few modifications in the firmware. In this regard, the predictive models, which are already trained and built, are ready to be used at any time. Therefore, it ensures the direct application on real environments, where the information about the network conditions is only used as input for the predictive scheme to obtain and update the appropriate EDCA values.

Traffic conditions may change rapidly, so the periodic update of the priority parameters becomes an important matter to consider. After evaluating different amounts of time, this recalculation period has been set to one second. Within the selected time, the AP of the network must determine the optimum priority values of that very moment and check whether they are equal to the current ones. If any difference is found, the AP must notify these new parameters to the stations. Once the optimum EDCA combination has been found, the updated information is embedded in an EDCA parameter set and transmitted via beacon frames. In this way, it avoids introducing additional control traffic into the network. The process described above can be observed in Figure 4.

The beacon interval is usually set to 100 ms; therefore, if required, the values could be updated at every interval. However, channel conditions do not usually suffer relevant changes in such a short period and it would only cause abrupt modifications in the parameters that would not allow the algorithm to converge properly. The tests performed to set this period showed that the results were similar for updating periods from 1 to 5 seconds, these being less accurate when selecting a longer interval due to the excessive amount of time without updating the network information. Given the similarity among the previous periods, the final selection of a one-second interval was also due to the intention of implementing a scheme as appropriate for real environments as possible, where noises or movements may change suddenly the channel status.

Table 2: Features of the dynamic adaptation schemes.

	IEEE 802.11 compatibility	Additional traffic	Efficiency	Prioritization	QoS performance with DCF and EDCA stations	Designed for saturated traffic	Main issues
Our proposal	Yes	No	Efficient	AIFSN + CW	Higher than EDCA	No	—
i-EDCA [7]	Yes	No	Efficient	CW	Lower than EDCA	No	Bad result in the presence of DCF stations. Only uses CW
DACKS [8]	Yes	No	Very inefficient	CW	Higher than EDCA	No	Very inefficient. Retransmits correct packets
AEDCA [9]	Yes	No	Efficient	CW	Lower than EDCA	No	Bad results in the presence of DCF stations. Only uses CW
MMDP-FMAC [10]	Yes	No	Efficient	Fair allocation	Lower than EDCA	No	Bad results in the presence of DCF stations. Problems with the deadline of the real-time applications
Hamidian and Körner [11]	Yes	No	Efficient	TXOP	Similar to EDCA	No	Problems with the deadline of the real-time applications
QHDCF [12]	No	Yes	Efficient	Centralized priority	Higher than EDCA	No	Incompatible with the standard. Uses modified packets
iPAS [14]	No	Yes	Inefficient	Centralized priority	Lower than EDCA	No	Incompatible with the standard. Requires stations changes. Introduces additional control traffic
Banchs and Vollero [17]	Yes	No	Efficient	CW	Lower than EDCA	Yes	Bad results in the presence of DCF stations. Only uses CW.

Table 2 compares the main features of our scheme with those described in Section 2.2. The Table shows that some proposals have compatibility issues with the IEEE 802.11 standard. Therefore, these schemes cannot be used in real-world scenarios. On the one hand, there is a set of approaches that do not consider the large number of wireless cards whose features are still based on the original IEEE 802.11. These schemes (i-EDCA, AEDCA, MMDP-FMAC, iPAS, and Banchs et al.) achieve very poor results in the presence of stations that use DCF. In other words, the QoS performance achieved by these approaches is lower than the one offered by EDCA. This problem arises from the modification of the channel access function in order to improve the performance of the voice and video applications. For that reason, the usage of these schemes when the network is also composed of legacy DCF stations involves an increase in the priority of these last stations with regard to the video applications.

On the other hand, most of the evaluated schemes achieve good results in terms of efficiency since they do not require introducing additional traffic in the network. However, DACKS and iPAS schemes should be noted as exceptions. To reduce the priority of the legacy stations, in DACKS not all the frames that are successfully transmitted are acknowledged. Consequently, many properly received frames are retransmitted, hence introducing unnecessary traffic in the network and increasing the CW size. Meanwhile, iPAS uses a dedicated control communication link to send feedback information. As a result, besides the data link, an additional one is established between server and clients for the control traffic.

5. Deployment Process

The design of the dynamic predictive model requires the prior implementation of the two first phases of the deployment.

TABLE 3: Set of AIFSN values analysed.

	S0	S1	S2	S3	S4	S5	S6	S7	S8	S9
BK	7	8	9	8	9	12	10	12	14	14
BE	3	4	5	4	5	6	6	8	10	12
VI	2	2	2	3	3	3	4	5	6	7
VO	2	2	2	2	2	2	2	2	2	2

As outlined in Section 4, these phases allow the independent adaptation of the AIFSN combination and the CW size for every AC in EDCA. This section presents the development performed in the *first* and *second phases*. The *first phase* also includes the description of the steps that have been followed in the training and learning phases to model the initial schemes. Given that the results are shared from one phase to another, the evaluation of these preliminary parts is also included.

5.1. AIFSN Adaptation. The models of the *first phase* aim to optimize the channel access by identifying the most suitable AIFSN combination. In this regard, a J48 decision tree classifier and an M5 regression model have been developed. The modeling of these schemes has been carried out independently in order to compare their capabilities and analyze how the different factors of a network determine their performance. Further details concerning this analysis, the development process and the evaluation performance, can be found in our previous works [26, 27].

According to the analyses carried out in [5, 6], a group of 10 AIFSN configurations has been selected with the goal of being an appropriate alternative to the one established in EDCA. These combinations, which can be observed in Table 3, have been chosen on the basis of the requirements previously introduced in Section 4.1. A gradual increase in the waiting time among each AC has been carried out for that selection with the aim of reducing the number of collisions and to enhance the QoS performance of the network. In particular, these combinations could be grouped into three different categories, as detailed below.

In the first category, 3 groups of combinations that consider an AIFSN value equal to 2 slots for both the AC_VO and the AC_VI can be found. The usage of these values does not allow reducing the collisions between voice and video frames. However, this intends to analyze the effect that increasing the AIFSN values for the BE and BK traffic flows has in reducing the aforementioned collisions. This group of combinations is advisable in the presence of a considerable amount of DCF traffic given that the legacy stations would not have a higher priority than the video applications to access the channel.

The second division includes 3 sets of combinations that establish an AIFSN equal to 2 slots for the AC_VO and equal to 3 slots for the AC_VI. These values will considerably reduce the collisions between voice and video applications due to the reserved time slot for the voice streams. Moreover, with the aim of studying their effect on the real-time applications performance, different AIFSN values have been assigned to the AC_BE and AC_BK. The values considered in this group are particularly suitable for a low DCF traffic load because the reduction achieved in the number of collisions compensates the decrease in the priority for the video transmissions.

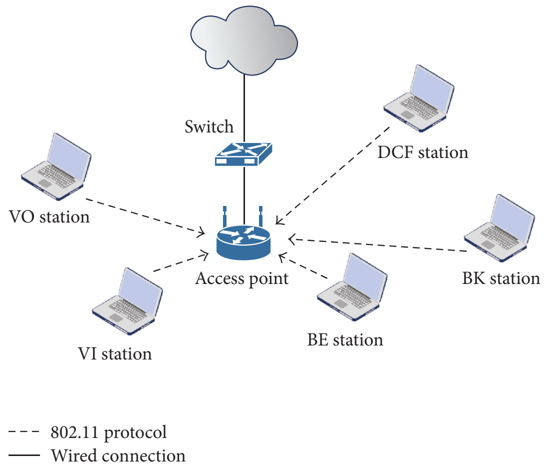


FIGURE 5: Network topology used in the deployment.

The last group is composed of 4 sets of values that carry out a greater increase in the AIFSN value for the AC_VI than the previously performed. The usage of these combinations involves a greater reduction in the multimedia traffic collisions with respect to the previous ones. This is particularly beneficial for the voice applications given that they would have several time slots for their exclusive use. Nevertheless, it should be noted that these combinations are not recommended for the cases in which there is a huge number of legacy stations due to the reduction in the priority for the video transmissions.

The model design to predict the AIFSN combination starts with a training phase, which requires us to acquire a considerable amount of information. Accordingly, a huge set of tests covering a wide range of traffic conditions must be specified. This makes it possible to discover valuable knowledge and offer more precise predictions. In this work, the training tests are composed of 18 scenarios which have been modeled using Riverbed Modeler 18.0.0 [28]. In our simulations, we model an IEEE 802.11g wireless LAN cell comprising legacy DCF-based and EDCA stations. The EDCA stations support four different types of services: Voice (VO), Video (VI), Best Effort (BE), and Background (BK). The DCF-based stations support data traffic. We assume the use of a wireless LAN consisting of several wireless stations and an AP connected to a wired node that serves as sink for the flows from the wireless domain. All the stations are located within a Basic Service Set (BSS); that is, every station is able to detect the transmission from any other station. The parameters for the wired link were chosen to ensure that the bandwidth bottleneck of the system is within the wireless LAN (see Figure 5).

The scenarios are divided into two major groups: the first one considers both DCF and EDCA stations; whereas, the second group is only composed of stations that make use of EDCA. Accordingly, this proposal can ensure compatibility

Table 4: Traffic proportions used for the training step.

# scenario	VO	VI	BE	BK	DCF
1	20%	20%	20%	20%	20%
2	60%	10%	10%	10%	10%
3	30%	30%	10%	10%	20%
4	10%	60%	10%	10%	10%
5	30%	20%	10%	10%	30%
6	10%	10%	10%	10%	60%
7	10%	10%	20%	20%	40%
8	10%	10%	40%	30%	10%
9	50%	50%	—	—	—
10	10%	10%	30%	30%	20%
11	60%	40%	—	—	—
12	40%	60%	—	—	—
13	—	—	40%	30%	30%
14	30%	—	20%	20%	30%
15	—	30%	20%	20%	30%
16	—	—	50%	50%	—
17	20%	—	40%	40%	—
18	—	20%	40%	40%	—

Table 5: Traffic parameters used for classifier construction.

	Packet size	Data rate
DCF	552 bytes	512 Kbps
BK	552 bytes	512 Kbps
BE	552 bytes	512 Kbps
VI	1064 bytes	800 Kbps
VO	104 bytes	20 Kbps

with the legacy DCF stations. Each scenario starts with ten stations. In each scenario, the number of stations is increased from 10 to 80 in steps of 10 in order to increase the load of the wireless network. All the scenarios are composed of a variable percentage of applications of each type of traffic (BK, BE, VI, and VO). The traffic distributions used during the training phase can be seen in Table 4.

In order to adapt the medium access parameters in EDCA, the AP needs to know the channel occupancy rate at a given moment. This rate for the downlink traffic can be easily calculated by the AP. However, the main difficulty lies in estimating periodically the uplink traffic on the network. For this reason, the usage of only uplink traffic has been considered as a design decision. Nevertheless, the scheme can be also used in the presence of downlink traffic. In this case, both the uplink and the downlink occupancy rate are used as input for the algorithm to calculate the EDCA configuration.

In 802.11 WLANs stations use different rates, which determine the channel occupancy. Accordingly, the usage of a set of values such as 12 and 36 Mbps has been considered for all the stations in the network, regardless of their AC. The remaining traffic features are unique to each type of traffic, as shown in Table 5. The BK, BE, and DCF traffic transmissions are modeled via a Pareto distribution with a location of 1.1 and a shape of 1.25. The voice and video flows are represented by the transmission of G728 [29] and H.264 [30] streams, respectively. In the case of congestion, the network traffic is susceptible to experience delivery delays. Voice and video communications are more sensitive to the effects of this phenomenon than data transmissions. In this regard, deadline delays of 10 and 100 ms have been established for voice and video streams, respectively, beyond which these streams are discarded.

The simulations of the training phase are carried out by defining scenarios where the conditions of all the factors that determine them remain static. The values for such factors are modified according to a given order until all possible combinations have been considered. In this way, the predictive models are allowed to acquire real knowledge. In fact, if variable and random information were provided to the aforementioned models, the learning process would be unfeasible.

Although a considerable amount of information is needed to train the models, not all the parameters used are included. Some of them do not add any relevant information and merely made these models more complex. With the aim of designing accurate but simple classifiers, they must undergo a supervised variable selection process to discard those unrelated parameters. After carrying out this process, only the global occupancy level of the wireless channel and the particular level of each type of traffic are considered by the models.

Each predictive algorithm is provided a unique set of values. The J48 tree classifier only considers those AIFSN combinations that maximize the voice + video normalized throughput in each scenario. In contrast, the M5 regression model is given the information to all the AIFSN combinations, regardless of the throughput achieved. Since this model makes use of a group of regression functions to maximize the aforementioned throughput, this value must also be provided. As a result, the M5 model finally generates ten sets of submodels; that is, it contains one group of rules per AIFSN tested combination. This whole process makes it possible to obtain a good approximation of the network conditions while allowing the design of simple and accurate predictive models.

Both the preprocessing step and the design of the models have been carried out using Weka 3.7.0 [25]. As a part of the design process, a 10-fold cross validation is performed to guarantee that both the training and the testing data sets are independent. This process results in a hit rate of 94.90% for the J48 tree. Meanwhile, the M5 model achieves an average correlation coefficient of 0.8916 and a mean absolute error of 0.0554. These values show the accuracy of the proposed models and the strong relation among the parameters involved.

The *second phase* of the work and the tuning of the CW size take as a starting point the enhancements achieved by the first one in order to address the existing weaknesses. For this reason, a performance analysis of the models is required. This process is performed via simulation using Riverbed Modeler 18.0.0, through which 20 scenarios different from those used during the training step have been designed. The network topology used in the evaluation is the same as that presented in the training phase (see Figure 5). The scenarios are composed of 100 stations which use DCF and EDCA as medium

Table 6: Description of the set of test scenarios.

# scenario	Voice	Video	BE	BK	DCF
1	20%	1.5%	2%	2%	2%
2	20%	5%	2%	2%	2%
3	8%	7%	2%	2%	2%
4	16%	6%	3%	3%	7%
5	6%	2%	3%	3%	10%
6	6%	3%	4%	4%	8%
7	5%	3%	7%	7%	4%
8	5%	6%	10%	5%	5%
9	5%	9%	6%	6%	6%
10	8%	—	8%	8%	8%
11	—	6%	6%	6%	9%
12	5%	6%	6%	6%	6%
13	20%	8%	—	—	—
14	9%	4%	—	—	—
15	5%	10%	—	—	—
16	6%	7%	7%	7%	—
17	8%	—	8%	8%	—
18	—	8%	7%	7%	—
19	8%	8%	6%	6%	—
20	8%	7%	8%	—	—

access functions. The first twelve scenarios consider both types of stations, while in the remaining eight only stations that support EDCA can be found. Furthermore, an equal proportion of stations of each type of traffic has been considered; that is, 20 stations per type of traffic have been included. However, not all the applications are active at the same time. On the contrary, they are enabled or disabled according to a probability that depends on their AC (see Table 6).

Every scenario takes 600 seconds to be simulated, with this simulation divided into two periods. During the first one, and every 30 seconds, the stations which are not transmitting any information try to start a new transmission according to the aforementioned probability. During the second period, the stations attempt to stop the current transmissions every 30 seconds, making use of the same probability. Due to all scenarios being simulated by using 60 random seeds and each of them being divided into 20 time intervals, in the end 24000 different intervals have been tested.

The traffic features for each station depend on the type of traffic it transmits. Regarding this, the bit rate and packet size used to verify the models are the same as those shown in Table 5. The stations are randomly distributed over the network coverage. Furthermore, the Ricean [31] model determines the signal propagation through the wireless medium. This model is characterized by a factor, k, which defines the ratio between the power in the line-of-sight component and the power in the scattered paths. In this work, a k factor of 32 has been used. Moreover, IEEE 802.11g [32] defines the physical layer of the network. The stations use all the transmission rates defined in this amendment regardless of the traffic type.

The results allow it to compare the performance achieved by the proposed schemes and EDCA. To summarize these

Table 7: Voice + video normalized throughput improvements in 30 s intervals (AIFSN tuning).

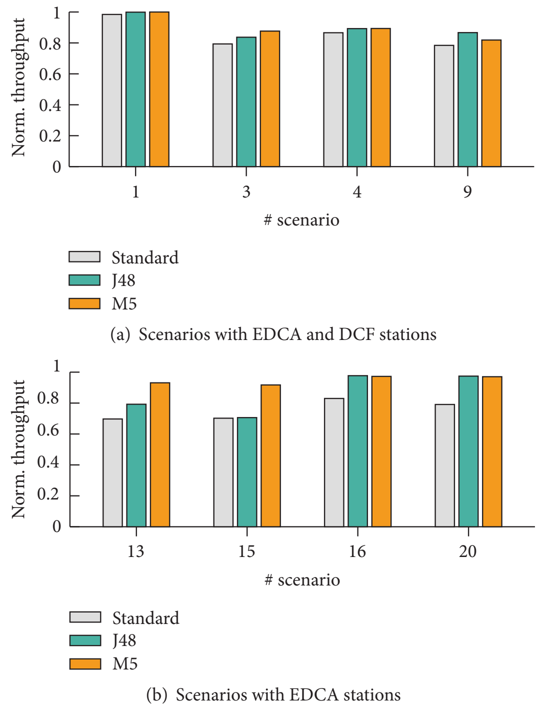
	With DCF traffic		Without DCF traffic	
	J48	M5	J48	M5
Unaltered	49.42%	52.11%	35.52%	31.30%
Losses	4.49%	5.48%	2.55%	1.44%
Gain [1%–5%]	27.20%	23.37%	23.64%	17.35%
Gain [5%–10%]	8.67%	12.78%	6.68%	5.93%
Gain [10%–15%]	6.06%	2.86%	7.41%	6.89%
Gain [15%–20%]	2.01%	1.52%	4.48%	4.55%
Gain [from 20%]	2.16%	1.90%	19.73%	32.55%

results properly, a group of statistics has been defined. These statistics show the voice + video normalized throughput, the number of retransmission attempts, the normalized throughput of the DCF applications, and the global throughput of the network. The first metric refers to the sum of the normalized throughput of both voice and video applications.

Table 7 shows the percentage of the 24000 time intervals evaluated in which the proposed models experience losses or gains with regard to EDCA. Notice that, especially if both DCF and EDCA applications are considered, almost half of the intervals remain unaltered. We have considered unaltered results to be those in which the gains or the losses are lower than 1%. This situation is a consequence of the stations attempting to both start and finish their transmissions during the first and the last five simulation intervals. The traffic load in many of these cases is low, so all the AIFSN combinations achieve the highest throughput. Moreover, in the scenarios in which many stations use DCF, the default EDCA combination is the most efficient option. Therefore, in these scenarios the predictive schemes also use these values to access the medium.

Table 7 also presents some losses with regard to EDCA. Losses have been defined as those intervals in which the models experience a performance decrease higher than 1%. These losses result from some certain cases in which our schemes miss the prediction. The use of a small number of parameters to design the models provides a good approximation of the network conditions. However, due to its simplicity, this image is not perfect, having as a consequence some errors in the parameter prediction. The aforementioned losses are also because the selection of a different AIFSN combination makes some minimum changes in the traffic status that might be considered as losses when they really are not. Nevertheless, the cases in which this phenomenon occurs are much fewer than those in which the schemes outperform EDCA.

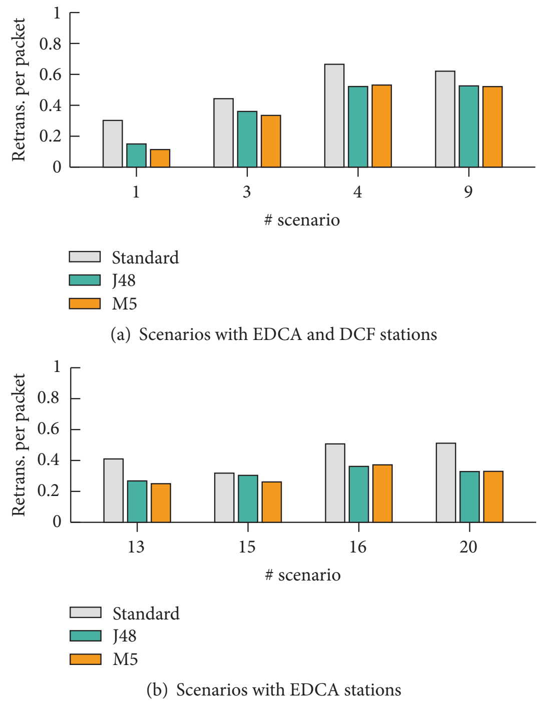
Nevertheless, Table 7 shows how our models improve the performance achieved by EDCA in many more scenarios. These improvements are from 20% in some cases, being even higher if only EDCA applications are considered. Thus, in 19.73% and 32.55% of the intervals, enhancements of 20% are achieved when using the J48 and M5 models, respectively. Given that in some intervals the traffic load level does not allow the observation of the performance difference between the proposal and EDCA, Figure 6 presents the voice + video



(a) Scenarios with EDCA and DCF stations

(b) Scenarios with EDCA stations

FIGURE 6: Voice + video norm. Throughput (AIFSN adaptation).



(a) Scenarios with EDCA and DCF stations

(b) Scenarios with EDCA stations

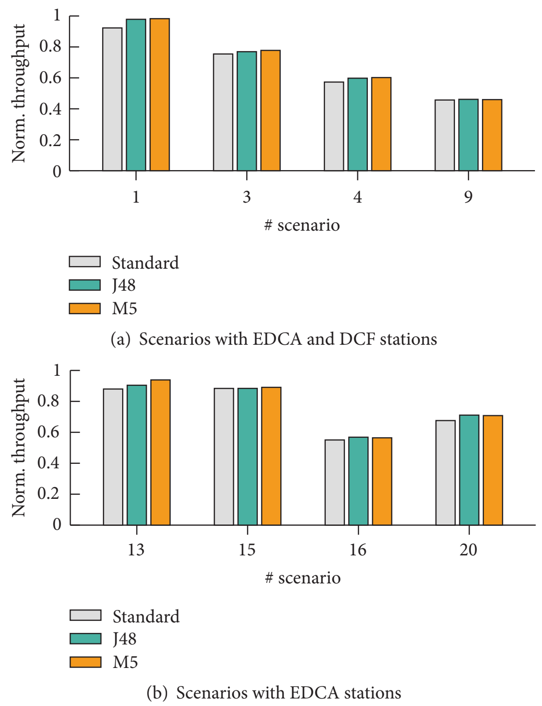
FIGURE 7: Overall retransmission attempts (AIFSN adaptation).

normalized throughput of the ten intervals in which traffic congestion becomes a key factor. In this figure, it is proved that in all the scenarios the predictive schemes outperform EDCA.

The proximity of the EDCA values, especially between the voice and video traffic, constrains the network performance due to the huge number of collisions. The appropriate AIFSN tuning of this approach allows it to optimize the channel access, resulting in a decrease in the number of collisions and retransmissions (see Figure 7). This is the main reason why the QoS level provided to real-time applications is enhanced. Furthermore, this improvement also leads to an increase in the overall throughput of the network, as can be seen in Figure 8.

A suitable separation of the AIFSN values, particularly in those cases in which the AIFSN for video traffic is higher than 2 slots, provides the stations that do not support QoS features with a higher priority to access the wireless medium. While ensuring compatibility with this type of stations, the proposed models improve their performance, as can be observed in Figure 9. It is important to point out that, despite this increase in the priority, the performance achieved by the multimedia communications is not only penalized but also enhanced in all the cases tested, as depicted in Figure 6.

Although both the presented predictive models enhance the performance of EDCA, they have certain features which make them quite different. On the one hand, the features of the J48 classifier allow it to achieve better results in the



(a) Scenarios with EDCA and DCF stations

(b) Scenarios with EDCA stations

FIGURE 8: Global norm. Throughput (AIFSN adaptation).

Table 8: Proposed values for the CW_{min} and CW_{max} limits.

	CW 1		CW 2	
	CW_{min}	CW_{max}	CW_{min}	CW_{max}
BK	$2 \cdot aCW_{min} + 1$	aCW_{max}	$2 \cdot aCW_{min} + 1$	aCW_{max}
BE	$2 \cdot aCW_{min} + 1$	aCW_{max}	$2 \cdot aCW_{min} + 1$	aCW_{max}
VI	aCW_{min}	$2 \cdot aCW_{min} + 1$	$2 \cdot aCW_{min} + 1$	$2 \cdot aCW_{min} + 1$
VO	$(aCW_{min} + 1)/2 - 1$	aCW_{min}	aCW_{min}	aCW_{min}

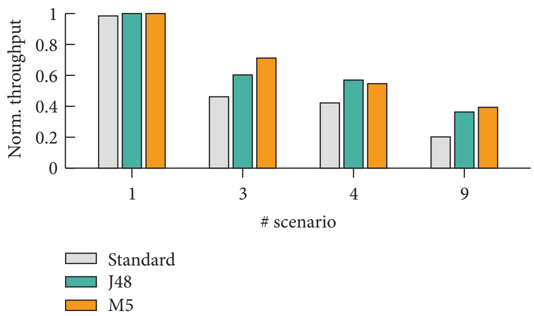


Figure 9: DCF norm. Throughput (AIFSN adaptation).

presence of a considerable load of DCF traffic. On the other hand, the gains experienced by the M5 model are higher in those scenarios in which DCF traffic is not considered or the number of these transmissions is relatively low. Moreover, in spite of the fact that neither model is computationally complex, the M5 regression model requires evaluating the voice + video normalized throughput for every AIFSN combination. Therefore, the J48 decision tree has a lower computational cost due to the minimum set of comparisons needed. This feature is especially important with regard to the execution of the medium access function in real-time.

5.2. CW Adaptation. The main aim of the *second phase* of this proposal is to further improve the multimedia communications through the reduction of the collisions among the flows of the same AC. Based on the evaluation performed in the *first phase*, it has been shown that in several scenarios there is still room for improvement. In this regard, in the *second phase*, the collision reduction problem is addressed by estimating the optimum values for the CW size without neither increasing severely the waiting time to access the channel nor reducing the priority of the real-time applications.

As is the case with the AIFSN, the IEEE 802.11e amendment establishes a set of values for the CW size (see Table 1). These values seek to ensure an appropriate QoS level, while maintaining compatibility with the legacy stations. Indeed, they aim to achieve good network performance when it hosts a considerable amount of legacy applications. In these cases, the selection of longer periods for the CW size would result in an earlier access to the medium by the DCF stations. In this way, the priority of this type of traffic would be increased with regard to video traffic, instead of being similar to that of the BE streams. For this reason, the adaptation of the CW parameter is of particular interest in scenarios where there is a low or null DCF traffic load.

Taking into account the huge number of collisions in Wi–Fi networks, mainly between voice and video transmissions due to the very small values of the CW_{min} and CW_{max} parameters, we have considered relevant to study the effect of carrying out these last increases. In order not to prejudice the priority of the multimedia applications the size of the CW_{min} for AC_BE and AC_BK has been also enlarged. The CW_{max} of these ACs has not been increased due to its high value.

Based on the prediction of the schemes designed in the *first phase* of the proposal, the CW_{min} and CW_{max} limits are increased with regard to the default EDCA values, as can be seen in Table 8. In the Backoff algorithm the CW size follows an incremental sequence of $2^c - 1$ whenever a station needs to retransmit a frame. This factor is given special attention in this *second phase*, with the proposal of two approaches called *CW 1* and *CW 2*.

The first modification of the CW size, called *CW 1*, increases the c parameter by one with respect to EDCA for every AC. The only exception is the CW_{max} limits for the BE and BK ACs, which maintain the default values. Given that these values are large enough, they are not modified with the aim of not introducing unnecessary waiting times into the network. The second version, called *CW 2*, follows a similar pattern to its predecessor. As the objective of this second proposal is to avoid the collisions between the voice and video traffic, only the duration of CW_{min} of these ACs is modified by increasing the c parameter by two units. Furthermore, the CW_{max} limits for all the ACs keep the same values as those used in the *CW 1* approach. Notice that neither of the proposed configurations considers a reduction in the size of CW. This reduction, especially for the ACs whose limits are relatively small such as voice and video ACs, would result in a huge increase in the number of collisions between the frames of the same type of traffic.

An increase in the size of CW involves longer periods of time for the stations to finish the Backoff algorithm. More specifically, some combinations of AIFSN values and CW sizes may result in a considerable decrease in the priority of the stations that use EDCA with regard to those that use DCF. For this reason, the four approaches presented in Table 9 reduce the distance among the AIFSN values of all the ACs in order not to accumulate excessive waiting periods. All of these approaches aim to show the performance achieved when combining an increase in the length selected for the CW_{min} and CW_{max} parameters with the use of lower values from those predicted for the AIFSN combination. In this

TABLE 9: Optimization configurations for the CW size and AIFSN combination.

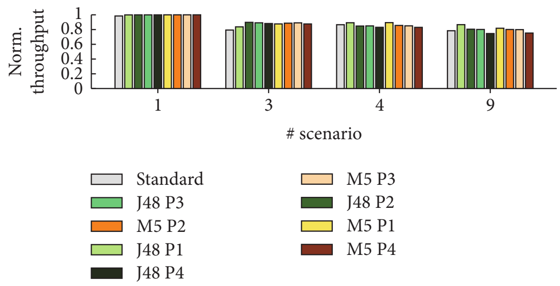
	Configuration 1	Configuration 2	Configuration 3	Configuration 4
AIFSN	Predicted	Predicted	Prev. combination	2nd Prev. combination
CW	Default	CW 1	CW 1	CW 2

way, the objective is to verify whether this strategy is able to outperform the results already achieved for voice and video communications. Finally, a total of eight predictive schemes are obtained given that each of the four proposed configurations can be used with both the J48 and the M5 models.

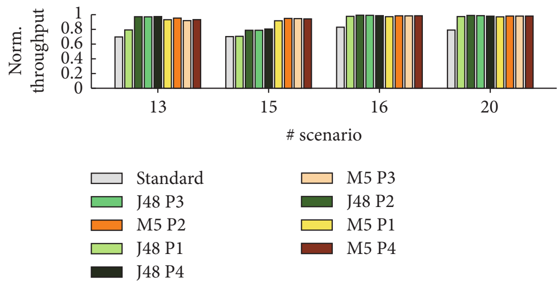
The first configuration proposed in Table 9, called *Configuration 1*, refers to the already modeled schemes in the *first phase*. In the remaining configurations, the adaptation of the CW size is carried out according to the values shown in Table 8. In this regard, *Configuration 2* and *Configuration 3* take the CW size of the combination *CW 1*. By contrast, *Configuration 4* takes the values related to the *CW 2* approach. Furthermore, the AIFSN combination can be chosen from among 3 different sets of values. Firstly, the prediction made for this parameter in the *first phase* can be maintained, as is the case with *Configuration 1* and *Configuration 2*. Moreover, as is done for *Configuration 3*, another option is to select the AIFSN combination immediately below the one predicted by the J48 and M5 models. Likewise, the AIFSN combination can be tuned using the values provided by the second combination immediately below the predicted one. More specifically, this combination, which is used by *Configuration 4*, establishes a smaller spacing among the values of all the ACs. The selection of the aforementioned AIFSN combinations is carried out whenever possible, provided that they do not exceed the default EDCA one.

The real aim of this proposal is to design a dynamic prediction scheme for the EDCA parameter set. For this reason, in addition to the results obtained from the AIFSN tuning in the *first phase*, it is necessary to evaluate the performance achieved by these eight new preliminary schemes. In fact, the results of both the two phases will become the basis for developing the dynamic predictive model in the *third phase*. In order to carry out this task, we have considered the same simulation conditions and scenarios defined in the previous evaluation. In this way, it is possible to ensure a fair comparison of the results of both phases.

In Figure 10 the results in terms of voice + video normalized throughput are presented. The quality of the multimedia applications is mainly conditioned by the presence of stations that use DCF. This is due to the fact that a too large increase in the size of CW, particularly when the traffic load of the legacy stations is relatively high, has as a consequence a reduction in the performance of the voice and video transmissions. Despite these aspects, a small increment in the size of CW under low DCF traffic loads contributes to outperforming the results achieved by the J48 and M5 initial models. Nevertheless, this situation differs in scenarios where there are no stations which use the DCF function. In most of these cases, the usage of a larger size for CW makes it possible to improve the performance of the real-time applications. Various examples of this phenomenon can be observed in Figure 10.



(a) Scenarios with EDCA and DCF stations



(b) Scenarios with EDCA stations

FIGURE 10: Voice + video norm. Throughput (CW adaptation).

The enhancements achieved in this *second phase* are mainly due to the reduction of both internal and external collisions among the traffic flows of the different ACs. As a consequence, a decrease in the number of retransmissions in the network is achieved. In the presence of a high number of stations that utilize the DCF function, using an AIFSN combination below the one that has been already predicted in the *first phase* of the proposal might result in an increase in the number of retransmissions. In spite of this fact, and as can be observed in Tables 10 and 11, the adequate selection of the size of CW allows the schemes to keep this number of retransmissions constant or even lower.

The increase in the waiting time to access the wireless medium for the video traffic flows has as a consequence an improvement in the performance of the legacy applications that use DCF. This phenomenon can be observed in Table 10, in which the legacy DCF stations achieve a higher throughput than in the *first phase* of the development. Nevertheless, this is not the real aim of these predictive schemes. In scenarios in which the DCF traffic load becomes high and the proposed approaches increase the duration of the CW parameter, the performance of these applications is improved, even if it is a secondary objective of the proposal. As a result, the global performance of the network is also enhanced. As it is shown

TABLE 10: Overview of the CW adaptation results (scenarios with DCF and EDCA stations).

	Standard	J48 P1	J48 P2	J48 P3	J48 P4	M5 P1	M5 P2	M5 P3	M5 P4
VO + VI Norm. Th.	0.9259	0.9440	0.9225	0.9227	0.9016	0.9438	0.9312	0.9242	0.9022
Retrans. per Pkt	0.5447	0.4459	0.3469	0.3636	0.3662	0.4271	0.3473	0.3659	0.3724
Global Norm. Th.	0.6014	0.6199	0.6420	0.6395	0.6425	0.6225	0.6399	0.6379	0.6390
DCF Norm. Th.	0.5498	0.6790	0.7965	0.7743	0.7715	0.7118	0.8056	0.7876	0.7812

TABLE 11: Overview of the CW adaptation results (scenarios with EDCA stations).

	Standard	J48 P1	J48 P2	J48 P3	J48 P4	M5 P1	M5 P2	M5 P3	M5 P4
VO + VI Norm. Th.	0.7759	0.8790	0.9415	0.9385	0.9419	0.9338	0.9596	0.9531	0.9549
Retrans. per Pkt	0.4604	0.3428	0.2271	0.2456	0.2234	0.3329	0.2288	0.2450	0.2267
Global Norm. Th.	0.6776	0.6965	0.7329	0.7320	0.7450	0.7010	0.7304	0.7296	0.7408

in Table 11, the predictive schemes achieve an enhancement in terms of voice + video normalized throughput in the absence of legacy DCF stations, which in turn involves an improvement in the global performance of the network.

Following the evaluation performed in this *second phase*, it can be concluded that the adaptation of the size of CW keeps enhancing the quality of the multimedia transmissions. Nevertheless, it must be taken into account that an excessive increase in this parameter in the presence of a considerable number of stations that use DCF would penalize the performance of the video applications. In fact, in view of these situations, this increment should not be carried out.

Finally, the wide variety of results that have been obtained in both the already presented phases is clearly noticeable. As a consequence, it requires carrying out a deep analysis process to define in a precise way the network conditions in which the different approaches achieve the optimum performance. In fact, this analysis becomes the starting point for the development of the dynamic prediction scheme for the medium access parameters in EDCA.

```
If (DCF_channel_ocuppancy == 0 &&
    BE_channel_ocuppancy == 0 &&
    BK_channel_occupancy == 0)
    M5 Proposal 2

else if (DCF_channel_ocuppancy == 0)
    J48 Proposal 3

else if (DCF_channel_ocuppancy <= 0.15)
    M5 Proposal 3

else if (DCF_channel_ocuppancy > 0.15)
    J48 Proposal 1
```

ALGORITHM 2: Structure of the dynamic prediction scheme.

6. Dynamic EDCA Prediction Scheme

In the previous section, four medium access parameter configurations are proposed for the J48 and M5 initial models (see Table 9). As a result, a total of 8 predictive schemes have been obtained. These schemes are named after the predictive model (M5 or J48) and the configuration used, for example, *M5 Configuration 1* for the scheme that uses the first parameter configuration of the M5 model.

Given the wide variety of schemes and their achieved results, it is essential to unify them into a single one which is able to discriminate among all of them. This unification must be carried out through an exhaustive analysis of the results obtained. In this regard, a set of traffic patterns, which determines the structure of the dynamic prediction scheme has been identified. This structure is shown in Algorithm 2, where the behavior of the algorithm is mainly conditioned by the presence of DCF traffic. In spite of having designed 8 predictive models, not all of them are able to predict the medium access parameters accurately. Moreover, it has been proved that several schemes achieve similar results. Therefore, the final dynamic scheme is only composed of 4 submodels.

The analysis of the results consists of studying the parameter configurations that achieve the highest performance in each one of the proposed scenarios. After that, the results have been filtered in order to choose only the most suitable combination in every case. However, in some scenarios, several configurations provide similar results in accordance with the established traffic patterns. On this basis, and to simplify the model, only one of these potential candidates has been finally selected for the corresponding traffic pattern.

The evaluation highlights that, in the presence of DCF traffic, its occupancy level plays the most important role. If this level is higher than 15%, the best results are provided by the first version of the J48 classifier (*J48 Proposal 1*). This model does not increase the size of CW, which is a proper decision that does not penalize the video applications when accessing the medium. In fact, in the first development phase, it was proved that, when the network carries a high DCF traffic load, the J48 model achieves better performance than the M5 one. By contrast, if this occupancy level is lower than 15%, *M5 Proposal 3* achieves the best performance. As this traffic level is relatively low, this selection is appropriate due to the fact that the increase in the CW size makes it possible to reduce the collisions among the frames of the same AC.

Table 12: Voice + video normalized throughput improvements in 30 s intervals (dynamic scheme).

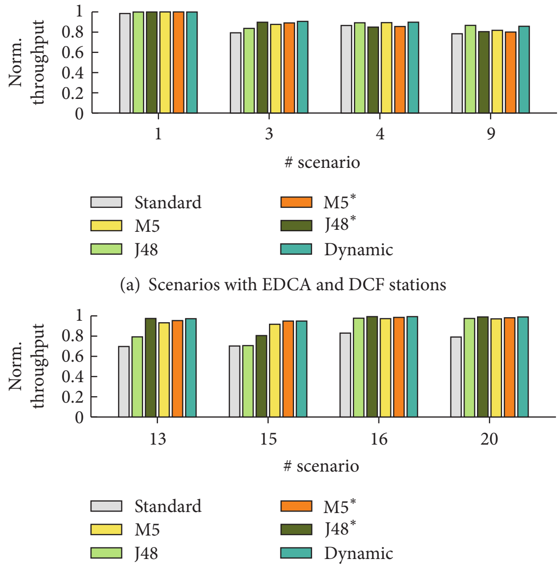
	With DCF traffic	Without DCF traffic
Unaltered	56.37%	34.27%
Losses	2.46%	1.19%
Gain [1%–5%]	23.29%	15.01%
Gain [5%–10%]	6.49%	5.95%
Gain [10%–15%]	2.96%	7.54%
Gain [15%–20%]	2.94%	6.19%
Gain [from 20%]	5.49%	29.85%

By contrast, if there are no applications without QoS support, voice and video applications become the most important factor. When the network only carries voice and video traffic, *M5 Proposal 2* provides the most accurate prediction. Given that this model keeps the AIFSN prediction and increases the CW size, it enables a reduction in the number of collisions among the multimedia streams and, as a result, an improvement in the performance of the network. Meanwhile, if other types of traffic are considered, the best option is to select the third version of the J48 classifier (*J48 Proposal 3*). This model offers a trade-off between the CW size and the differentiation given by the AIFSN values chosen. In this context, with the exception of the first proposal of this model which does not take into account the size of CW, the remaining ones achieve a similar performance.

In addition to implementing the design presented above, the real adaptation capacity of the proposal must be verified. In this respect, to compare the performance achieved, the statistics show the results for EDCA, both the J48 and M5 initial models and the schemes of the *second phase* that achieve a higher voice + video normalized throughput. These latter schemes are denoted as J48* and M5*. The results for the modeled dynamic scheme are also included.

Table 12 presents the results of the voice and video performance in terms of the percentage of time intervals into which simulations are divided. Notice that the percentage of unaltered intervals presents only minor changes with regard to previous evaluations (see Table 7). However, taking advantage of the strengths of each model makes it possible to reduce the amount of losses with regard to EDCA. In fact, the percentage of intervals in which any type of improvement is achieved reaches 41.17% and 64.54% in the presence and absence of DCF traffic, respectively. These enhancements are more meaningful when there are no stations without QoS support in the network, such enhancements being from 20% in 29.85% of the cases. In Figure 11 it can be seen that the dynamic model outperforms the remaining ones in almost all the tested scenarios. This results from properly choosing the prediction scheme that selects the most suitable parameter configuration at every moment.

The appropriate selection of the AIFSN combination and the CW size reduces the collisions in the network, which results not only in an increase in the voice + video normalized throughput but also in an improvement in the overall network performance (see Tables 13 and 14). The results indicate that



(a) Scenarios with EDCA and DCF stations

(b) Scenarios with EDCA stations

Figure 11: Voice + video norm. Throughput (dynamic adaptation).

the dynamic scheme reduces the number of collisions with regard to both EDCA and the models of the first phase (J48 and M5). However, this scheme obtains a greater number of collisions than the J48* and M5* approaches. This is due to the fact they penalize the real-time applications. The same reasoning applies to the enhancement of the global performance of the network.

The DCF traffic throughput has also been analyzed. As can be seen in Table 13, this throughput depends on its occupancy level in the network. If this value remains low, the performance of these stations is the same or better when using the dynamic model than when using EDCA. Nevertheless, if the network carries a considerable amount of DCF transmissions, the dynamic parameter combination allows it to transmit less DCF traffic than the remaining analyzed options. In particular, these last options damage heavily the performance of video applications. Therefore, the dynamic approach is logical, given that it seeks to improve the quality of the real-time transmissions.

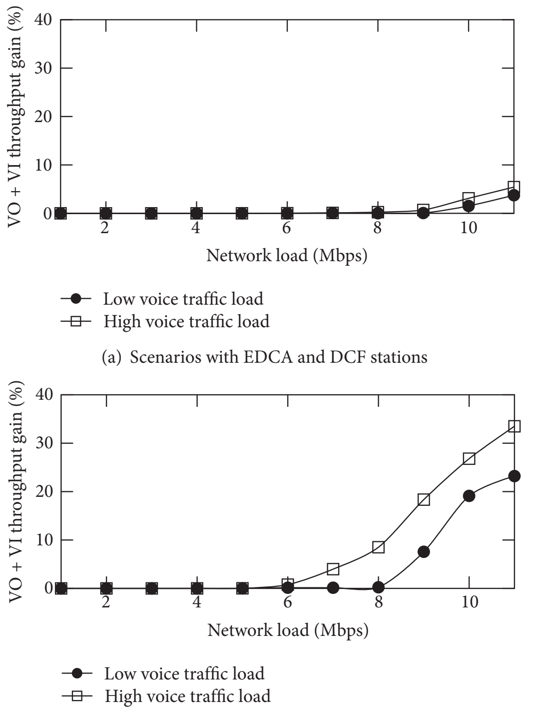
To conclude this analysis, the improvement in terms of voice + video throughput achieved by our scheme is evaluated. The improvement shown is based on the traffic load of the voice and video applications. To that end, both the voice and video traffic flows have been divided into two groups: the first group contains the results for the 10 scenarios that hold the lowest voice traffic load (called *low video traffic load*), while the second one, referred to as *high voice traffic load*, is made up of the 10 scenarios with the highest voice traffic load. The same description applies to the video applications. During the performance analysis it has been proved that the existence of stations that use the DCF function plays a decisive role in the network performance. Therefore, the results are

Table 13: Overview of the dynamic adaptation results (scenarios with DCF and EDCA stations).

	Standard	J48	J48*	M5	M5*	Dynamic
VO + VI Norm. Th.	0.9259	0.9440	0.9227	0.9438	0.9312	0.9512
Retrans. per Pkt	0.5447	0.4459	0.3636	0.4271	0.3473	0.4267
Global Norm. Th.	0.6014	0.6199	0.6395	0.6225	0.6399	0.6263
DCF Norm. Th.	0.5498	0.6790	0.7743	0.7118	0.8056	0.7496

Table 14: Overview of the dynamic adaptation results (scenarios with EDCA stations).

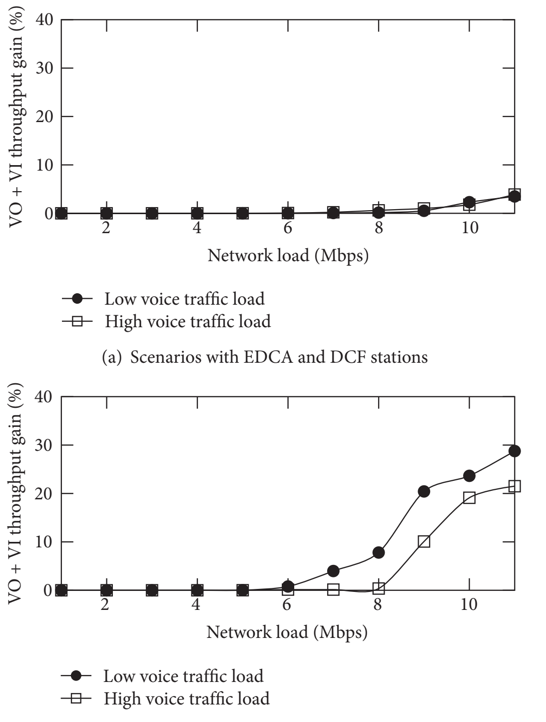
	Standard	J48	J48*	M5	M5*	Dynamic
VO + VI Norm. Th.	0.9259	0.9440	0.9419	0.9438	0.9596	0.9683
Retrans. per Pkt	0.4604	0.3428	0.2234	0.3329	0.2288	0.2247
Global Norm. Th.	0.6776	0.6965	0.7050	0.7010	0.7304	0.7331



(a) Scenarios with EDCA and DCF stations

(b) Scenarios with EDCA stations

Figure 12: Voice + video throughput gain achieved by the dynamic scheme on the basis of two different voice traffic loads.



(a) Scenarios with EDCA and DCF stations

(b) Scenarios with EDCA stations

Figure 13: Voice + video throughput gain achieved by the dynamic scheme on the basis of two different video traffic loads.

also presented according to the presence or absence of these legacy DCF stations in the wireless network. These results can be observed in Figures 12 and 13.

Figure 12 shows the performance gain for the two aforementioned voice traffic loads. In this figure it is clearly seen that the enhancement achieved is higher when the scenarios are only composed of stations that use EDCA, as depicted in Figure 13(b). In fact, as the network load increases, this enhancement becomes higher, being it from 30% in the case of a high voice traffic load. By contrast, in Figure 13(a) it is shown that the improvement made by our scheme is slightly smaller in the presence of legacy stations. This is because, when the network holds a high DCF traffic load, the proposed scheme usually uses the default values for the parameter set, as EDCA does. Moreover, it can be also observed that further improvements are done in scenarios in which a high voice traffic load is transmitted.

Similar results are found when analyzing the behavior of the two defined video traffic loads (see Figure 13). In the case of scenarios without stations that use DCF to access the channel, the enhancement achieved by our proposal is markedly increased. Moreover, when a high amount of video applications are transmitted in the network, the throughput improvement is much higher. Finally, notice that, in the analysis of the video applications, the differences found between a low and a high video traffic load are lightly smaller than in the voice transmissions case.

7. Conclusions

In this paper, we have proposed a new dynamic prediction scheme which aims to enhance the quality of voice and video transmissions over IEEE 802.11 WLANs. This scheme dynamically adapts the AIFSN combination and the CW size to optimize the access to the medium for stations that use EDCA, while ensuring compatibility with those that only support DCF. The proposal is composed of several predictive submodels made up of J48 decision tree classifiers and M5 regression models, and it is only used by the AP of the network. As a consequence, no changes need to be made to current commercial network cards. Furthermore, periodic communication of the medium access parameters is carried out through the use of beacon frames, avoiding in this way the introduction of additional control traffic into the network.

The results have proved that the proposal improves the EDCA capacities, boosting the performance of the multimedia communications by more than 20% in some scenarios. This improvement is achieved via both suitable separation of the AIFSN values and the appropriate selection of the CW size for each AC. Furthermore, this approach leads to a reduction in the number of retransmission attempts and contributes to enhancing the global throughput of the network.

Competing Interests

The authors declare that they have no competing interests.

Acknowledgments

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2.2. Wi-Balance: Channel-Aware User Association in Software-Defined Wi-Fi Networks

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Wi–Balance: Channel–Aware User Association in Software–Defined Wi–Fi Networks

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***Abstract*—In traditional 802.11 networks stations usually try to associate to the AP with the highest signal strength. However, especially in case of very dense deployments, this may lead to uneven wireless clients distribution, and thus to poor network performances. Software Defined Networking (SDN) has recently emerged as a novel approach for network control and management. In this paper we present *Wi–Balance*, a novel SDN–based solution for joint user association and channel assignment in Wi–Fi networks. An experimental evaluation in a real–world testbed showed that *Wi–Balance* outperforms the RSSI–based user association schemes in terms of throughput and channel utilization by up to 25% and 30%, respectively. We release the entire implementation including the controller and the data–path under a permissive license for academic use.**

***Keywords*—Software Defined Networking, IEEE 802.11, WLANs, channel assignment, mobility management**

I. Introduction

The past years have witnessed a sustained increase in mobile traffic demands that is forecast to reach 49 exabytes per month by 2021 [1]. Due to its low deployment and operational costs, Wi–Fi [2] has emerged as an efficient way to satisfy such demands. Originally relegated to residential and enterprise scenarios, Wi–Fi is becoming a viable traffic offloading solution for cellular networks. Nevertheless, its unplanned nature coupled with its contention–based channel access scheme lead to sub–optimal performances when the network density increases. Moreover, Wi–Fi networks operate in unlicensed bands as opposed to the licensed spectrum used by cellular networks. While this makes Wi–Fi networks extremely easy to deploy, it also makes them more vulnerable to interference from co–located deployments. The growing popularity of 5 GHz–capable devices is mitigating this issue in indoor settings, where the penetration through the walls of high frequency signals is limited. However this does not apply to outdoor scenarios or to networks in the 2.4 GHz band.

In addition to the mentioned pitfalls, Wi–Fi networks leave clients in charge of selecting the optimal Access Point (AP). The actual algorithm used by the clients for the AP selection is not specified by the standard and is left as implementation choice for the vendor. RSSI measurements are typically used to perform this operation, i.e. the client selects the AP with the highest RSSI. Such approach however does not consider the AP load and may lead to an uneven clients distribution across the network. Finally, only a limited number of channels are available in both the 2.4 GHz and the 5 GHz bands. As a result, a severe throughput degradation is expected when multiple APs are in the same collision domain, especially when the number of active APs per unit of area increases. Therefore, an effective collision domain isolation and channel assignment strategy becomes essential to ensure optimal performances [3].

In recent years different solutions have emerged to solve the aforementioned problems. Nevertheless, the traditional Wi–Fi architectures makes it hard to add new mechanisms without modifying the standard. Software Defined Networking (SDN) has recently emerged as a new way of refactoring network functions. By clearly separating data–plane from control–plane and by providing high–level programming abstractions, SDN allows to implement traditional network control and management tasks on top of a *logically* centralized controller. However, albeit SDN is already an established technology in the wired domain, with OpenFlow playing the role of de–facto standard [4], equivalent solutions for wireless and mobile networks have only recently started to appear [5], [6].

In this work we present *Wi–Balance*, a joint channel selection and user association scheme for Wi–Fi–based WLANs. Our contribution is two–fold. On the one hand, a constraint programming algorithm is designed to isolate possible collision domains among the APs. On the other hand, we present a user association scheme capable of detecting situations in which the traffic is not efficiently distributed and to transparently reschedule to other APs the clients whose transmissions are causing performance issues. Based on a real–world evaluation we have demonstrated an improvement of up to 25% and 30% in terms of network throughput and channel utilization compared to a standard RSSI–based user association mechanism. We release the entire implementation, including the controller and the data–path, under a permissive APACHE 2.0 license[1] for academic use.

The rest of the paper is organized as follows. In Sec. II we present the related work. The proposed user association and channel selection scheme is described in Sec. III. Section IV provides the implementation details. Section V reports the measurements campaign. Finally, Sec. VI concludes the paper and discusses the future work.

[1]Online resources available at: http://empower.create-net.org/

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II. Related Work

The amount of literature on user association mechanisms in WLANs is significant. The majority of the works in this domain set to achieve some of the following targets: (i) minimize the number of stations per AP; (ii) maximize the average signal quality; or (iii) maximize the average throughput of the network. Moreover, according to the entity responsible for these tasks, the approach may be distributed or centralized.

Regarding the first described target, authors in [7] propose an algorithm to balance the network load that aims to minimize the number of stations per AP based on the signal strength. A similar approach is followed in [8]. However, the number of attached clients alone is not an accurate estimator of the workload of an AP since traffic conditions may significantly vary among the stations.

Selecting the target AP according to the signal strength may lead to ping–pong effects, which are even more difficult to handle in the absence of communication and coordination among the APs. To address this problem, in [9] a station periodically looks for the most suitable AP in terms of both traffic load and RSSI level. However, the handover is not performed until a given AP is not identified as the best choice for *n* consecutive times. The signal perceived by the stations is also taken into account in [10], where two wireless adapters are used at the client side to simultaneously allow data exchange with the AP and channel monitoring. The AP selection is also performed at the stations side in [11]. Although these approaches aim to improve independently the throughput of each station, they do not consider the network–wide performance.

In [12] the average workload of the network is used to redistribute the traffic when a new station joins the network or when the signal quality of a client deteriorates. The proposed approach however requires changes to the standard beacon frames and is thus hardly a practical choice. A similar scheme is presented in [13] where the stations are migrated to the least loaded AP. Nevertheless, since the channel quality is not considered, this approach may significantly reduce the aggregated throughput of the network.

In [14] a distributed algorithm is run on the APs, which makes use of an RSSI threshold to take handover decisions. In [15] the problem is addressed using a Mixed Integer Non Linear Programming (MINLP) problem formulation by taking into account the differences among the bandwidth demand of the users. An analytical model is also introduced in [16] with the novelty of assessing the Enhanced Distributed Channel Access (EDCA) parameters defined in 802.11e [17], along with the load–balancing problem. Lastly, a mixture of several parameters such as RSSI level, users location and link quality are considered for the association process in [18].

A handover usually leads to a re–association process, which in time can generate performance degradation due to the period needed by the station to reconnect to the target AP. In [19] the RSSI of the clients drives the association decision. However, the APs are required to operate on non–overlapping channels limiting the size of the deployment. A mechanism is proposed in [20] to set a different channel for each AP.

The client association problem is also studied from the point of view of the Software–Defined WLANs. In [21] the authors formulate the problem through a Markovian analytical model with the aim of minimizing the interpacket delay. In [22] an SDN–based scheme is proposed to reconfigure the transmission power of the APs when the controller detects that the load distribution is unbalanced. This, in time, forces the stations to perform a handover. However, this proposal uses a fixed relationship between transmissions bitrate and Signal–to–Noise Ratio. Moreover, the results are only shown via simulation. Mininet is used in [23] to test an algorithm where the SDN controller compares the load of each AP with a fixed value in order to decide whether to accept or reject new stations. The user association problem is modelled using graph theory in [24]. To make the association decision the channel busyness time and the interference are considered. However, the APs are also required to operate on the same channel.

In [25] the authors present the concept of virtual resource chain, which refers to all the resources in a WLAN, to improve the resource utilization and the network balance. A similar work is presented in [26], where a Mixed Integer Linear Programming (MILP) model is designed to maximize the total bandwidth assigned over different connection modes, i.e. 2.4 GHz Wi–Fi, 5 GHz Wi–Fi and Ethernet. A real implementation is proposed in [27], [28] to enable the balance over multiple channels by building on the use of virtual access points. In the first work the handover decision is based on the maximum and minimum traffic load of the APs and the RSSI perceived by the stations. A sniffer interface is also used in the second approach to gather the network statistics through a periodical scanning of the channels.

An effective user association scheme must consider the global network status and evaluate different load metrics in order to ensure optimal performances. Since interference is a determinant issue, a channel assignment procedure must be also performed along with the user association algorithm. Finally, the stations redistribution over the APs must not lead to a transmission interruption. To the best of the authors knowledge, *Wi–Balance* is the first scheme which supports all the above mentioned requirements and that can be implemented with no changes to the Wi–Fi standard.

III. Channel–aware user association

In this section we introduce the main features of the *Wi–Balance* channel–aware user association solution. Moreover, based on a preliminary analysis, the most determining factors in multichannel user association are identified and used to motivate this work.

A. Motivation

Interference and collisions are the most important cause of performance degradation in WLANs [29], [30]. When several clients attached to the same AP transmit at the same time, the network may suffer delays, service interruptions and

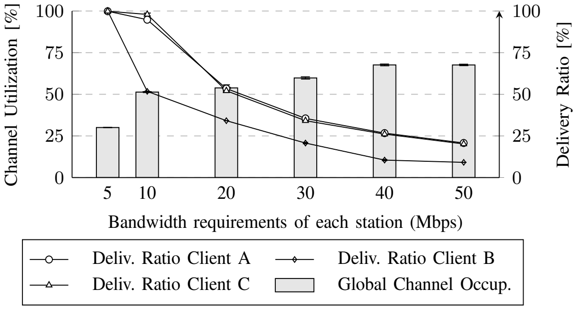

Fig. 1: Delivery ratio of three stations attached to a single AP performing uplink transmissions with different bandwidths.

Algorithm 1 Channel assignment procedure

Input:

$neighbors$: graph storing the neighbours of each AP.
$channels$: list of available channels.
$overlaps$: dictionary storing the overlapping channels.

Output:

$assignment$: dictionary of (AP, channel) assignment

procedure SOLVE($neighbors, channels, assignment$)
  $remainingAPs \leftarrow APs \notin assignment$
  if $len(remainingAPs) == 0$ **then**
    return $assignment$ ▷ It becomes the ***solution***
  Sort $remainingAPs$ by the lowest number of available channels and the highest number of neighbors in $assignment$
  $nextAP \leftarrow remainingAPs[0]$
  $possibleCh \leftarrow channels$
  for each $AP \in neighbors\,[nextAP]$ **do**
    $APCh \leftarrow assignment[AP]$
    $possibleCh \leftarrow possibleCh - APCh$
    $possibleCh \leftarrow possibleCh - overlaps[APCh]$
  if not $possibleCh$ **then**
    $possibleCh \leftarrow min(assignment)$
  for each $channel \in possibleCh$ **do**
    $assignment[NextAP] \leftarrow channel$
    return SOLVE($neighbors, channel, assignment$)

performance drops. Figure 1 depicts the relationship between channel utilization and network performance. During the measurement three clients were transmitting with bandwidth requirements ranging from 5 to 50 Mbps towards the same AP. As can be seen, when the channel occupancy is higher than 60%, the delivery ratio dramatically drops. This is due to the collisions in the wireless medium and the decrease in the data rates used for the transmission. We remind the reader that the Modulation and Coding Scheme (MCS) adaptation algorithms tend to select lower data rates upon several failed transmissions, which in time increases the channel utilization.

This simple scenario demonstrates the importance of an efficient network resource allocation in terms of both channel assignment and user association. This aspect acquires even more relevance when considering mobile clients. In order to address this challenge we propose an SDN–based joint user association and channel assignment algorithm.

B. Channel Assignment Algorithm

Channel assignment must be done in such a way to minimize interference between APs that are in the same collision domain. Two APs are in the same collision domain if they are tuned on the same channel and if they are within carrier sensing range of each other. In this case, if multiple transmissions start at the same time they can either collide or one of the transmissions must be delayed. In either case a reduction in the aggregated network throughput is to be expected.

The efficiency of a channel assignment procedure depends on the number of available channels and on the number of APs in the same collision domain. The higher the number of available channels, the lower the probability of finding two APs using the same one. Therefore, it is crucial to identify the channels used by the APs in the neighbouring networks, since they may share the same collision domain. However, after identifying these channels, the set of available channels for the assignment may be very limited, especially in congested areas such as office buildings or universities.

A channel assignment algorithm must have as input the interference map of the WLAN. In other words, it must consider for each AP, the set of surrounding APs that must not operate on the same channel, as well as the list of available channels. The interference map is built in the first step of the algorithm and its data is designated as the constraints of the problem. Moreover, a periodic analysis of the wireless medium must be carried out to update the network information. Notice that SDN–based solutions allow the channel assignment algorithm to have a complete view of the network (which is collected and maintained by the SDN controller).

In light of this, a constraint programming algorithm has been designed to solve the channel assignment problem. The recursive algorithm is shown in Algorithm 1. The algorithm first tries to assign a channel to the set of APs with the lowest number of available channels. We refer to available channels as those that have not been still assigned to the neighbouring APs of a certain AP and do not overlap with the ones already assigned to them. Then, the algorithm selects in this set of APs the one with the highest number of neighbours already assigned. Furthermore, if all the channels have been already taken by the neighbouring APs, the algorithm selects the channel that has been used by the lowest number of APs. In case that multiple channels match this condition, the channel with the lowest occupancy ratio is chosen. The algorithm finishes when it finds a configuration that minimizes the number of APs in the same collision domain.

Although after performing an efficient channel assignment the network interference may have been significantly reduced, there is still room for improvement. In the next section we will introduce the *Wi–Balance* user association algorithm.

C. User Association Algorithm

After the channel assignment, the controller performs a neighbour discovery process in order to build the channel quality map. This map includes for each station the channel quality with respect to all the APs in the network. The channel quality map is built by the SDN controller by retrieving from each AP the list of stations in its coverage area. Similarly, the controller periodically gathers the statistics of the rate adaptation algorithm maintained by each AP. In particular, for each station and for each supported MCS, the Exponentially Weighted Moving Average (EWMA) of the delivery probability and the expected throughput in the last observation window are reported. Moreover, the number of successful and failed transmissions are also reported. We remind the reader that this information is maintained by the rate adaptation algorithm implemented by the AP. Therefore, no extra computation is added to the APs. Gathering this statistical data needs some limited signalling between the controller and the APs. The details of this protocol are outside the scope of this paper and can be found online [31]. It is also important to highlight that *Wi–Balance* does not require any change to either to the IEEE 802.11 protocol nor to the wireless devices. The whole process is sketched in Fig. 2.

Let us define U as the set of stations in the network, M as the set of Wi–Fi APs and $\Omega(u) \subseteq M$ as the set of APs within the coverage area of the user $u \in U$. Using the statistical data collected by the controller, *Wi–Balance* computes the channel utilization $\mu(n)$ for each $n \in M$ and the average channel occupancy across all the APs in the network $\overline{\mu}$. If a significant difference between $\overline{\mu}$ and any occupancy ratio is found a user re–association process is triggered for the affected AP. In particular, *Wi–Balance* collects, for each user u attached to the affected AP n, the channel utilization of the surrounding APs, $\Omega(u)$, and the RSSI level between each AP $m \in \Omega(u)$ and the station u, let us call this quantity R_u^m.

After that, *Wi–Balance* selects as candidate AP for the handover the AP offering the lowest result of the product between the current occupancy ratio of AP n, i.e. $\mu(n)$, and the perceived signal strength R_u^m for each $m \in \Omega(u)$. Then, the client handover is performed. The average channel occupancy $\overline{\mu}$ is recalculated to check if the network redistribution was efficient. Otherwise, the handover is reverted. This process is also triggered in case of observing a sudden change in the RSSI value for any client, which could result from the movement of that client.

D. Complexity Analysis

In this section we will analyse the computational complexity of *Wi–Balance*, distinguishing between the channel assignment and the user association algorithms.

The channel assignment algorithm is a recursive procedure that is called n times until a channel has been selected for each AP. The recursive nature makes the algorithm have two cases: a base and recursive case. In order to solve this problem, we will use a recurrence relation denoted as $T(n)$. The base case encompasses the scenario in which all the APs have been visited, and thus, $n = 0$. At this point, the complexity of $T(0)$ is essentially constant and equals to $O(1)$. In the recursive case, i.e. when $n > 0$, two aspects must be considered: i) the function is recursively called with $n-1$; ii) the channel search operations are internally performed for that AP. The cost of (i) is $T(n-1)$, while the cost of (ii) must be further explored. First, the n remaining APs are sorted by the lowest number of available channels and the highest number of neighbours. The complexity of this step is $O(n \cdot log(n))$. Then, the list of neighbouring APs for the first AP in the list is traversed to discover the available channels, which results in a cost $O(n)$. In the worst case in which there are no available channels, the algorithm will select the channel less used by the neighbours, hence adding a complexity $O(n)$. After that, the algorithm must iterate through the list of possible channels, which in the worst case will be as long as n. On this basis, the cost of (ii) is estimated as $O(n)$, and hence the relation $T(n)$ can be expressed as $T(n) = T(n-1) + O(n)$. Thus, the complexity of the channel assignment is $O(n^2)$.

Every time the user association algorithm is called, the list of APs must be traversed to compute their channel occupancy ratio. Therefore, the complexity of this operation is $O(n)$. Computing this ratio requires to calculate the fraction of time used by the stations attached to each AP. In the worst case, all the stations in the network, s, will be attached to the same AP, which results in a computational complexity $O(s)$. On this basis, the cost of computing the channel utilization will be as high as $O(n \cdot s)$. Moreover, the average channel utilization must be calculated. Notice that this estimation depends on the number of APs, hence it being as complex as $O(n)$. Then, the individual ratio of each AP must be compared with the average one to find imbalances in the distribution of the network load. Therefore, the list of APs must be once again traversed, resulting in a complexity of $O(n)$. In case of finding an imbalanced AP, the list of all its clients must be traversed, and for each client, the algorithm must iterate through all its possible APs to perform a handover. Thus, the complexity of the use association algorithm is $O(n \cdot s)$.

Finally, the overall computational complexity of the joint channel assignment and user association algorithm is $O(n \cdot s + n^2)$ which can be approximated as $O(n \cdot s)$ since in most cases $s >> n$.

IV. Implementation Details

A. Overview

The proposed user association algorithm has been implemented on the 5G–EmPOWER platform [5]. 5G–EmPOWER is a Multi–access Edge Computing Operating System (MEC–OS) which converges SDN and NFV into a single platform supporting lightweight virtualization and heterogeneous radio access technologies[2]. A high level view of the the 5G–EmPOWER MEC–OS architecture is sketched in Fig. 3.

The 5G–EmPOWER MEC–OS consists of a hardware abstraction layer converging several radio access networks

[2]Online resources available at: http://empower.create-net.org/

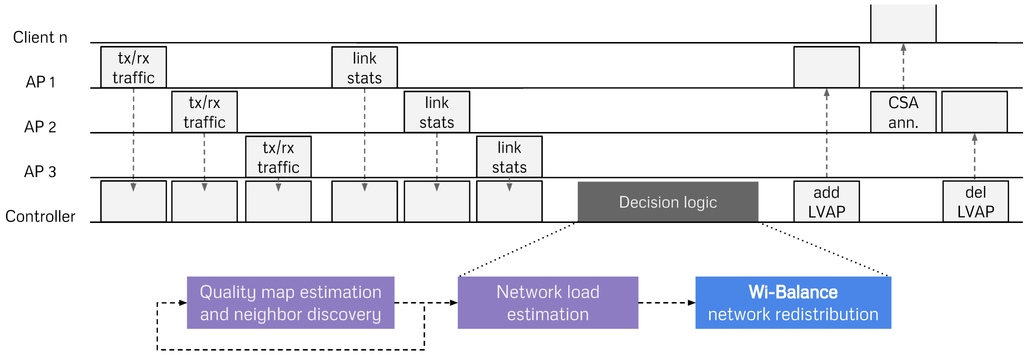

Fig. 2: Scheme of the working mode of *Wi–Balance*.

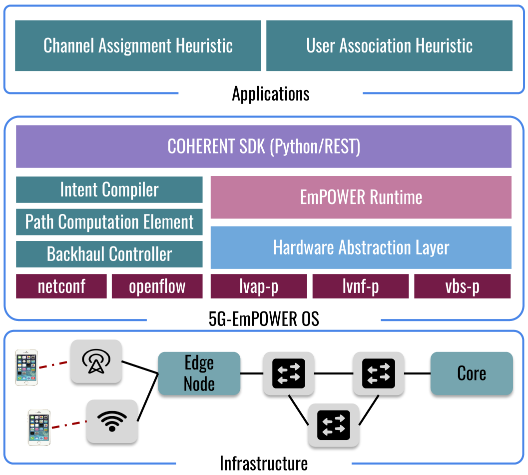

Fig. 3: The 5G–EmPOWER MEC–OS System Architecture.

control and management protocols into a unified set of abstractions that are then exposed to the application layer. Such abstractions allow the applications layer to implement joint NFV and SDN resource management operations. This includes, for example, joint mobility management and NFV placement/migration schemes as well as radio access and backhaul load–balancing. The 5G–EmPOWER MEC–OS currently supports Wi–Fi and LTE radio access nodes. Interaction with SDN–based backhauls is enabled trough an Intent–based networking interface. In the rest of this section we will provide a short summary of the *Light Virtual Access Point* and of the *Network Graph* abstractions used to implement the user association algorithm presented in this paper. For a more extensive description we refer the reader to [5].

B. Light Virtual Access Point (LVAP)

The *LVAP* abstraction [6] provides a high–level interface for the state management of the wireless clients. The implementation of such an interface handles all the technology–dependent details such as association, authentication, handover and resource management. A client attempting to join the network will trigger the creation of a per–client virtual access point (the *LVAP*) which becomes a potential candidate AP for the client to perform an association. Similarly each AP will host as many *LVAPs* as the number of wireless clients that are currently under its control. Removing an *LVAP* from an AP and instantiating it on another AP effectively results in a handover.

C. Network Graph

The Network Graph provides network programmers with a full view of the network state. The network graph is exposed as a directed graph $G = (V, E)$ where V is the set of clients and radio access network elements (i.e. the Wi–Fi APs) and E is the set of edges or links. A weight $\omega_e(e_{n,m})$ is assigned to each link $e_{n,m} \in E : \omega(e_{n,m}) \in \mathbb{R}$. Another weight $\omega_v(n)$ is assigned to each node $n \in N : \omega_v(n) \in \mathbb{R}$. The weights assigned to nodes and links can model different aspects of the wireless system. In the current implementation of the 5G–EmPOWER MEC–OS the following types of complex data structures can be associated to the vertexes and the edges of the Network Graph:

- *RSSI*. The received signal strength indicator as reported by the Wi–Fi APs (uplink direction) and wireless clients (downlink direction). Measurements in the downlink direction are taken using the radio resource management features introduced by the 802.11k amendment [2].
- *Rate Control Statistics*. The statistics of the MCS selection algorithm at the AP (downlink). For each supported MCS, the frame delivery ratio and the estimated throughput in the last observation window are reported. Historical, EWMA–filtered values, are also available.
- *Channel Occupancy*. The fraction of the time the channel is busy at each Wi–Fi AP. This is an estimated value computed using the rate control statistics and by sniffing the transmissions within the decoding range of the AP. Corrupted frames are however not taken into account.
- *Traffic Matrix*. The number of packets and bytes transmitted/received by each wireless client. The absolute packets/bytes values as well as the bitrate in the last observation window are available to applications.

D. Seamless Handover Across Different Channels

Notice how the original seamless handover enabled by the *LVAP* concept does not work when the APs operate on different channels. In this work we remove this limitation by using the Channel Switch Announcement (CSA) defined by the IEEE 802.11 standard. The CSA procedure was originally designed to allow APs to inform the attached client that the operating channel of the hotspot was about to change. This information is delivered inside the standard beacons frames. In particular, an AP that is planning to switch the operating channel will start advertising the new channel in its beacons. A countdown is started and the channel is switched after a configurable number of beacons (three usually).

In traditional Wi–Fi networks, beacons are sent as broadcast management frames. Conversely, in our case each *LVAP* sends its own beacons using unicast frames. This is possible because an *LVAP* is created for each station attached to an AP. Such a design choice allows us to target a CSA message to a particular station by enabling it only for the *LVAP* that was created for that station. The seamless handover across APs tuned on different channels and/or bands is enabled by first creating an *LVAP* on the target AP. This *LVAP* is initially inactive since the station that it is mapping is tuned on a different channel. Then the controller instructs the *LVAP* on the source AP to start a CSA procedure. At the end of this CSA procedure the *LVAP* at the source AP is automatically removed. In the meantime, the station will have switched channel and will have found its *LVAP* on the target AP. The full process is sketched on the right–hand side of Fig. 2.

It should be also noted that the performance impact of these unicast beacons is very low given their short duration and length. However, a trade–off can be set between the duration of the handover and the number of beacons in the network. If this feature is disabled, the impact on the network will be decreased at the price of a longer period of time to perform the handover. If it is enabled, a faster handover is possible at the price of a little increase in the management traffic.

V. Performance Evaluation

In this section we report on the results of the performance evaluation. In particular we compare the network performance using *Wi–Balance* with the network performance using an RSSI–based user association algorithm.

A. Evaluation Methodology

The performance evaluation is carried out on a real–world testbed composed of three APs. The layout of the testbed is depicted in Fig. 4. The APs are built upon the the PCEngines ALIX 2D (x86) processing board and run OpenWRT 15.05.01. The Wi–Fi cards are based on the Atheros AR9220 chipset. All the experiments are carried out on the 5 GHz frequency band using the IEEE 802.11n physical layer [32]. The channels used by the APs are selected by the channel assignment algorithm presented in Sec. III-B. The scenario also comprises the 5G–EmPOWER controller (not shown in the picture) and a set of 10 stations. One of these stations moves following the path marked in blue in Figure 4. The remaining stations are static and are deployed randomly across the entire floor. Dell–branded laptops powered by an Intel i7 CPU and running Ubuntu 16.04.02 are used as wireless clients. It should be noted that our solution can be applied to other scenarios in the 2.4 GHz band and including both uplink and downlink traffic, as well as different number of stations and APs.

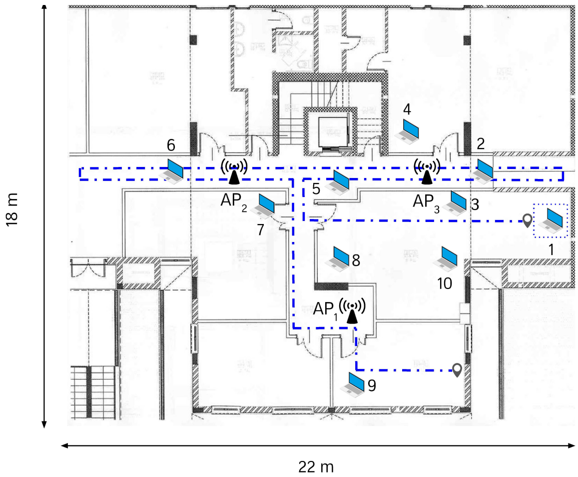

Fig. 4: Testbed deployment layout and APs–users distribution.

TABLE I: Configuration of the measurements campaign.

Test	Traffic type	User groups traffic dist.	Bandwidth (Mbps)
A	UDP	Constant - Intermittent	10
B	UDP	Constant - Intermittent	5
C	TCP	Constant - Intermittent	-
D	UDP	Intermittent - Constant	10
E	UDP	Intermittent - Constant	5
F	TCP	Intermittent - Constant	-
G	UDP	Constant - Constant	10
H	UDP	Constant - Constant	5
I	TCP	Constant - Constant	-

Nine experiments, identified with the letters from A to I, have been conducted. Each test has a duration of 5 minutes and consists of a single UDP or TCP stream between wireless clients and a server sharing the same backhaul with the APs. In the case of UDP traffic, different bitrates are used. The set of 10 users is divided into 2 groups with 5 stations each for the tests from A to F. The first group performs transmissions with a constant bitrate that is maintained for the entire duration of the measurement. By contrast, the second group uses intermittent transmissions. These stations transmit traffic for 40 seconds, and after that, they stop the transmission for 20 seconds. This pattern is repeated until the end of the experiment. Then, the role of the groups is inverted, i.e. the stations with constant bandwidth perform intermittent transmissions, and vice versa. In the experiments from G to I, all the stations generate constant bitrate traffic. A summary of the different scenarios can be found in Table I.

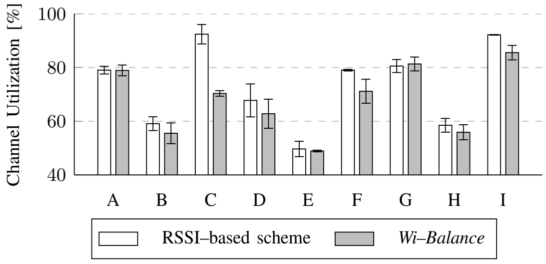

Fig. 5: Network–wide channel utilization for both the UDP and the TCP traffic transmissions.

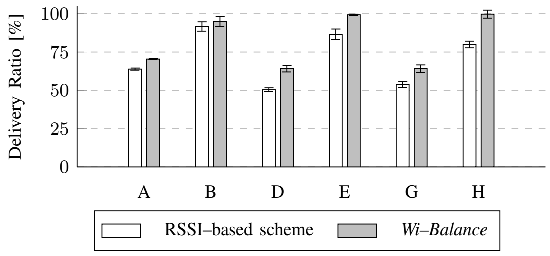

Fig. 7: Average delivery ratio for the UDP traffic transmissions at 5 and 10 Mbps.

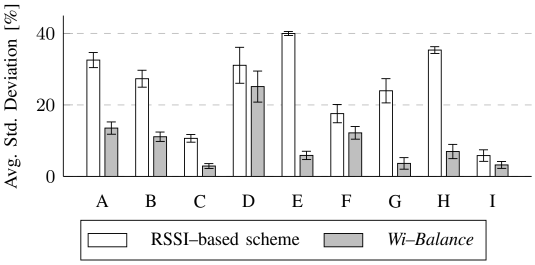

Fig. 6: Average deviation of the channel utilization of each AP with regard to the network–wide ratio for both the UDP and the TCP traffic transmissions.

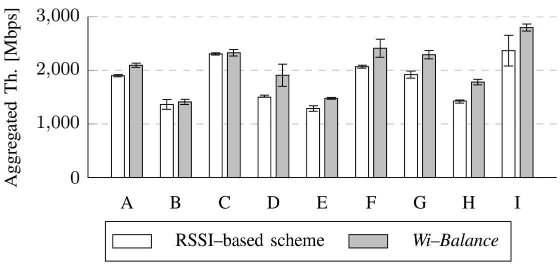

Fig. 8: Network–wide aggregated throughput for both the UDP and the TCP traffic transmissions.

The effectiveness of our proposal is compared with the RSSI–based scheme, in which the stations intend to associate to the AP providing the strongest signal. As evaluation metrics we have considered the delivery ratio, the aggregated throughput, the wireless channel utilization, the Jain's fairness index [33] and the retransmission ratio. Apart from the uplink transmissions, no downlink traffic exists between the APs and the stations.

B. Experimental Results

Especially in situations of congestion, an uneven distribution of the stations may cause some of the APs to be saturated, while some others are idle. As a consequence, the users connected to first group of APs will share the available bandwidth which in time could result in a lower aggregated network throughput compared to a situation with an even distribution of the stations across the various APs. From the results shown in Fig. 5 it can be observed that the average channel occupancy ratio with *Wi–Balance* is up to 30% lower than the channel occupancy ratio with the RSSI–based scheme. This is achieved through a more efficient users distribution, which results in a more balanced network and a decrease in the channel contention.

In addition to reducing the overall channel utilization, it is even more important that the APs have an occupancy ratio that is as similar as possible. This situation is displayed in Fig. 6, where the average deviation of the channel utilization of each AP with regard to the average network–wide ratio using *Wi–Balance* is compared with the channel utilization obtained using the RSSI–based scheme. As can be seen, the utilization of each AP widely differs for the reference scheme, while this ratio is more balanced in the case of *Wi–Balance*.

Figure 7 plots the delivery ratio achieved in the tests using UDP traffic. It can be seen that in all the experiments *Wi–Balance* outperforms the results obtained by the RSSI–based scheme by an average of 17%, and up to 25% in the experiments D and H. The network–wide aggregated throughput is presented in Fig. 8 for the UDP and TCP traffic. The figure shows that the efficient scheduling of the stations leads to an increase in the throughput by an average of 16% and up to 25% in the scenarios D and H.

In addition to enhancing the performance, an efficient load–balancing algorithm must distribute the bandwidth evenly among the stations. To demonstrate this effect, Fig. 9 compares the Jain's fairness index of the stations throughput using *Wi–Balance* and the RSSI–based scheme. As can be seen, *Wi–Balance* delivers a better fairness in all the experiments.

Wi–Balance performs better than the RSSI–based scheme also for the mobile users, as can be observed in Fig. 10. This is because when a station moves over the coverage area, the AP to which it is connected is not chosen only according to the signal strength, on the contrary also the AP traffic load is considered. For this reason, the throughput improvement

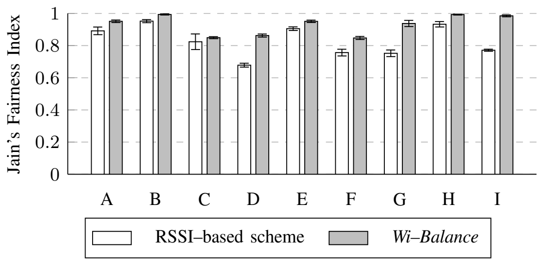

Fig. 9: Jain's fairness index of the throughput achieved by all the wireless clients for both the UDP and the TCP traffic transmissions.

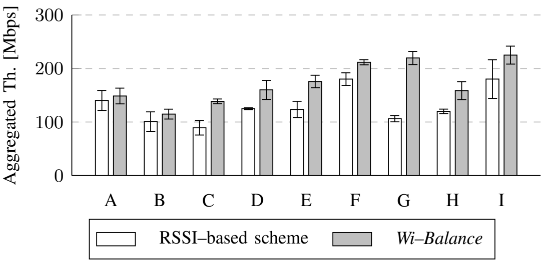

Fig. 10: Average throughput achieved by the mobile station for both the UDP and the TCP traffic transmissions.

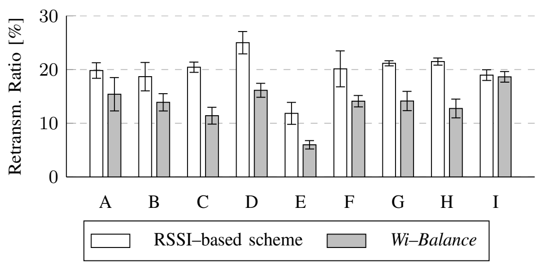

Fig. 11: Network–wide average retransmission ratio for both the UDP and the TCP traffic transmissions.

is notably higher for the mobile users. Finally, the efficient usage and scheduling of the network resources makes also possible to enhance the network reliability. Since *Wi–Balance* results in a more uniform wireless client distribution, the retransmission ratio is also decreased by an average of 30%. This phenomenon is displayed in Fig. 11.

VI. Conclusions

In this paper we presented *Wi–Balance*, a novel SDN–based solution for joint user association and channel assignment in WiFi networks. Moreover, we also introduced a seamless handover mechanism for Wi–Fi networks capable of operating in a multi–channel environment.

The performance of *Wi–Balance* has been evaluated in a real–world testbed under different scenarios considering mobile and static users. More specifically, compared to RSSI–based user association schemes, *Wi–Balance* can reduce the channel utilization by up to 30% and can improve the aggregated network throughput by up to 28% without penalizing the network fairness. Conversely a slight improvement in the Jain's fairness index can be noticed when using *Wi–Balance*.

As future work we aim to extend *Wi–Balance* to consider the wired backhaul in the user association algorithm. Moreover, we plan to support the traffic prioritization and aggregation features supported by the 802.11e and 802.11n standards.

Acknowledgements

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2.3. Programming Abstractions for Wireless Multicasting in Software-Defined Enterprise WLANs

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Programming Abstractions for Wireless Multicasting in Software–Defined Enterprise WLANs

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***Abstract*—The increasing demand for multimedia content and for live broadcasting is bringing renewed interest in multicast applications. In many cases, users access such streams using Wi–Fi networks. However, multicast over Wi–Fi poses several challenges including low–data rates and coexistence issues with regard to other unicast streams. Software Defined Networking (SDN) has recently emerged as a novel approach to network control and management. In this paper we present *SDN@Play*, a novel SDN–based solution for multicast rate–adaptation in Wi–Fi networks. The solution builds upon a new abstraction, named *Transmission Policy* which allows the SDN controller to reconfigure or replace a certain rate control policy if its optimal operating conditions are not met. An experimental evaluation carried out over a real–world testbed shows that this approach can deliver an improvement of up to 80% in terms of channel utilization compared to legacy 802.11 multicast. We release the entire implementation including the controller and the data–path under a permissive license for academic use.**

***Keywords*—WLANs, IEEE 802.11, multicast, rate adaptation, software defined networking.**

I. Introduction

Multimedia content delivery has witnessed a dramatic increase in popularity in the last decades. The growth in usage of platforms like YouTube and Netflix is a clear statement in support of this trend. Multicast transmissions are a particular use case within the generic multimedia content delivery domain where the same content is to be delivered to multiple destinations or receptors. Common examples of multicast applications are live broadcasting, online courses and tutorials, and multiplayer gaming. Due to the popularity of such applications in both the home and the enterprise networking domains, it is of capital importance to properly support them on 802.11–based WLANs (which is the most popular Wireless LAN technology). However, in addition to the challenges raised by multicast transmissions in the wired domain, such as routing and group management, WLANs pose a completely new set of difficulties.

WLANs based on the 802.11 family of standards dynamically choose among differed modulation and coding schemes (MCS) for frame transmission. For example, in the case of 802.11a/g networks, devices can choose bit–rates varying from 1 to 54 Mb/s, while in the case of the 802.11n/ac networks higher throughput MCSes are also available. This process, known as rate adaptation, is however restricted to unicast frame transmissions. This is due to the fact that, 802.11 uses a two–way handshake protocol where each data transmission must be acknowledged by the receiver. However, in the case of multicast transmissions, acknowledgements cannot be used as they would inevitably collide at the transmitter. As a result, multicast transmissions are usually performed at the lowest MCS (in order to increase both the range and reliability of the transmission) and do not use any form of transmission feedback mechanism. This has several drawbacks: (i) it severally limits the throughput for multicast transmissions, (ii) it consumes a significant portion of the available airtime affecting also the capacity available to other (unicast) flows, and (iii) given that multicast frames cannot be retransmitted, the reliability of the multicast streams can be adversely impacted.

Software Defined Networking (SDN) has recently emerged as a new way of refactoring network functions. By clearly separating data–plane from control–plane and by providing high–level programming abstractions, SDN allows to implement traditional network control and management tasks on top of a *logically* centralized controller. However, albeit SDN is already an established technology in the wired domain, with OpenFlow playing the role of de–facto standard [1], equivalent solutions for wireless and mobile networks have only recently started to appear [2], [3].

The contribution of this paper is twofold. First, we introduce a new programming abstraction for multicast communications. Second, we use such an abstraction to implement *SDN@Play*, an SDN multicast rate adaptation scheme for 802.11–based WLANs. The proposed solution allows utilizing higher bitrates for multicast transmissions while preserving the reliability of the communication. Based on a real–world testbed evaluation we have been able to demonstrate an improvement of up to 80% in terms of channel utilization compared to standardized multicast schemes for 802.11 networks. We release the entire implementation, including the controller and the data–path, under a permissive APACHE 2.0 license[1] for academic use.

The rest of this paper is structured as follows. In Sec. II we discuss the related work. We delve into the *SDN@Play* design in Sec. III, whereas in Sec. IV the implementation details are presented. Section V describes the evaluation methodology and discusses the results of the measurements. Finally, Sec. VI draws the conclusions pointing out future work.

[1] Available at: http://empower.create-net.org/

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II. Related Work

In this section we shall first provide a short background on multicast communications in IEEE 802.11–based WLANs. Then, we will review the most relevant related work highlighting our technical contributions.

Multicast communications are an efficient way to send the same information to many clients. In fact, by leveraging on the broadcast nature of the wireless medium, it is possible to deliver the same frame to multiple wireless terminals instead of transmitting it individually to each of them. However, according the 802.11 standard, multicast frames are never retransmitted nor acknowledged. As a consequence, transmission reliability is highly reduced. Moreover, the lack of feedback information makes it impossible to adapt the transmission data rate, hence being the basic rate used instead.

The IEEE 802.11aa amendment has been introduced to improve multicast communications performance while keeping the compatibility with current devices. The amendment improves the multicast frame transmission reliability by introducing the Group Addressed Transmission Service. This service specifies several retransmission policies and is composed of two different mechanisms: Direct Multicast Service (DMS) and Groupcast with Retries (GCR). In DMS mode each multicast frame is converted into as many unicast frames as the number of receptors in the multicast group. Each unicast frame may be retransmitted as often as necessary until the Access Point (AP) receives the ACK or the retransmission counter reaches its limit. In spite of ensuring high communication reliability, the DMS mode does not scale with the number of receptors in the multicast group.

GCR is a flexible service composed of three retransmissions methods: Legacy Multicast, Unsolicited Retries (UR) and Block ACK (BACK). The Legacy Multicast mode is the one defined in the original IEEE 802.11 standard. The UR policy specifies a number of retry attempts, N, in a manner that a frame is transmitted $N + 1$ times. In this way, the probability of a successful transmission is increased. However, UR may unnecessarily retransmit frames, hence increasing the overall network utilization. In BACK mode the AP reaches an agreement with the multicast receptors about the number of consecutive unacknowledged frames. After that, the AP sends a burst of multicast packets up to that number and requests a Block ACK from each receptor. Both this request and the corresponding ACKs are sent in unicast mode. Despite the control traffic overhead is reduced, also this approach does not scale with the number of receptors in the group. A comprehensive description of the various multicast schemes supported by the 802.11 standard can be found in [4].

Multicast rate selection may be achieved by defining feedback gathering mechanisms allowing the transmitter to discover the wireless medium status. Leader-based schemes are the most common proposals in the literature. LBP [5] aims at improving multicast communications by enabling ACKs. For this purpose, the receptor exhibiting the worst signal quality is selected as a leader and is in charge of sending ACKs. However, a procedure for the leader selection is not provided. Meanwhile, ARSM [6] divides its operation mode into two phases: in the first one, the group leader is selected, whereas in the second step the Signal-to-Noise Ratio (SNR) derived from the leader ACKs is used to adapt the transmission rate. H-ARSM [7] is an evolution of ARSM for hierarchical video transmissions over WLANs that ensures a minimum quality of the video sequence for all the receptors. The rate adaptation based on the SNR is also used in SARM [8]. In this scheme, the AP identifies the worst receptor by sending beacon frames to which the stations must reply indicating their own SNR. After that, the APs must let the remaining stations know about the new situation. However, changes at the client side are needed in order to implement this scheme. In this same category, mechanisms based on the frame delivery probability estimation have been also proposed [9].

Quality of Experience (QoE) has often been used as basis for rate adaptation in multimedia applications. In [10] a neural network is designed to build a model that maps QoE measurements into MCSes. PSQA [11] is developed as a hybrid objective–subjective metric that simulates how humans perceive impairments to video transmissions. A similar consideration can be made for [12]. In [9] authors address the multicast video delivery using a real–life testbed. In this solution the time is split into a transmission and a polling period. During the transmission period, stations collect the received sequence numbers. After that, APs gathers that information to calculate the link delivery probabilities. The transmission rate is then selected by comparing these values with the values obtained from the two previous rounds. However, changes at the client side are needed.

Multicast is not the only strategy to improve video delivery over wireless networks. For example in [13] dynamic channel switching is used in order to ensure that wireless video streaming takes place over the channel whose condition is most likely to provide a good video quality. Equally important are the evaluation methodology–focused works. For example in [14] the authors evaluate the effectiveness of streaming video over wireless LANs using the H.264 codec. The study concludes that streaming video content over 802.11n is a viable option and that perceptual quality of video is affected by the amount of background traffic and the presence of interfering nodes.

Service quality can be also enhanced by SDN–based approaches. In [15], [16] some controls are exposed to the users to manage the service quality. Poor performance due to link failures is addressed in [17], proving that SDN-based management can be used to improve WLAN networks. MultiFlow [18] aims to improve multicast communications using SDN principles. However, results are only presented as a numerical analysis and the channel utilization ratio may exceed the Legacy multicast one when the size of group is greater than a certain threshold.

In spite of the improvements made, most of the aforementioned works have either only been tested via simulations or require significant modifications to the wireless client's stack making them incompatible with the IEEE 802.11 standard.

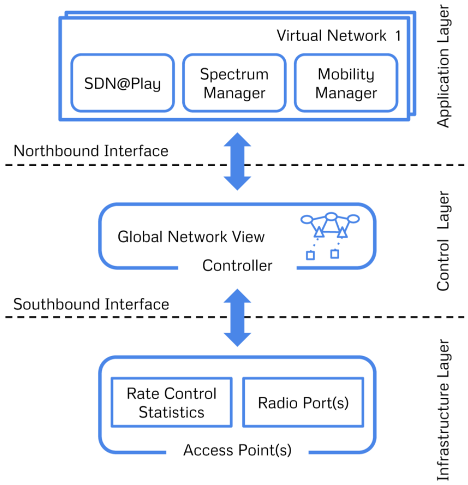


Fig. 1: *SDN@Play* System Architecture.

Conversely, in this work we aim at providing a practical *programmable* multicast rate–adaptation solution that is fully compatible with the IEEE 802.11 standard and that, by being fully software–defined, can be customized to the requirement of the particular multimedia application.

III. *SDN@Play* Design

Current networking technologies have several problems whose solutions often require substantial changes to the network stack. SDN has emerged as a new paradigm capable of addressing such limitations by introducing a fully programmable and modular network, making it possible to implement control and management tasks on top of a (logically) centralized control plane instead of implementing them as distributed applications. Figure 1 depicts the high–level reference system architecture used in this work. As can be seen, it consists of three layers: infrastructure, control and application. The infrastructure layer includes the data–plane network elements (i.e. the 802.11 APs) which are in constant communication with the (logically) centralized controller situated at the control layer. Applications run at the application layer leveraging on the global network view exposed by the controller in order to implement the network intelligence.

As discussed before, OpenFlow is one of the most widely adopted options to implement the link between the data–plane and the control–plane (the so–called southbound interface). Nevertheless, its features are mostly targeted at wired packet switched networks and are poorly suited for controlling wireless networks [2]. As a consequence, in the last years several SDN solutions for wireless and mobile networks have emerged, examples include EmPOWER [2] and Odin [3].

A. The Multicast Transmission Policy Abstraction

The fundamentals of SDN call for a clear separation between control–plane and data–plane. This requires identifying how network resources are exposed (and represented) to software modules written by developers and how those can affect the network state. Due to the stochastic nature of the wireless medium, the physical layer parameters that characterize the radio link between a Wi–Fi AP and a wireless client, such as transmission power, modulation and coding schemes, and MIMO configuration must be adapted in real–time to the actual channel conditions. As a consequence, any programming abstraction for rate–adaptation in Wi–Fi networks must clearly separate fast–control operations that must happen very close to the air interface, such as rate adaptation, from operations with looser latency constrains, such as mobility management.

In this work we propose the *Transmission Policy* abstraction which allows an SDN controller to reconfigure or replace a certain rate control policy if its optimal operating conditions are not met. More specifically, the *Transmission Policy* specifies the range of parameters the AP can use for its communication with a wireless client. Such parameters include:

- *MCSes*. The set of MCSes that can be used by the rate selection algorithm.
- *RTS/CTS Threshold*. The frame length above which the RTS/CTS handshake must be used.
- *No ACK*. The AP shall not wait for ACKs if true.
- *Multicast policy*. Specifies the multicast policy, which can be Legacy, DMS, or UR.
- *UR Count*. Specifies the number of UR retransmissions.

Transmission Policy configurations can be specified on a L2 destination address basis. As a result, for each destination address and for each AP in the network a specific *Transmission Policy* configuration can be created. Notice how the *Transmission Policy* abstraction allows the controller to specify the set of MCSes that can be used by the rate control algorithm. However, the frame–by–frame selection of the MCS is implemented at the AP and not at the controller.

Table I lists four *Transmission Policy* configurations, two for unicast addresses and two for multicast addresses. The first multicast entry (*01:00:5e:b4:21:90*) specifies *Legacy* as multicast mode. This instructs the AP to send every multicast frame with the specified destination address using 24 Mb/s as transmission rate. We remind the reader that in Legacy mode multicast frames are sent only once and no acknowledgment is generated by receptors. The second multicast entry (*01:00:5e:40:a4:b4*) specifies DMS as multicast mode. In this case for every multicast frame with this destination address, the AP will generate as many unicast frames as the number of receptors in the multicast group. The transmission rate for such unicast frame will be selected by the AP using the list of available MCSes specified by the corresponding unicast *Transmission Policy* configuration.

The content of Table I is manipulated by the controller via the southbound interface using a CRUD (Create, Retrieve, Update, Delete) model. The details of the signaling protocol are omitted due to space constrains.

TABLE I: *Transmission Policy* Configurations.

Destination	Type	MCS	RTS/CTS	No ACK	Multicast	UR Count
20:47:47:ac:61:5f	unicast	6, 12, 18, 24, 36, 48, 54	2436	False	n.a.	n.a.
5c:e0:c5:ac:b4:a3	unicast	6, 12, 18, 24, 36, 48, 54	2436	False	n.a.	n.a.
01:00:5e:b4:21:90	multicast	24	n.a.	n.a.	Legacy	n.a.
01:00:5e:40:a4:b4	multicast	n.a.	n.a.	n.a.	DMS	n.a.

B. Multicast Rate Adaptation

In this section we illustrate how the *Transmission Policy* abstraction can be used to implement the *SDN@Play* multicast rate adaptation mechanism. This algorithm has the goal of intelligently steering the data rate selection for multicast applications toward a more efficient operating point.

The idea behind *SDN@Play* is to use the link delivery statistics collected by the rate adaptation algorithm implemented at the AP and available at the controller to dynamically adapt the MCS used for multicast transmissions in Legacy mode. However, as discussed before, the rate adaptation algorithm is used only for unicast transmissions. As a result, if there are no ongoing unicast transmissions between an AP and a wireless client, no link delivery statistics will be computed. In order to circumvent this issue we introduce a two phases scheme.

During the *first phase*, which represents the smallest percentage of the algorithm time, the controller sets *DMS* as multicast policy for the multicast address M. We remind the reader that in DMS mode multicast transmissions are replaced by as many unicast transmissions as the number of receptors in a group[2]. This allows the rate adaptation algorithm to kick–in and to gather the link delivery statistics for all the receptors in the group. In the *second phase* the controller uses the link delivery statistics collected during the first phase to compute the MCS with the highest successful delivery probability for every receptor in the group. Based on this information, a worst receptor approach is used to compute the optimal transmission MCS R_{opt}. The controller then sets *Legacy* as multicast policy for the multicast address M and specifies R_{opt} as single entry in the list of available MCSes for that destination.

The whole process, sketched in Fig. 2, is repeated periodically with a configurable ratio between DMS and Legacy periods. This allows the programmer to trade accuracy for airtime utilization. Specifically, by increasing the fraction of time of the DMS mode it is possible to improve the link delivery ratio at the price of higher channel utilization. Conversely, by increasing the fraction of time of the Legacy mode, the airtime utilization is improved at the price of a possible lower frame delivery ratio (especially if channel conditions are fluctuating).

Based on the aforementioned link delivery statistics, the optimal transmission rate for a given multicast group is calculated by the Wi–Fi AP as follows. Let M be the set of $n = |M|$ multicast receptors in a multicast group and let R_i be the set MCS supported by the multicast receptor $i \in M$. If $p_i(r_j)$ is the delivery probability of the MCS index j at the multicast receptor i, we can define the valid multicast group transmission rates R_{valid} as follows:

$$R_{valid} = \bigcap_{i=1}^{n} \{r \in R_i | p_i(r) > p_{th}\} \quad (1)$$

This is the list of MCS indexes with a delivery probability higher than an input threshold p_{th} for all the receptors in the multicast group, i.e. any of those rates would results in a delivery probability of *at least* p_{th}. The optimal multicast transmission rate R_{opt} is then computed as follows:

$$R_{opt} = \begin{cases} max\,(R_{valid}) & if\ R_{valid} \neq \emptyset \\ min\Big(\bigcap_{i=1}^{n} \underset{r}{\text{argmax}}(p_i(r))\Big) & otherwise \end{cases} \quad (2)$$

This approach ensures that the selected multicast rate has a high delivery probability even for the multicast receptors experiencing bad channel conditions. Notice how if for a receptor there are not MCS indexes whose delivery probability is higher than the input threshold, then our algorithm selects for each receptor the MCS index with the highest delivery probability and from this set picks the lowest MCS index. This is done in order to ensure that the transmission can be decoded by the receptor with the weakest link to the AP.

It should be pointed out that signaling between the AP and the controller for the link delivery statistics interchange requires some limited adjustments on the AP. Nevertheless, it is worth emphasising that the deployment of this scheme does not need to perform changes to neither the wireless terminals nor the IEEE 802.11 protocol.

IV. Implementation details

To demonstrate the usefulness of *SDN@Play* in real-world settings, we implemented it over the EmPOWER platform. In particular: (i) we extended the southbound interface allowing it to collect link delivery ratio statistics; (ii) we extended the data–path implementation in order to properly handle multicast frames; and (iii) we added support for the new *Transmission Policy* primitive in the EmPOWER SDK.

A. Statistics gathering

The EmPOWER platform, on which *SDN@Play* is based, provides a rich set of programming primitives made available to the programmers trough a Python–based SDK. The list of primitives can be found in [2]. Primitives can operate in either *polling* or *trigger* mode. In the former mode (*polling*) the controller periodically polls one or more APs for a specific information, e.g. the number of packets received by a client.

[2]Notice how creation and maintenance of the multicast group is out of the scope of this work.

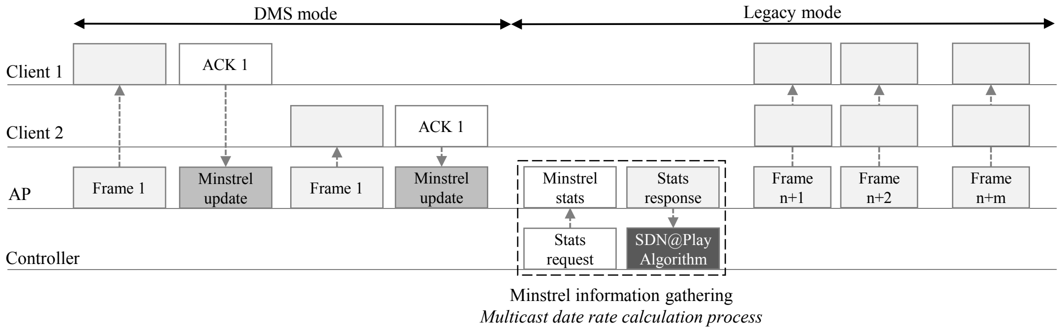


Fig. 2: *SDN@Play*'s two phases scheme. In the *first phase DMS* is used as multicast policy allowing the rate adaptation algorithm to gather link delivery statistics. In the *second phase* the multicast policy is switched to *Legacy* and the collected link delivery statistics are used to compute the optimal multicast MCS.

In the latter mode (*trigger*) a thread is created at one or more APs. Such thread is identified by a firing condition, e.g. the RSSI of one client going below a certain threshold. When such condition is verified a message is generated by the AP.

In this work we added support for a new *polling*–based primitive allowing the access to the rate adaptation algorithm statistics for a given client. For each supported MCS, the Exponentially Weighted Moving Average (EWMA) probability and the expected throughput in the last observation window are reported. Moreover, the total number of successful and failed transmissions are also reported. This primitive is used by the *SDN@Play* application to gather the link delivery statistics for all the wireless clients involved in multicast transmissions. We remind the reader that this information is maintained by the rate adaptation algorithm implemented by the AP. Therefore, no extra computation is added to the APs logic. More information on the particular rate adaptation algorithm used in our prototype is provided in the next subsection.

B. Data–path Implementation

Each AP consists of two components: one Open-vSwitch [19] instance managing the communication over the wired backhaul; and one Click modular router [20] instance implementing the 802.11 data–path. Click is a framework for writing multi–purpose packet processing engines and is being used to implement just the wireless client/AP frame exchange, while all the network intelligence is implemented at the centralized controller. Communications between Click and the controller take place over a persistent TCP connection (i.e. the southbound interface).

Rate adaptation is also implemented in Click using the Minstrel [21] algorithm (ported to C++ from its Linux Kernel implementation). Minstrel operations follow a multi–rate retry chain model in which four rate–count pairs, $r0/c0$, $r1/c1$, $r2/c2$ and $r3/c3$, are defined. Each pair specifies the rate at which a unicast frame shall be transmitted and a fixed number of retry attempts. Once the packet is successfully transmitted, the remainder of the retry chain is ignored. Otherwise the AP will move to the next pair in the chain. When the last pair has been also tried, the frame is dropped. For each supported MCS, Minstrel tracks the link delivery ratio and the expected packet throughput given the probability of success. Statistics are recomputed every 100ms. In particular the rates with the highest throughput, second highest throughput, and highest delivery probability are maintained by Minstrel.

TABLE II: Minstrel Retry Chain Configuration.

Rate	Look–around		Normal transmission
	Random $<$ Best	Random $>$ Best	
$r0$	Best rate	Random rate	Best rate
$r1$	Random rate	Best rate	Second best rate
$r2$	Best probability	Best probability	Best probability
$r3$	Base rate	Base rate	Base rate

In order to adapt to changes in channel conditions, Minstrel spends part of its time in a so-called *look–around* mode. Specifically, 90% of the time, Minstrel configures the retry chain using the collected link delivery statistics. In the remaining 10% of the time it randomly tries other MCSes to gather statistics. Table II summarizes the criteria used by Minstrel to fill the retry chain in both normal and *look–around* mode.

We extended the Click data–path implementation in order to support generalized transmission policies for unicast, multicast, and broadcast addresses as opposed to the original transmission policies that could be specified only for unicast addresses. According to the new transmission policies, the rate adaptation algorithm (i.e., Minstrel) will use the first entry in the list of available MCSes if the multicast mode is set to Legacy. Conversely, if the multicast mode is set to DMS, the frame will be duplicated for each receptor in the group and will be fed back to the rate control algorithm which will then apply the unicast transmission policy associated to that receptor. Finally, if the multicast mode is set to UR, the frame will be transmitted N times at the specified multicast rate.

C. The Multicast Transmission Policy Abstraction

The *Transmission Policy* abstraction is exposed through an object mapping properties to operations. Such an interface allows programmers to fetch the *Transmission Policy* configuration for a certain address by accessing the `tx_policy` property of a *Resource Block* object. A *Resource Block* is the minimum allocation block in the network and is defined as a 2–tuple $\langle f, b \rangle$, where f and b are, respectively, the center frequency and the band type. For example, the *Resource Block* made available by an 802.11n AP tuned on channel 36 and supporting 40 MHz–wide channels is represented by the tuple $(36, HT40)$. The prefix HT is used to indicate that this band supports the High Throughput MCSes. Each AP has as many *Resource Block*s as the number of available Wi–Fi interfaces.

The following Python listing shows how to access the *Transmission Policy* configuration for the *04:F0:21:09:F9:96* unicast address:

```
>>>block.tx_policy['04:F0:21:09:F9:96']
(<12,36,48,54>, 2436, False, None, None)
```

As can be seen, the object above contains a single entry mapping a unicast address with a *Transmission Policy* configuration. In this example, the address *04:F0:21:09:F9:96* has been assigned a configuration specifying which range of parameters the AP can use for its communication with the client (in this case adaptive rates selection will select the actual MCS).

Configuring the *Transmission Policy* is simply a matter of assigning new values to any of the port properties, for example the following listing sets DMS as transmission policy for the *01:00:5e:00:00:fb* multicast address:

```
>>>txp = block.tx_policies['01:00:5e:00:00:fb']
>>>txp.mcast = TX_MCAST_DMS
```

Similarly, the following listing sets the multicast mode back to Legacy and specifies also a new multicast rate:

```
>>>txp = block.tx_policies["01:00:5e:00:00:fb"]
>>>txp.mcast = TX_MCAST_LEGACY
>>>txp.mcs = [24]
```

The proposed solution allows the specification of flexible transmission policies for each multicast group. As a result, each group is assigned a rate that is calculated considering the conditions of all its multicast receptor.

V. Performance evaluation

The evaluation presented in this section has been carried out in a real environment with the goal of comparing *SDN@Play* with the multicast schemes currently defined in the 802.11 standard, namely Legacy and DMS. In this section we shall first describe the testing environment and the evaluation methodology, then we will discuss the outcomes of the measurements campaign.

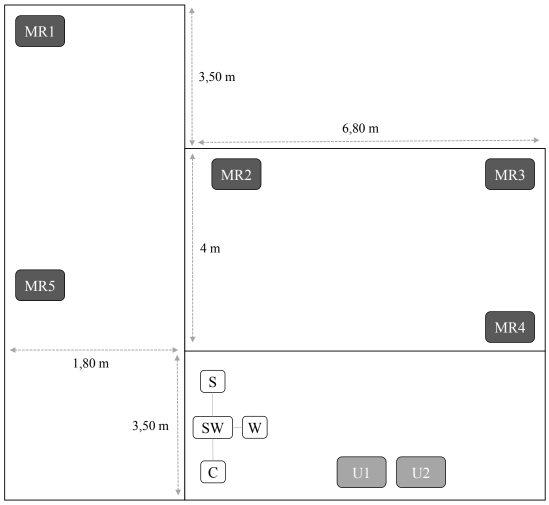


Fig. 3: Testbed deployment layout.

A. Evaluation Methodology

The testbed is composed of one AP (W), five multicast receptors (MRx), two wireless background traffic generators ($U1, U2$), one controller (C), one video server (S), and one Ethernet switch (SW). The testbed layout is sketched in Fig. 3.

The AP is based on the PCEngines ALIX 2D (x86) processing board and is equipped with a single Wi–Fi interface (Atheros AR9220 chipset). The AP runs the OpenWRT Operating System (15.05.01). All experiments are carried out on the 5 GHz band. The controller, the background traffic generators, and the multicast receptors are all Dell laptops equipped with an Intel i7 CPU, 8GB of RAM, and running Ubuntu 16.04.

A variable number of multicast receptors, ranging from 1 to 5, has been used in our measurements. In order to present a more realistic scenario we have also introduced some artificial background traffic in the network. For this purpose, two stations, denoted as *U1* and *U2*, generate a saturated UDP connection addressed at the AP. A multicast video stream is generated by the video server S and delivered to an increasing number of receptors. The video stream consists of a one minute sequence encoded using the High Efficiency Video Coding Standard (HEVC) and transmitted at 1.2 Mbps using FFmpeg [22].

Five scenarios have been defined in this study: Legacy, DMS, and *SDN@Play*. In the case of *SDN@Play* we considered three configurations, namely: $100/900$, $500/2500$ and $500/4500$. The first number refers to the duration (in ms) of the DMS period while the second one refers to the duration of the Legacy period. As evaluation metrics we considered delivery ratio and wireless channel utilization. Between each measurement the rate adaptation statistics have been cleared. Moreover, apart from the multicast video stream, no downlink traffic exists between AP and receptors. Therefore, the rate

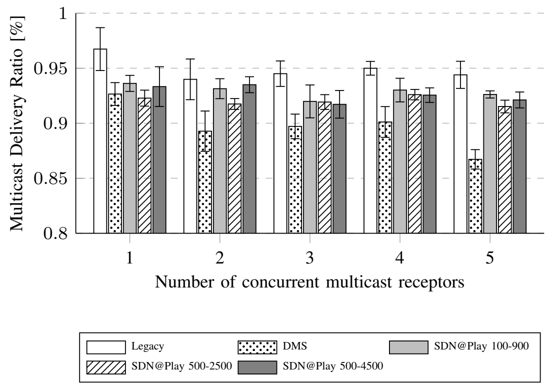


Fig. 4: Delivery ratio for the multicast video transmission Vs an increasing number of multicast receptors.

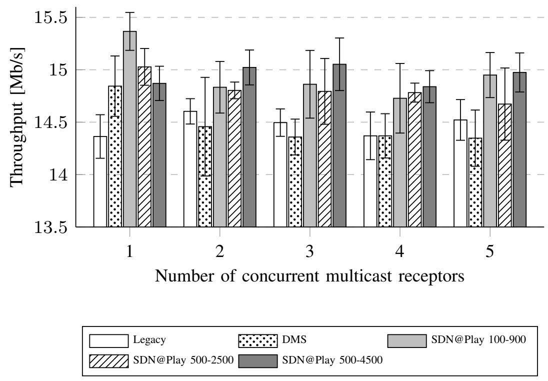


Fig. 5: Background unicast throughput Vs an increasing number of multicast receptors.

adaptation algorithm can only be executed during the DMS periods. Every measurement has been repeated 5 times.

B. Results

Figure 4 plots the average delivery ratio using different multicast strategies. As can be seen, the Legacy multicast strategy provides the highest frame delivery ratio. This is due to to fact that, in Legacy mode, multicast frames are transmitted at the lowest rate (which is usually also the more robust). DMS on the other delivers the worst performance due to the fact that DMS converts each multicast stream into as many unicast streams as the number of MRs. Each stream uses ACKs and retransmissions leading to a higher wireless channel utilization. Conversely, the *SDN@Play* scheme delivers in general the same performance level irrespective of the number of receptors. Nevertheless, the overall performance is slightly worse than the one provided by the Legacy multicast scheme. The reason for this behavior is that *SDN@Play* tries in general

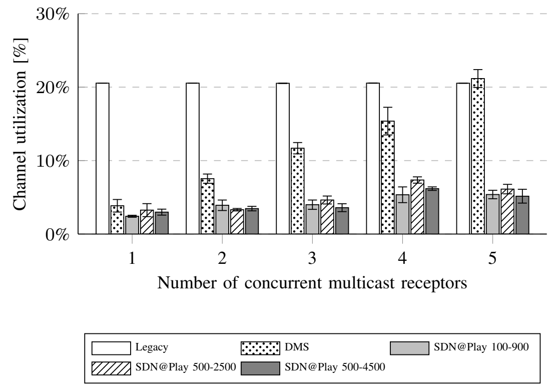


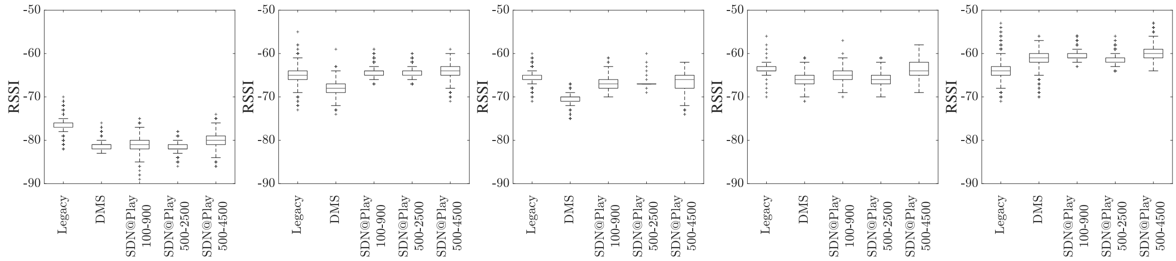
Fig. 6: Channel utilization Vs an increasing number of multicast receptors.

to use a high MCS index for multicast frames which may result in higher packet loss in case of channel quality fluctuations.

Figure 5 plots the average throughput of the two unicast streams for an increasing number of MRs. From this figure we can notice that, when the Legacy mode is used, the unicast streams have the lowest throughput. This is due to the fact that, in Legacy mode, multicast frames are transmitted at the lowest MCS which in time results in less resources being available to the unicast streams. In fact, *SDN@Play* outperforms the unicast throughput by up to 500 kbps in comparison with the standard schemes. Such behavior is more evident in Fig. 6, where the airtime utilized by the multicast stream is plotted. As can be seen, when operating in Legacy mode, 20% of the channel resources are used by the multicast stream. It is also interesting to notice that, when 5 multicast receptors are active, the DMS and the Legacy airtime utilization are approximately the same. On the other hand, the airtime used by *SDN@Play* only marginally increases with the number of receptors. As a result, a reduction in the channel utilization up to 80% is achieved. Figure 8 shows the total traffic associated to the multicast stream in the various scenarios. It is interesting to notice that *SDN@Play* essentially delivers as much traffic as the legacy scheme while using only a fraction of the resources.

Figure 9 reports the distribution of the MCS used in the case of 5 MRs. The Legacy scheme is omitted because only the lowest MCS is used. As can be seen, in DMS mode almost all transmissions happen at the highest MCS (54 Mb/s). On the contrary, in the three *SDN@Play* scenarios the MCS distribution is significantly different. In particular it can be noticed that using long DMS periods ($500-2500$ and $500-4500$) allows the system to quickly converge on the best MCS. Conversely, with short DMS periods ($100-900$) the rate adaptation algorithm may not have enough time to converge on the optimal MCS, thus the presence in the distribution of low MCS indexes.

In order to provide additional information about the conditions in which the performance evaluation described in this



(a) Multicast receptor 1. (b) Multicast receptor 2. (c) Multicast receptor 3. (d) Multicast receptor 4. (e) Multicast receptor 5.

Fig. 7: Signal Strength perceived by the five multicast receptors.

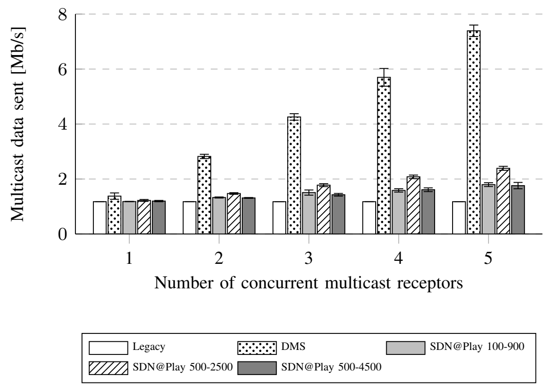


Fig. 8: Multicast traffic Vs an increasing number of receptors.

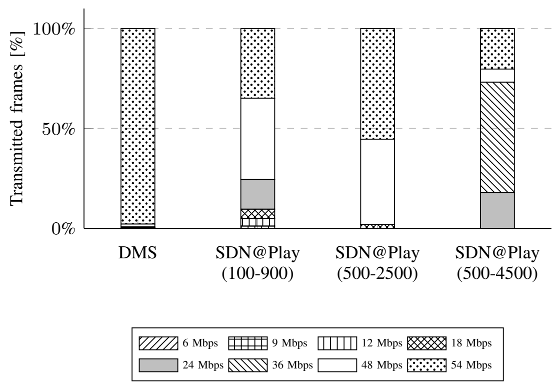


Fig. 9: Distribution of the rates used by each multicast scheme.

section has been performed, we collected the RSSI values for all the frames decoded by the MRs during the measurements. Figure 7 plots the distribution of the RSSI samples in the case of 5 multicast receptors. As can be seen the RSSI conditions for the 5 multicast schemes are essentially constant across all the multicast receptors. Similar considerations can be made for the other measurements with a smaller number of receptors.

VI. Conclusions

In this paper a new SDN–based approach is proposed to adapt the multicast data rate in IEEE 802.11–based WLANs. For this purpose, a two–phase algorithm, named *SDN@Play*, has been designed and implemented. *SDN@Play* coordinates the usage of different retransmission policies allowing the Wi–Fi APs to always use the most efficient multicast transmission rate. An experimental evaluation carried out over a real–world testbed shows that *SDN@Play* can deliver an improvement of up to 80% in terms of channel utilization compared to legacy 802.11 multicast while maintaining full backward compatibility with standard 802.11 wireless terminals.

As future work we plan to extend *SDN@Play* in order to account for multiple multicast groups as well as multiple Wi–Fi APs. Moreover, we also plan to jointly address mobility management and rate adaptation for both unicast and multicast flows. Finally, we are also considering extending the scope of the work to encompass also the wired segment of the network.

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2.4. Joint Mobility Management and Multicast Rate Adaptation in Software-Defined Enterprise WLANs

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1

Joint Mobility Management and Multicast Rate Adaptation in Software–Defined Enterprise WLANs

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***Abstract*—The ever–increasing demand for mobile content delivery and multimedia services is bringing renewed interest in multicast communications in Wi–Fi based WLANs. Nevertheless, multicast over Wi–Fi raises several challenges including low data rates and coexistence issues with other unicast streams. Some amendments to the Wi–Fi standard, such as 802.11aa, have introduced new delivery schemes for multicast traffic as well as finer control on the low–level aspects of the 802.11 medium access scheme. However, the logic for using such features is left to the implementer of the standard. In this paper we present *SDN@Play Mobile*, a novel SDN–based solution for joint mobility management and multicast rate–adaptation in Wi–Fi networks. The solution builds upon a new abstraction, named *Transmission Policy*, which allows the SDN controller to reconfigure a multicast transmission policy when its optimal operating conditions are not met. An experimental evaluation carried out over a real–world testbed shows that our approach can deliver significant improvements in terms of both throughput and channel utilization compared to the legacy 802.11 multicast scheme. Finally, we release the entire software implementation under a permissive APACHE 2.0 license for academic use.**

***Keywords*—WLANs, IEEE 802.11, multicast, rate adaptation, software defined networking, mobility, multimedia.**

I. Introduction

Wireless and mobile communications are witnessing an exponential growth in the amount of traffic exchanged. For example, the latest CISCO Visual Network Index [1] reports that Wi–Fi and mobile traffic will account for 49% in 2020. Multimedia communications are becoming dominant, and it is also expected that 78% of the mobile traffic will be video by 2021. Moreover, the emergence of mobile devices and the demand for constant connectivity have led Wi–Fi networks to be deployed everywhere. In an effort to improve the performance, these networks are typically composed of multiple Access Points (APs) to increase the capacity of the network and to provide support for roaming users. Both the industry and the academia are well aware of the importance of multicasting services. This is demonstrated by the undergoing standardization efforts for the emerging 5G networks, which has led to the release of the Multimedia Broadcast Multicast Service (MBMS) by 3GPP [2] and by the increasing body of literature on this topic [3], [4], [5].

Multicast and broadcast services are a particular class of video traffic where contents must be delivered to a group of users. Due to the high bandwidth and low delay requirements of this traffic, multicast transmissions become an effective solution to optimize network resources. Sport events, conferences, game streaming, airports services, and real–time lessons are just some of the scenarios where multicast transmissions can be used. Moreover, wireless multicast can be used also in machine–to–machine communications in scenarios such as transport and emergency systems. Lastly, software upgrades can be also further improved using multicast transmissions.

IPTV services over Wi–Fi are also a good example of multicast video distribution in which most of the users tend to connect to the network through mobile devices. This fact shows how this technology can be widely used for business and entertainment purposes as well as the importance of ensuring reliable transmissions and user mobility. An example of its applicability can be found in the deployment of a campus network IPTV system to enable efficient distribution of multicast traffic over a WLAN [6].

IEEE 802.11–based WLANs dynamically choose among different Modulation and Coding Schemes (MCSes) for frame transmissions. For example, in 802.11a/g WLANs, devices can choose among MCSes resulting in bitrates ranging from 1 to 54 Mb/s, while in 802.11n/ac WLANs higher MCSes are available. However, since according to the IEEE 802.11 standard each frame must be acknowledged by the receiver, the rate selection mechanism is restricted to unicast traffic. This is due to the fact that, in case of multicast transmissions, acknowledgments cannot be used given that they would inevitably collide at the transmitter. As a result, multicast frames are sent at the lowest MCS and do not make use of any feedback mechanism. This implies several drawbacks: (i) the throughput of multicast transmissions is very limited; (ii) the use of basic data rates consumes more radio resources affecting also the capacity available to other (unicast) flows; and (iii) since multicast frames are not retransmitted, the reliability of the multicast streams can be adversely impacted by the channel conditions.

The traditional Wi–Fi network architecture hinders the introduction of new solutions to overcome the problems presented above while maintaining the compatibility with the 802.11 standard. In this regard, Software Defined Networking (SDN) has recently emerged as a new way of refactoring network functions. By clearly separating data–plane from control–plane and by providing high level programming abstractions, SDN allows implementing traditional network control and management tasks on top of a *logically* centralized controller. However, albeit SDN is already an established technology in the wired domain, with OpenFlow playing the role of de–facto standard [7], equivalent solutions for wireless networks have only recently started to appear [8], [9], [10].

In this paper we propose a joint mobility management and multicast rate adaptation algorithm for Software–Defined WLANs. Our work aims at improving the performance of multicast communications while reducing the utilization of radio resources. This goal is achieved in a two–step procedure: (i) selecting the multicast data rate that can deliver the expected quality in terms of performance; and (ii) associating the multicast receivers to the APs in a way that the radio resource utilization across the entire network is minimized. This paper builds upon our previous work [11] by extending the proposed algorithm to account also for mobile multicast receivers and association management. Moreover, we also report on an updated proof–of–concept implementation of the proposed solution and on its field evaluation. The entire implementation, including the controller and the data–path, is released under a permissive APACHE 2.0 license[1] for academic use.

The rest of the paper is structured as follows. In Section II we discuss the related work. The system architecture as well as the joint mobility management and rate control algorithms are presented in Section III. Section IV reports on the implementations details, while Section V describes the experimental evaluation and discusses the results of the measurements campaign. Finally, Section VI draws the conclusions pointing out future research directions.

II. Related Work

In this section we first provide a background on multicast communications in 802.11 WLANs. Then, we review the most relevant related work highlighting our technical contributions.

Multicast communications are an efficient way to send the same information to many clients. In fact, by exploiting the broadcast nature of the wireless medium, it is possible to deliver the same frame to multiple wireless terminals instead of transmitting it individually to each of them. Nevertheless, in IEEE 802.11 WLANs multicast frames are never retransmitted nor acknowledged. As a consequence, the transmission reliability is highly reduced. Moreover, the lack of feedback information prevents the devices from adapting the transmission rate. Consequently, the 802.11 standard recommends the use of the basic data rate for the multicast traffic.

The IEEE 802.11aa amendment [12] has been introduced to improve the performance of the multicast communications while keeping the compatibility with current devices. The amendment improves the multicast reliability level by introducing the Group Addressed Transmission Service. This service specifies several retransmission policies and is composed of two different mechanisms: Direct Multicast Service (DMS) and Groupcast with Retries (GCR). In DMS mode each multicast frame is converted into as many unicast frames as the number of receivers in the multicast group. Each unicast frame may be retransmitted as often as necessary until the AP receives the ACK or the retransmission counter reaches its limit. In spite of ensuring high reliability, DMS does not scale well with the number of receivers in the multicast group.

GCR is a flexible service composed of three retransmission methods: Legacy Multicast, Unsolicited Retries (UR) and Block ACK (BACK). The Legacy Multicast mode is the one defined in the original IEEE 802.11 standard. The UR policy specifies a number of retry attempts, N, in a manner that a frame is transmitted $N + 1$ times. In this way, the probability of a successful transmission is increased. However, UR may unnecessarily retransmit frames, hence increasing the overall network utilization. In BACK mode the AP reaches an agreement with the multicast receivers about the number of consecutive unacknowledged frames. After that, the AP sends a burst of multicast packets up to that number and requests a Block ACK from each receiver. Both this request and the corresponding ACKs are sent in unicast mode. Despite the control traffic overhead is reduced, also this approach does not scale with the number of receivers in the group. A comprehensive description of the various multicast schemes supported by the 802.11 standard can be found in [13].

Multicast rate selection may be achieved by defining feedback gathering mechanisms allowing the transmitter to gain a better knowledge of the status of the wireless medium. Leader–Based Protocols (LBP) are the most common proposals in the literature. LBP [14] aims at improving multicast communications by enabling ACKs. For this purpose, the receiver exhibiting the worst signal quality is selected as a leader of the group and is in charge of sending ACKs. However, a procedure for the leader selection is not provided. The Auto Rate Selection Multicast (ARSM) mechanism [15] divides its operation mode into two phases: in the first one, the group leader is selected, whereas in the second step the Signal–to–Noise Ratio (SNR) derived from the ACKs of the leader is used to adapt the transmission rate. Hierarchical–ARSM (HARSM) [16] is an evolution of ARSM for hierarchical video transmissions over WLANs that ensures a minimum quality of the video sequence for all the receivers. The rate adaptation based on the SNR is also used in SNR–based Auto Rate for Multicast (SARM) [17]. In this scheme, the AP identifies the worst receiver by sending beacon frames to which the stations must reply indicating their own SNR. After that, the APs must inform the remaining stations about the new situation. However, changes at the client side are needed to implement this scheme.

The multicast rate adaptation problem is exacerbated when considering mobile users since their channel conditions constantly change. Based on these conditions, efficient handover solutions are required to migrate these clients from one AP to another in order to ensure that the quality of service requirements of the end–user are met. This is precisely the target pursued in [18], [19]. The mobility problem in multicast is also analysed from the point of view of the wired backbone interconnecting the Wi–Fi APs [20], [21]. These proposals, however, focus on balancing the bandwidth in the backhaul, neglecting the challenges related to the radio access segment.

An efficient handover process must ensure that the communication is not interrupted while performing the association

[1] Available at: http://empower.create-net.org/

with the new AP. Nevertheless, this concept, called seamless handover, is difficult to achieve in traditional network architectures and has motivated the emergence of some SDN–based works. In M–SDN [22] the central controller tracks channel quality information to identify the best APs for a handover. After that, a route from the current AP to the target ones is computed. This approach reduces the service disruption time at the price of generating additional traffic in the network. A multi–channel architecture is introduced in [23], in which several APs share the same MAC address to ensure seamless handover. The validity of this proposal is tested via simulation and over an OpenFlow–based testbed. However, it should be noted that these approaches are targeted at unicast traffic, and to the best of our knowledge, no current work addresses the user mobility problem in multicast environments over SDN–based WLANs.

Quality of Experience (QoE) has often been used as basis for rate adaptation in multimedia applications. In [24] a neural network is designed to build a model that maps QoE measurements into MCSes. PSQA [25] is developed as a hybrid objective–subjective metric that simulates how humans perceive impairments to video transmissions. Similar consideration can be made for [26]. In [27] the authors address the multicast video delivery using a real–life testbed. In this solution the time is split into a transmission and a polling period. During the transmission period, the stations collect the sequence numbers of the received frames. After that, the APs gather that information to calculate the link delivery probabilities. The transmission rate is selected by comparing these values with the ones obtained from the two previous rounds. Changes at the client side are needed to implement this scheme. MultiFlow [28] aims to improve multicast communications using SDN principles. However, results are only presented as a numerical analysis and the channel usage of the proposed scheme may exceed the legacy multicast one when the size of group is greater than a certain threshold.

In spite of the improvements made, most of the aforementioned works have either been tested via simulations or require significant modifications to the wireless client's stack, hence making them incompatible with the IEEE 802.11 standard. Moreover, the mobility problem is further aggravated when considering multicast communications given that both the data rate selection and the handover time affect all the receivers in the network. In this regard, no research work in the literature jointly address association management and multicast rate selection in 802.11–based WLANs. Conversely, in this work we aim at providing a practical and *programmable* multicast rate adaptation and mobility management solution that is fully compatible with the IEEE 802.11 standard and that, by being fully software–defined, can be customized to the requirements of the particular multimedia application.

III. System Design

Current networking technologies have several problems whose solutions often require substantial changes to the network stack. SDN has emerged as a new paradigm capable of addressing such limitations by introducing a fully

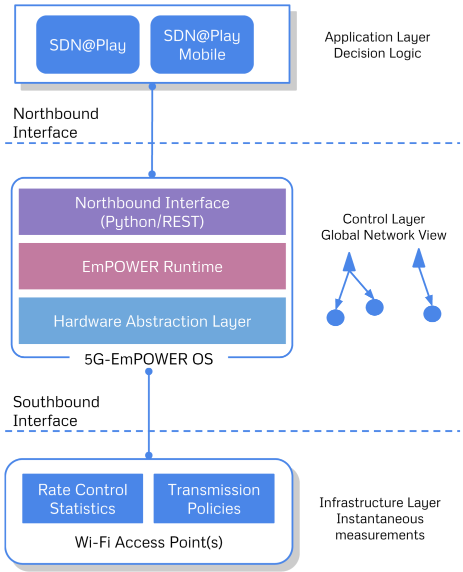


Fig. 1: *SDN@Play Mobile* System Architecture.

programmable and modular network, making it possible to implement control and management tasks on top of a (logically) centralized control plane instead of implementing them as distributed applications. Figure 1 depicts the high–level reference system architecture used in this work. As can be seen, it consists of three layers: infrastructure, control and application. The infrastructure layer includes the data–plane network elements (*i.e.* the 802.11 APs) which are in constant communication with the (logically) centralized controller situated at the control layer. Applications run at the application layer leveraging on the global network view exposed by the controller to implement the network intelligence.

As noticed before, OpenFlow is one of the most popular options to implement the link between data–plane and control–plane (the so–called southbound interface). Nevertheless, its features are mostly targeted at wired networks and are poorly suited for controlling 802.11–based WLANs [8]. As a consequence, in the last years several SDN solutions for wireless and mobile networks have emerged. Examples include 5G–EmPOWER [8], Odin [9], and OpenSDWN [10].

The mobility management and multicast rate adaptation scheme presented in this paper has been implemented and tested on top of the 5G–EmPOWER platform [8]. Nevertheless, it should be noted that our work is very general and can be in principle applied to any centrally controlled enterprise WLAN. The system design is described in this section. First, we will summarize the Light Virtual Access Point (LVAP) abstraction which is used to control Wi–Fi stations association [9]. Then, we will introduce the *Transmission Policy* abstraction designed to allow an SDN controller to configure a rate adaptation policy of a Wi–Fi AP. Finally, we will show how these abstractions can be used to implement a joint multicast rate selection and mobility management algorithm.

A. The Light Virtual Access Point Abstraction

Different link layer technologies, or as a matter of fact even different releases of the same technology, can differ significantly in how a client's state is handled. For example, QoS and handover management changed significantly over the lifespan of the IEEE 802.11 family of standards. As a consequence, exposing the implementation details of these technologies would increase the complexity for the programmer and would severely limit the adoption of a certain solution.

The *LVAP* abstraction [9] is a per–client virtual AP that provides a high–level interface for wireless clients state management. The implementation of such an interface handles all the technology–dependent details (*i.e.* the complexities of the IEEE 802.11 protocol) such as association, authentication, handover and resource management, and introduces seamless mobility support. A client attempting to join the network will trigger the creation of a new *LVAP*. For this purpose, a wireless client generates a probe request that will be received at an AP and forwarded to the controller. In case of a new client, the controller will generate a probe response frame through the creation of an *LVAP* at the requesting AP. The *LVAP* will thus become a potential AP for the client to perform an association. Since an *LVAP* is created for each each wireless client, after generating an *LVAP*, probe requests received from the same client by any AP in the network will be ignored.

The controller can also decide whether the network has enough resources to handle the new client and might suppress the generation of the *LVAP*. Similarly, each AP will host as many *LVAP*s as the number of wireless clients that are currently under its control. Such *LVAP* has an identifier that is specific to the newly associated client (in a Wi–Fi network the *LVAP* can be thought as a Virtual AP with its own BSSID). Removing an *LVAP* from an AP and instantiating it on another AP effectively results in a handover.

B. The Transmission Policy Abstraction

The fundamentals of SDN call for a clear separation between control–plane and data–plane. This requires identifying how network resources are exposed (and represented) to software modules written by developers and how those can affect the network state. Due to the stochastic nature of the wireless medium, the physical layer parameters that characterize the radio link between a Wi–Fi AP and a wireless client, such as transmission power, MCS, and Multiple Input Multiple Output (MIMO) configuration must be adapted in real–time to the actual channel conditions. Therefore, any programming abstraction for rate–adaptation in Wi–Fi networks must clearly separate fast–control operations that must happen very close to the air interface, such as rate adaptation, from operations with looser latency constrains, such as mobility management.

In this work we propose the *Transmission Policy* abstraction which allows an SDN controller to reconfigure or replace a certain rate control policy if its optimal operating conditions are not met. The *Transmission Policy* specifies the range of parameters the AP can use for its communication with a wireless client. Such parameters include:

- *MCSes*. The set of MCSes that can be used by the rate selection algorithm.
- *RTS/CTS Threshold*. The frame length above which the RTS/CTS handshake must be used.
- *No ACK*. The AP shall not wait for ACKs if true.
- *Multicast policy*. Specifies the multicast policy, which can be Legacy, DMS or UR.
- *UR Count*. Specifies the number of UR retransmissions.

Transmission Policy configurations can be specified on a L2 destination address basis. As a result, for each destination address and for each AP in the network a specific *Transmission Policy* configuration can be created. Notice that the *Transmission Policy* allows the controller to specify which MCSes can be used by the rate control algorithm implemented at the AP. However, the actual frame–by–frame selection of the MCS is done at the AP and not at the controller.

Table I lists four *Transmission Policy* configuration examples, two for unicast addresses and two for multicast addresses. The first multicast entry (*01:00:5e:b4:21:90*) specifies Legacy as multicast mode. This instructs the AP to send every multicast frame with the specified destination address using 24 Mb/s as transmission rate. We remind the reader that in Legacy mode multicast frames are sent only once and that no acknowledgement is generated by the receivers. The second multicast entry (*01:00:5e:40:a4:b4*) specifies DMS as multicast mode. In this case, for every multicast frame with this destination address, the AP will generate as many unicast frames as the number of receivers in the multicast group. The transmission rate for such unicast frame will be selected by the AP using the list of available MCSes specified by the corresponding unicast *Transmission Policy* configuration. The content of the table is manipulated by the controller using a CRUD (Create, Retrieve, Update, Delete) interface. The details of the signalling protocol can be found in [29].

C. Multicast Rate Adaptation

In this section we illustrate how the *Transmission Policy* abstraction is used to implement *SDN@Play*. This algorithm has the goal of intelligently steering the data rate selection for multicast applications toward a more efficient operating point.

The idea behind *SDN@Play* is to use the link delivery statistics collected by the rate control algorithm implemented at the AP to dynamically adapt the MCS used for multicast transmissions in Legacy mode. However, as stated before, the rate control algorithm is used only for unicast transmissions. As a result, the link delivery statistics will be only computed if unicast traffic is transmitted between an AP and a client. In order to circumvent this issue, we introduce a two phases scheme, sketched in Fig. 2, which is marked by the alternation of two multicast policies defined in IEEE 802.11aa.

In the *first phase* the controller uses the *Transmission Policy* abstraction to set DMS as the multicast policy for a multicast address. We remind the reader that in DMS multicast transmissions are replaced by as many unicast transmissions as the number of receivers in a group[2]. This allows the rate control

[2]Notice that the creation and maintenance of the multicast group is out of the scope of this work.

TABLE I: *SDN@Play* Configuration Examples.

Destination	Type	MCS	RTS/CTS	No ACK	Multicast	UR Count
20:47:47:ac:61:5f	unicast	6, 9, 12, 18, 24, 36, 48, 54	2436	False	n.a.	n.a.
5c:e0:c5:ac:b4:a3	unicast	6, 9, 12, 18, 24, 36, 48, 54	2436	False	n.a.	n.a.
01:00:5e:b4:21:90	multicast	24	n.a.	n.a.	Legacy	n.a.
01:00:5e:40:a4:b4	multicast	n.a.	n.a.	n.a.	DMS	n.a.

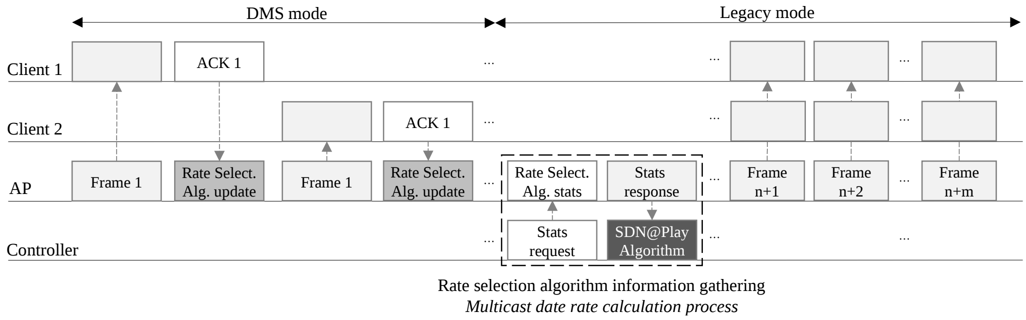


Fig. 2: *SDN@Play*'s two phases scheme. In the *first phase DMS* is used as multicast policy allowing the rate adaptation algorithm to gather link delivery statistics. In the *second phase* the multicast policy is switched to *Legacy* and the collected link delivery statistics are used to compute the optimal multicast MCS.

algorithm at the AP to gather the link delivery statistics for all the receivers (more information about the rate adaptation algorithm is provided in Sec. IV). In the *second phase* the controller uses the link delivery statistics collected during the *first phase* to calculate for each receiver the MCSes with the highest delivery probability. Based on this information, a worst receiver approach is used to compute the MCS currently used for the multicast group, as explained in more detail below.

Let M be the set of receivers in a multicast group, R the set of MCSes supported by the multicast receivers, and U the set of Wi–Fi APs. Moreover, let $P_r^{n,n'}$ be the delivery probability between AP $u \in U$ and receiver $n' \in M$ using MCS $r \in R$. On this basis, we can select the valid MCSes R_{valid}^n for each AP $n \in U$ as the set of MCS indexes with a delivery probability higher than a given threshold r_{th}:

$$R_{valid}^n = \bigcap_{n' \in M} \left\{ r \in R | P_r^{n,n'} > r_{th} \right\} \quad \forall n \in U \qquad (1)$$

The multicast transmission rate R_{tx}^n is then given by:

$$R_{tx}^n = \begin{cases} max\left(R_{valid}^n\right) & if\ R_{val}^n \neq \emptyset \\ min\left(\bigcup_{n' \in M} \left\{ \underset{r \in R}{\operatorname{argmax}}(P_r^{n'}) \right\} \right) & otherwise \end{cases} \quad \forall n \in U \qquad (2)$$

The threshold r_{th} allows the programmer to set a relation between the reliability level and the channel occupancy ratio. It should be noted that, especially in poor channel conditions, lower rates may have also higher delivery probabilities given that, at a low rate, frames are more likely to be properly transmitted. As an example, let 50% and 95% be two possible values for the threshold r_{th}. The first example would enable the selection of the rates whose delivery probability could be as low as 50%, meaning that each frame will be transmitted on average twice. By contrast, since $1/0.95 \simeq 1$, each frame could be successfully transmitted at the first attempt. In this regard, the use of high values for r_{th} increases the reliability of the multicast transmission with the drawback of increasing also the amount of time the channel remains busy due to the utilization of less efficient MCSes. Hence, a tradeoff for this value must be selected according to the channel conditions. Considering the lack of retransmissions and ACKs in multicast communications, in this work all the measurements have been performed using 95% for the threshold r_{th}.

The two–phases process shown in Fig. 2 is repeated periodically with a configurable ratio between the DMS and Legacy periods. This allows the programmer to trade accuracy for airtime utilization. More specifically, increasing the portion of time of DMS leads to an improvement in terms of reliability at the expense of a higher channel utilization. Conversely, by increasing the fraction of time that Legacy is used, the airtime utilization is improved at the expense of a possible lower delivery ratio (especially if channel conditions are fluctuating). Furthermore, this approach ensures that the selected rate has a high delivery probability even for the receivers experiencing bad channel conditions. Notice that if for a receiver none of the MCS indexes has a delivery probability higher than the input threshold, our algorithm selects for each receiver the MCS index with the highest delivery probability, and from this set, it chooses the lowest MCS index. This is done in order to increase the probability of the transmission being properly decoded by all the receivers in the multicast group.

SDN@Play has been preliminary introduced in [11] for stationary multicast receivers. In the next subsection we describe how the multicast rate selection algorithm has been paired with a mobility management scheme in order to account also for mobile multicast receivers.

D. Mobility Management

SDN@Play Mobile aims at jointly improving client association and multicast rate selection while minimizing the network–wide channel occupancy. To this goal, the stations periodically report to the serving AP the list of neighboring APs together with the experienced channel quality[3]. This information is gathered using the *Beacon Reports* available in IEEE 802.11k [30] and included in the 2012 version of the IEEE 802.11 [31] standard. *Beacon Reports* enable an AP to request its stations to report the list of APs from which they can receive beacon frames on a specified channel or channels. Stations report the measurements obtained from the beacons and probe response frames using a *Beacon Report*. Finally, *Beacon Reports* are aggregated by the AP and reported back to the controller where they are used to build a network–wide downlink channel quality map.

Based on the information obtained from the *Beacon Reports*, *SDN@Play Mobile* periodically checks the channel quality between each multicast receiver and all its neighbouring APs (including the serving AP). If the signal strength between a multicast receiver and its serving AP is below a certain level for five consecutive checks *or* if another AP can provide a considerable channel quality improvement for five consecutive checks, then a handover is triggered. This is intended to reduce the ping–pong effect. It should be noted that these values can be freely set by the implementer based on the sensitivity of the devices or on the quality requirements of the application.

Let $S(n)$ be the set of receivers served by AP $n \in U$, with $S(n) \subset M$. Also, let $\rho^n_{n'}$ be the channel quality between the AP $n \in U$ and the multicast receiver $n' \in M$, *i.e.* the RSSI level of the receiver measured *at* the AP. When a handover process for a given receiver n' is triggered, we compute the average channel quality $\rho(n)$ and the standard deviation $\sigma(n)$ for all the APs in the network:

$$\rho(n) = \frac{\sum_{n' \in S(n)} \rho^n_{n'}}{|S(n)|} \tag{3}$$

$$\sigma(n) = \sqrt{\frac{1}{|S(n)|} \sum_{n' \in S(n)} (\rho^n_{n'} - \rho(n))^2} \tag{4}$$

Notice that, if an AP is not serving any receiver or it does not fall in the coverage area of the receiver n', then the two quantities above are undefined. Furthermore, it is important to highlight that, only in the case that the set of receivers attached to an AP is empty *and* the AP is within range of receiver n', the previous quantities will be set as $|S(n)| = 1$ and $\sigma(n) = 0$.

We then compute the list $\Omega(n')$ of candidate APs for the multicast receiver n'. Remember that the multicast rate of the receiver n' after the handover will be influenced by the channel quality of the receivers already served by the target AP. As a result, in order to ensure that we do not handover the receiver n' to an AP which is serving receivers with much worse channel conditions, we set a lower bound for the construction of the list of candidate APs $\Omega(n')$:

$$\Omega(n') = \{n \in U | \rho(n) - \sigma(n) \leq \rho^n_{n'}\} \tag{5}$$

Notice how this definition could result in an empty set in case that there are no APs within the range of n' that satisfy the channel quality condition. In this case, the channel quality constrain is removed and all the APs are taken into consideration. Using this method the set of candidate APs contains at least the AP that is currently serving the receiver n'.

Once this process is finished, the algorithm chooses the AP in $\Omega(n')$ that would allow the receiver n' to receive the multicast transmission using the most efficient MCS. For more information about how the handover is implemented we refer the reader to [10]. After performing the handover, the multicast transmission rate for all the APs in the network is recomputed. Then the controller can calculate the new network–wide channel occupancy. If as result of the handover the channel utilization has increased, the handover is reverted. If this occurs, and in order to avoid oscillations, the new AP is not considered as candidate AP for the receiver n' for the next 5 iterations of the algorithm.

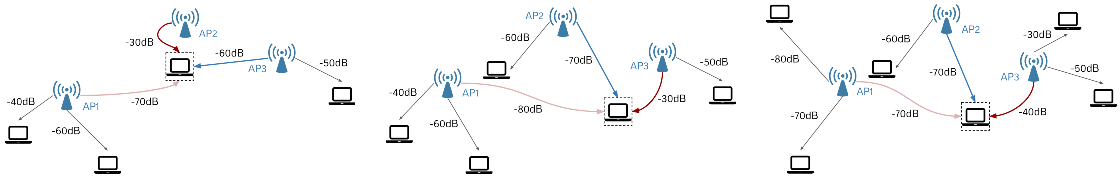
Figure 3 depicts a set of representative network configurations to show how the algorithm would select the best AP for a certain receiver (the dashed one). The link between this receiver and its serving AP is indicated by a blue arrow, while the ones between the remaining stations and their serving APs are represented by grey arrows. The arrows in dark red refer to stronger links with regard to the current one for the evaluated receiver, and which enabled the handover evaluation process. Finally, other equal or weaker links in terms of signal quality are presented by light red arrows. The numerical results derived from the quantities $\rho(n)$, $\sigma(n)$, and $\Omega(n')$ for the scenarios (a), (b) and (c) are reported in Table II.

Figure 3a shows a scenario with an idle AP (AP_2) and where the evaluated receiver is initially attached to AP_3. After computing the quantities mentioned before, the algorithm selects as candidates AP_2 and AP_3. AP_2 is selected as target AP for the handover since it provides the best channel quality. In the second example, shown in Fig. 3b, AP_1 does not meet the quality requirements, as a result, AP_3 is selected for the handover. Finally, a scenario where several stations are attached to the same AP is presented in Fig. 3c. Albeit all the APs qualify as candidates for the handover, AP_3 is selected for the association given that it provides the best channel quality.

IV. IMPLEMENTATION DETAILS

To demonstrate the usefulness of *SDN@Play Mobile* in real–world environments, we have implemented it over the 5G–EmPOWER platform. In particular: (i) we extended the southbound interface allowing it to collect link delivery ratio statistics and *Beacon Reports*; (ii) we extended the data–path implementation to properly handle multicast frames; and (iii) we added support for the new *Transmission Policy* primitive in the 5G–EmPOWER Software Development Kit (SDK).

[3]Notice that, in this work we use the Received Signal Strength Indicator (RSSI) as an estimator of the channel quality. Nevertheless, other channel quality indicators, such as packet loss, could be also used.



(a) Best RSSI from an AP serving no receivers.

(b) Best RSSI from an AP serving a single receiver.

(c) At least two receivers are served by each AP.

Fig. 3: Examples of different network distributions for the selection of the candidates APs.

TABLE II: Numerical results of the different network distribution examples for the selection of the candidates APs.

		$\rho(n)$	$\sigma(n)$	**Interval**	**RSSI**	**Cand.**
Fig. 3a	AP1	-56.67	12.47	[-69.14, -44.20]	-70	No
	AP2	-30	-	[-30, -30]	-30	Yes
	AP3	-55	5	[-60, -50]	-60	Yes
Fig. 3b	AP1	-60	16.33	[76.33, -43.68]	-80	No
	AP2	-65	5	[-70, -60]	-70	Yes
	AP3	-40	10	[-50, -30]	-30	Yes
Fig. 3c	AP1	-73.33	4.70	[78.08, 68.63]	-70	Yes
	AP2	-65	5	[-70, -60]	-70	Yes
	AP3	-40	8.16	[-48.16, -31.84]	-40	Yes

A. *Statistics Gathering*

The 5G–EmPOWER platform provides a rich set of programming primitives exposed to the programmer trough a Python–based SDK. The list of primitives can be found in [8]. Primitives can operate in either *polling* or *trigger* mode. In the former mode (*polling*) the controller periodically polls the APs for specific information, *e.g.* the number of packets received by a client. In the latter mode (*trigger*) a thread is created at one or more APs and is identified by a firing condition, *e.g.* the RSSI of one client going below a certain threshold. When such condition is verified, a message is generated by the AP.

In this work we added support for a new *polling*–based primitive allowing the controller to access to the rate adaptation algorithm statistics for a given client. For each supported MCS, the Exponentially Weighted Moving Average (EWMA) of the frame delivery probability and the expected throughput in the last observation window are reported. Moreover, the total number of successful and failed transmissions in the last observation period are also reported. This primitive is used by *SDN@Play Mobile* to gather the link delivery statistics for all the wireless clients involved in multicast transmissions. We remind the reader that this information is maintained by the rate adaptation algorithm implemented by the AP. Therefore, no extra computation is added to the APs logic. More information on the particular rate adaptation algorithm used in our prototype is provided in the next subsection.

The IEEE 802.11k amendment introduces a set of mechanisms to collect WLAN radio measurements. The reports are presented as a request/response procedure in the form of action frames that allows to gather statistical reports from the stations. Whenever a station receives a *Beacon Request* from its serving AP, it must report the information contained in the beacon frames received from other APs of the network in its coverage area. In spite of the improvements, not many commercial devices apart from Apple's ones have support for such features [32]. Therefore, other options to obtain this same information must be explored. In this work, Scapy [33] is used to mock the behaviour of 802.11k. Scapy is a powerful packet manipulation tool whose main capabilities include packet generation and sniffing. Since it offers support for decoding a wide range of network protocols, it becomes a real alternative to gather statistical feedback similar to that offered by IEEE 802.11k. In particular, it makes possible to gather the information provided by a *Beacon Request* given that, among other information, the signal strength of the beacons frames of the neighbouring APs in the same network can be obtained.

B. *Data–path Implementation*

Each AP runs one Click modular router [34] instance implementing the 802.11 data–path. Click is a framework for writing multi–purpose packet processing engines and is used to implement just the wireless client/AP frame exchange, while all the network intelligence is implemented at the centralized controller. Communications between Click and the controller takes place over a persistent TCP connection (*i.e.* the 5G–EmPOWER southbound interface).

Rate adaptation is also implemented in Click using the Minstrel [35] algorithm (ported to C++ from its Linux Kernel implementation). Minstrel operations follow a multi–rate retry chain model in which four rate–count pairs, $r0/c0$, $r1/c1$, $r2/c2$ and $r3/c3$, are defined. Each pair specifies the rate at which a unicast frame shall be transmitted and a fixed number of retry attempts. Once the packet is successfully transmitted, the remainder of the retry chain is ignored. Otherwise, the AP will move to the next pair in the chain. When the last pair has been also tried, the frame is dropped. For each supported MCS, Minstrel tracks the link delivery ratio and the expected packet throughput given the probability of success. Statistics are recomputed every 500 ms. In particular, the rates with the highest throughput, second highest throughput, and highest delivery probability are maintained by Minstrel.

In order to adapt to changes in channel conditions, Minstrel spends part of its time in a so–called *look–around* mode. Specifically, 90% of the time, Minstrel configures the retry chain using the collected link delivery statistics. In the remaining 10% of the time it randomly tries other MCSes to gather statistics. Table III summarizes the criteria used by Minstrel to fill the retry chain in both normal and *look–around* mode.

TABLE III: Minstrel Retry Chain Configuration.

Rate	Look–around		Normal transmission
	Random < Best	Random > Best	
$r0$	Best rate	Random rate	Best rate
$r1$	Random rate	Best rate	Second best rate
$r2$	Best probability	Best probability	Best probability
$r3$	Base rate	Base rate	Base rate

We extended the Click data–path implementation in order to support generalized transmission policies for unicast, multicast, and broadcast addresses as opposed to the original transmission policies that could be specified only for unicast addresses. According to the new transmission policies, the rate adaptation algorithm (*i.e.* Minstrel) will use the first entry in the list of available MCSes if the multicast mode is set to Legacy. Conversely, if the multicast mode is set to DMS, the frame will be duplicated for each receiver in the group and will be fed back to the rate control algorithm which will then apply the unicast transmission policy associated to that receiver.

V. Performance Evaluation

The evaluation presented in this section has been carried out in a real scenario to compare *SDN@Play Mobile* with the multicast scheme defined in the 802.11 standard and with our previous work *SDN@Play*. In [11] *SDN@Play* is compared to the Legacy Multicast and the DMS policies defined in IEEE 802.11. Measurements demonstrate that *SDN@Play* can reduce the network–wide channel utilization by up to 80% while maintaining the required performance level. As opposed to the performance evaluation conducted in [11], in this work we leverage on a larger testbed and we introduce multicast receivers mobility. In this section we shall first describe the testing environment and the evaluation methodology. Then, we will discuss the outcomes of the measurements campaign.

A. Evaluation Methodology

The testbed used for our experimental evaluation is depicted in Fig. 4. The evaluation setup consists of four multicast receivers (MR_i), 3 APs (AP_j), a central controller (C), a video server (S), and an Ethernet switch (SW). The receiver MR_1 is a mobile station, while the remaining three are static.

The measurement campaign is executed over one floor of a typical office environment. During the measurements three receivers ($MR_{2,3,4}$) keep a fixed position, while one receiver (MR_1) moves along a 50 m long corridor. The receiver MR_1 is initially located in close proximity of AP_1 at one end of the corridor (see Fig. 4). Then, the receiver moves from its starting point to the other end of the corridor. The corridor is divided into 10 segments. At the end of each segment the receiver stops for 20 s. This results in an average speed of the mobile client of 0.5 m/s if the stops are not considered.

The scenario presented above is not restricted to office buildings. In fact, a similar video delivery use case applies also to other environments such as conferences, universities, or stadiums. Furthermore, it is important to emphasize that, in contrast to simulations where the mobility model of a station can be precisely controlled, real–world experiments present additional factors that are hard to control. In this regard, the mobility pattern of the receiver has been selected in such a way to improve the reproducibility of the experiments.

The APs are based on the PCEngines ALIX 2D (x86) boards and are equipped with a single Wi–Fi interface (Atheros AR9220 chipset). The AP runs the OpenWRT Operating System (15.05.01) and a Click instance implementing the 802.11 data–path. All experiments are carried out on the 5 GHz band (IEEE 802.11a). The devices running the controller and the multicast receivers are all laptops equipped with an Intel i7 CPU, 8GB of RAM, and running Ubuntu 16.04.1.

During the measurements, a video stream is generated by the video server S and delivered to a group of multicast receivers. The video stream consists of a five minutes sequence encoded using the High Efficiency Video Coding Standard (HEVC) [36] and transmitted using FFmpeg [37]. Two different compression schemes resulting in a final average bitrate of 1.2 Mb/s and 6.2 Mb/s are considered. In this way, it is possible to obtain detailed information regarding how different bitrates determine the performance of the network. Finally, it should be noted that the results presented in this evaluation are also valid for shorter or longer transmissions since the stream duration does not determine the behaviour of the system. Moreover, the resolution and video standard used to encode the sequence is just a way to set the transmission bitrate since other video configurations would only lead to different bitrates. The same applies to other parameters relative to the spatial and temporal aspects of the encoding.

The experiments conducted in this work aim at evaluating how user mobility and bitrate affect the system performance. Conversely, the scalability of SDN@Play was already studied in [11]. Although mobility was not accounted, the conclusions of the previous work are also applicable to the scenario presented in this paper, thus this aspect was left aside in the interest of clarity.

Three different multicast strategies have been considered in this study: *Legacy Multicast*, *SDN@Play*, and *SDN@Play Mobile*. As evaluation metrics we considered delivery ratio and wireless channel utilization. Notice that, since all the experiments are conducted with the wireless interfaces operating in 11a mode, the basic rate used for *Legacy Multicast* is 6 Mb/s. Moreover, in the case of *SDN@Play*, the algorithm has been configured to spend 500 ms in DMS mode and 2500 ms in Legacy mode. Between each measurement the rate adaptation statistics have been cleared. Apart from the multicast video stream, no downlink traffic exists between the APs and the multicast receivers. Consequently, the only opportunity for the Minstrel algorithm to be executed is during the DMS periods. Every measurement has been repeated 5 times to avoid possible fluctuations.

Based on the results obtained in previous experimental analyses we have observed severe performance degradation when the signal quality from an AP is below -75 dB. Similarly, we have noticed that an improvement of 20 dB in terms of signal quality can provide a significant boost in terms of both delivery ratio and channel utilization while at the same time avoiding ping–pong effects.

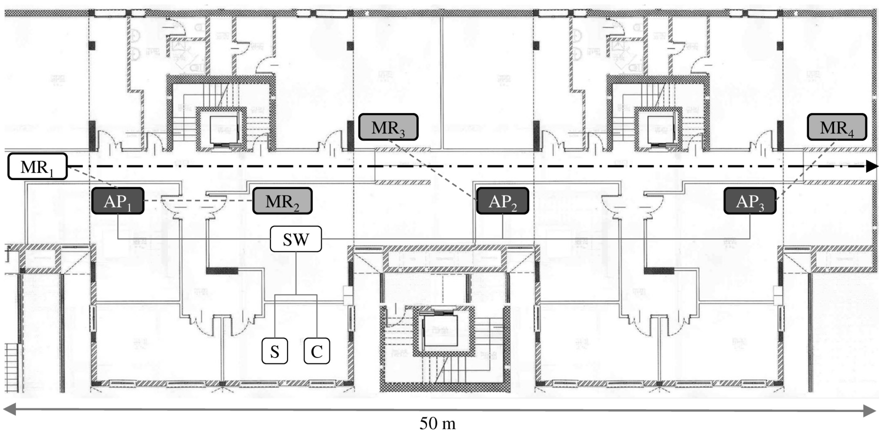


Fig. 4: Testbed deployment layout.

B. Experimental Results

Figure 5 plots the delivery ratio for each receiver using different multicast schemes. At 1.2 Mb/s the performance of the static receivers (MRs 2 to 4) is not affected by the particular multicast scheme. Conversely, *SDN@Play* and *SDN@Play Mobile* provide a significant performance boost to the mobile receiver (MR_1). This is because *Legacy Multicast* cannot adapt to the changing channel conditions experienced by the mobile receiver. Moreover, given the absence of ACKs and retransmissions, the mobile receiver suffers from heavy packet losses as it moves away from AP_1. *SDN@Play* performs better than *Legacy Multicast* since it can configure the multicast rate according to the channel status. Moreover, in DMS mode, *SDN@Play* can retransmit some of the lost frames. Nevertheless, *SDN@Play* does not provide mobility support and, as a result, the mobile station remains attached to the initial AP until the connection is lost and it is reassociated with another AP. By contrast, *SDN@Play Mobile* significantly enhances the performance of the mobile receiver. This is possible due to two main reasons. On the one hand, the algorithm selects the network configuration that minimizes the channel utilization. On the other hand, the receiver is always associated to the AP offering the best channel conditions among the APs that ensure high data rate and good transmission quality. As a consequence, these considerations result in a throughput improvement with regard to the other multicast schemes.

Figure 6 plots the delivery ratio for each multicast receiver using different multicast schemes for a video transmission at 6.2 Mb/s. As can be seen, in the case of *Legacy Multicast* using a video with a higher bitrate results in a sudden performance drop for all the multicast receivers (both static and mobile). The performance drop is particularly significant for the mobile station, which experiences a 70% frame loss ratio. Conversely, *SDN@Play* can improve the performance of the static receivers showing a delivery ratio as good as the one found for the 1.2 Mb/s video. Finally, *SDN@Play Mobile* can improve the delivery ratio of the mobile receiver by 180% bringing it at the same performance level achieved for the 1.2 Mb/s video.

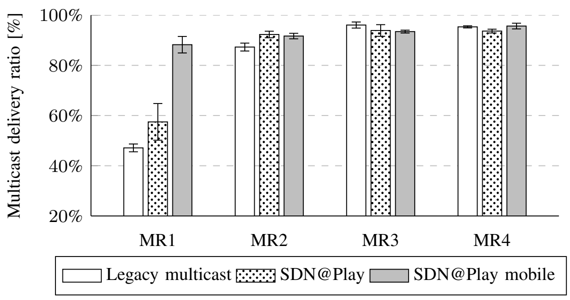


Fig. 5: Delivery ratio for the multicast video transmission at 1.2 Mb/s for each multicast receiver.

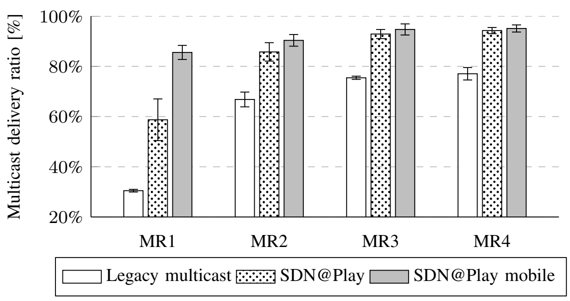


Fig. 6: Delivery ratio for the multicast video transmission at 6.2 Mb/s for each multicast receiver.

Furthermore, given that MR_2 is connected to the same AP than the mobile terminal, it is also worthy to note the performance improvement of *SDN@Play Mobile* with regard to *SDN@Play* for this station. As *SDN@Play* does not provide support for the mobility management and the mobile station keeps attached to the first AP until it losses the connection, it makes a greater number of frames be retransmitted due to the

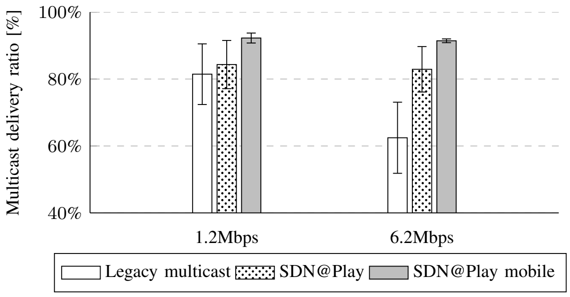


Fig. 7: Network–wide delivery ratio using different multicast schemes at different bitrates.

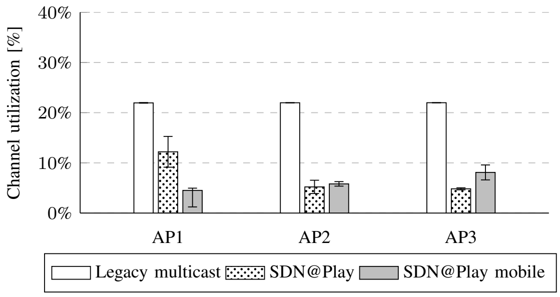


Fig. 8: Channel utilization per AP for the multicast video transmission at 1.2 Mb/s.

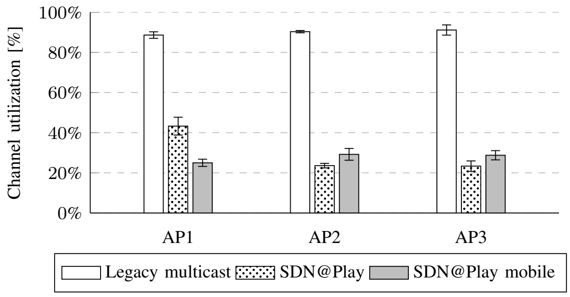


Fig. 9: Channel utilization per AP for the multicast video transmission at 6.2 Mbps.

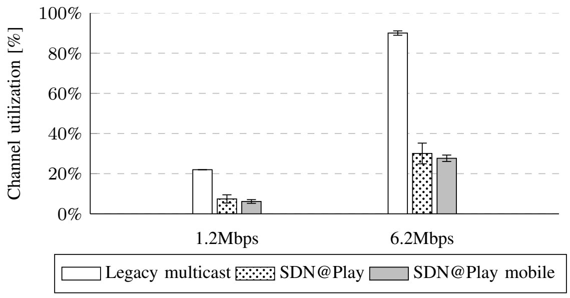


Fig. 10: Network–wide channel utilization using different multicast schemes at different bitrates.

increasing distance and interference. Performing the handover of the mobile receiver when it starts to experience performance drops allows *SDN@Play Mobile* to address this issue and not to impair the quality perceived by the receiver MR_2.

The same behavior can be seen in Fig. 7, which summarizes the average delivery ratio using different multicast schemes and bitrates. Due to the low performance achieved by *Legacy Multicast* and *SDN@Play* for mobile stations, the deviation shown is much higher than the *SDN@Play Mobile* one, which indicates that all the multicast receivers receive practically the same data, regardless of their position.

The fact that *Legacy Multicast* always uses the basic data rates results in a very high channel utilization. As can be seen in Fig. 8, this ratio for the 1.2 Mb/s stream is as high as 20%, while in the case of the 6.2 Mb/s stream (Fig. 9) the utilization reaches 90%, making the channel unavailable for other traffic. By using higher MCS indexes, *SDN@Play* can effectively reduce the channel utilization for both the static and the mobile receivers. This improvement is even more significant in the case of *SDN@Play Mobile*. As a matter of fact, in contrast to the previous case, *SDN@Play Mobile* can specifically address the needs of the mobile receiver by both reducing the channel utilization and balancing the workload across the entire network.

Figure 10 summarizes the network–wide channel utilization using different multicast schemes and bitrates. In this sense, it is shown that both the *SDN@Play* and the *SDN@Play Mobile* multicast schemes achieve a significant reduction in the global channel utilization with regard to *Legacy Multicast*.

Figure 11 plots the instantaneous channel utilization at AP_1 using different multicast schemes. As can be seen, in the case of *Legacy Multicast*, the channel utilization remains constant during the entire transmission. The utilization ratio of this scheme is in most cases higher than the one achieved by the other two multicast schemes. This is due to the fact that *Legacy Multicast* always uses the basic MCS (6 Mb/s in this case). Conversely, when the channel conditions allow it, *SDN@Play* can select higher MCS indexes which in time results in lower channel utilization. However, while the mobile receiver moves away from AP_1, *SDN@Play* is forced to use lower MCS indexes in order to provide the mobile receiver with the expected transmission quality. Eventually, this may lead to choose the basic MCS when the mobile receiver reaches the other end of the corridor. This problem is overcome by *SDN@Play Mobile*, which jointly improves the MCS selection and the receiver association. As can be observed in Fig 11, when *SDN@Play Mobile* is used, the channel utilization of AP_1 remains constant during the entire measurement. The same considerations apply to the scenario with the 6.2 Mb/s video stream (see Fig. 12). However, in this case, *SDN@Play* never reaches the channel utilization of *Legacy Multicast*. This is because the transmission at 6.2 Mb/s makes the channel be fully occupied when the *Legacy Multicast* scheme is used.

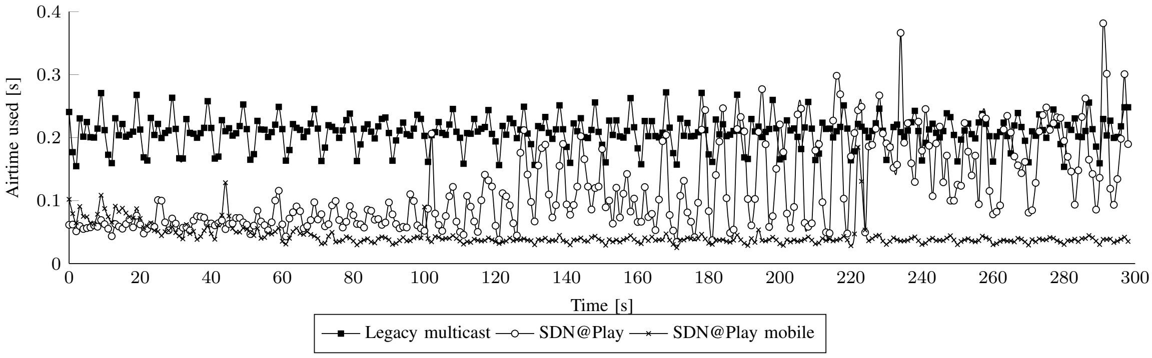


Fig. 11: Channel utilization over time of the AP1 for the multicast video transmission at 1.2 Mbps.

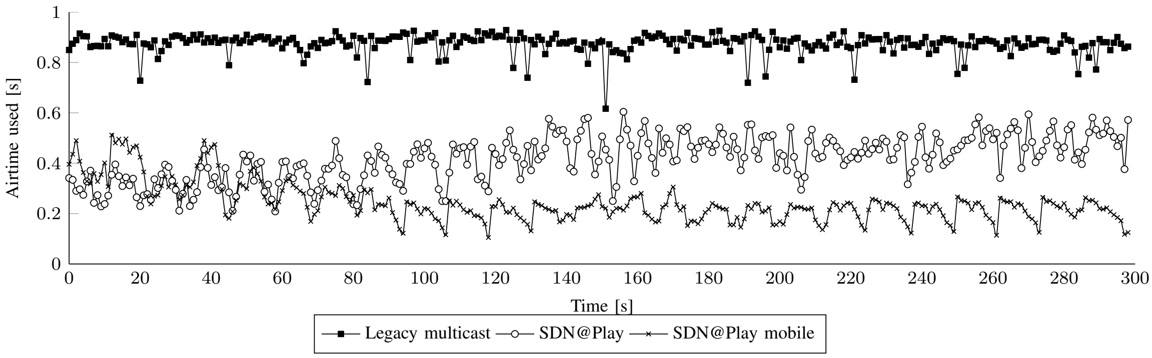


Fig. 12: Channel utilization over time of the AP1 for the multicast video transmission at 6.2 Mbps.

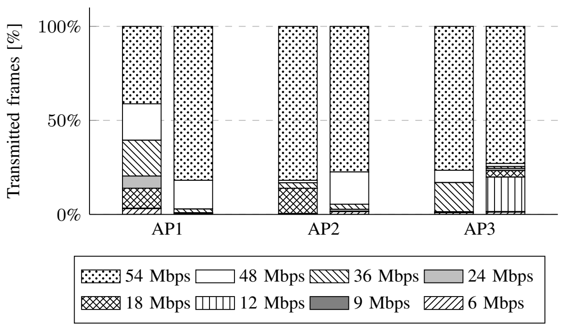


Fig. 13: Distribution of the rates used by *SDN@Play* and *SDN@Play Mobile* per each AP at 1.2 Mbps.

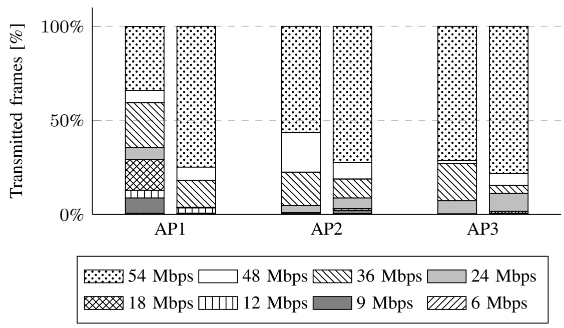


Fig. 14: Distribution of the rates used by *SDN@Play* and *SDN@Play Mobile* per each AP at 6.2 Mbps.

Finally, Figs. 13 and 14 report the distribution of the MCSes used by each AP at 1.2 Mb/s and 6.2 Mb/s, respectively. It should be noted that the *Legacy Multicast* scheme is omitted in this analysis because the lowest MCS index is always used. Although especially at high transmission bitrates *SDN@Play Mobile* selects high MCSes indexes for AP_2 and AP_3 for longer periods than *SDN@Play*, this ratio is considered to be small in comparison with the distribution obtained in AP_1. In this last case, it can be noticed how *SDN@Play Mobile* transmits 70% of the data at the highest MCS (54 Mb/s). This is due to the fact that *SDN@Play Mobile* is able to properly handover the clients to the AP that provides the highest network performance. On the contrary, this value is approximately the half for *SDN@Play* due to the distance of the mobile station from the AP that it is connected to.

VI. Conclusions

In this paper we have presented a novel multicast rate adaptation and mobility management scheme for 802.11–based WLANs. The proposed scheme uses an SDN approach where the global network view available at a logical centralized controller is exploited in order to coordinate the operations of different APs. The scheme, named *SDN@Play Mobile*, jointly optimizes the multicast rate selection *and* the multicast receivers association with the goal of reducing network–wide radio resource utilization while maintaining the expected transmission quality. *SDN@Play Mobile* has been implemented and evaluated over a real–word testbed using the 5G–EmPOWER platform. Experimental measurements show that *SDN@Play Mobile* can deliver a significant improvement in terms of channel utilization compared to the legacy multicast scheme while maintaining full backward compatibility with the 802.11 standard.

As future work we plan to extend *SDN@Play Mobile* to account for multiple multicast groups. Furthermore, we plan to study the behaviour of the system under different situations. This includes analysing the impact of using different values for the delivery probability threshold r_{th} in the MCS selection as well as studying the impact of the other parameters of the algorithm on the network–wide delivery ratio and channel utilization. Finally, we intend to assess the behaviour of *SDN@Play Mobile* under different user mobility models.

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2.5. Efficient Real-Time Content Distribution for Multiple Multicast Groups in SDN-Based WLANs

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- **Impact Factor:** 3.134
- **JCR Ranking:** Q1

Efficient Real–time Content Distribution for Multiple Multicast Groups in SDN–based WLANs

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***Abstract*—Wireless networks research and development efforts are largely driven by the increasing interest in multimedia applications. Video streaming services, which often involve strict Quality of Service (QoS) requirements and are very sensitive to delays, represent a significant proportion of these applications. In IEEE 802.11–based WLANs, these services raise several challenges in terms of robustness, reliability and scalability, specially when supporting multiple multicast streams at the same time. Nevertheless, traditional network architectures make it difficult to address these problems. In this context, the Software Defined Networking (SDN) paradigm opens new research possibilities by decoupling the control decisions from the data–plane and by improving network management and programmability. In this paper, we present *SM-SDN@Play*, an SDN–based solution for joint multicast rate selection and group formation in 802.11–based networks. Experimental results show the high performance and reliability capabilities of the scheme, regardless of the application bitrate, the number of clients, and the number of concurrent multicast streams. Furthermore, the channel utilization is greatly reduced with regard to the standard multicast schemes, which allows other applications to be supported without experiencing a performance degradation. We release the entire software implementation under a permissive APACHE 2.0 license for academic use.**

***Keywords*—Software Defined Networking, WLANs, IEEE 802.11, multicast, rate adaptation, multimedia, video distribution.**

I. INTRODUCTION

The emergence of platforms such as Netflix and Youtube has made multimedia content distribution a popular service in the recent years. Furthermore, it is becoming more common that real–time events such as conferences, social events or educational courses are simultaneously transmitted to a wide range of users. In view of this scenario, multicast communications represent an efficient way of delivering the same information to multiple destinations in a scalable fashion.

The IEEE 802.11 [1] standard is one of the most widespread technologies for the deployment of Wireless Local Area Networks (WLANs) and is found in domestic and professional settings such as enterprises, campuses and hotels. Nevertheless, multicast communications over 802.11–based WLANs incur in severe reliability issues. In fact, due to the lack of acknowledgements and retransmissions, multicast transmissions in 802.11 are performed using the basic Modulation and Coding Scheme (MCS) which results in a high channel occupation. This issue is exacerbated as the applications bitrate increases, and becomes worse in the cases of multiple simultaneous multicast streams. To address these reliability concerns, the IEEE 802.11aa [2] amendment introduces a set of multicast retransmission policies. Nevertheless, no mechanisms for the delivery rate adaptation and multicast group management are specified by the standard. Moreover, due to the widespread use of Wi–Fi compatible devices, IEEE 802.11 amendments aim at maximizing backward compatibility at the expense of innovation. In view of this, Software Defined Networking (SDN) changes the traditional network architecture by effectively decoupling the data–plane from the control–plane and by providing network developers with powerful programming abstractions to affect the state of the network.

The contribution of this paper is twofold. First, we take advantage of our previous work *SDN@Play* [3], a multicast MCS selection algorithm, in order to propose a novel multicast group management scheme. This new scheme, named *Scalable Multigroup SDN@Play* (*SM-SDN@Play*), jointly drives the multicast MCS selection and the multicast group formation in order to minimize the network–wide airtime utilization and maximize the multicast services reliability. Second, we implement and test *SM-SDN@Play* over a real world 802.11–based WLAN and we release the entire implementation under a permissive APACHE 2.0 license for academic use[1].

This work extends [3] in three ways. First, as opposed to *SDN@Play*, the solution presented in this paper independently selects the optimal MCS for each multicast group. Second, *SDN@Play* introduced a two–phase algorithm alternating unicast and multicast periods, however the duration of such periods was static. *SM-SDN@Play*, on the other hand, dynamically adapts the duration of the unicast and multicast periods according to the number of active multicast groups. Third, *SM-SDN@Play* distributes the unicast periods of each multicast group in such a way to minimize the chances that multiple multicast group will operate in unicast mode at the same time. Experimental results show that *SM-SDN@Play* outperforms the standard IEEE 802.11 multicast schemes in terms of both throughput and channel utilization without requiring any change to the wireless clients.

The remainder of this paper is structured as follows. Section II introduces the technical background on multicast communications in 802.11 WLANs. Section III provides an

[1] http://empower.create-net.org/

overview of most relevant related work. In Section IV we introduce the design of *SM-SDN@Play*, while in Section V the implementation details are presented. The results of the measurements campaign are discussed in Section VI. Finally, Section VII concludes the paper pointing out the future work.

II. Technical Background

Multicast transmissions are an efficient way to send the same data to many wireless clients. However, in IEEE 802.11, multicast services are specified as a simple broadcasting mechanism that does not make use of Acknowledgment (ACK) frames. As a result, multicast transmissions are usually performed at the lowest MCS (in order to increase both the range and the reliability of the transmission) and do not use any form of transmission feedback mechanism.

This problem is partially addressed by the IEEE 802.11v amendment [4], where the Direct Multicast Service (DMS) is introduced. DMS replicates each multicast frame into as many unicast frames as the number of receptors in a multicast group. In this way, each frame is retransmitted as many times as required until the Access Point (AP) receives the ACK or the retransmission counter reaches the limit. Although this approach ensures the same reliability level of a unicast transmission, it also presents serious scalability issues as the number of stations in a multicast group increases.

To partially address this scalability limitation, the IEEE 802.11aa amendment [2] introduces the Group Addressed Transmission Service. An in–depth analysis of the performance of this service is carried out by Daldoul *et al.* [5]. The Group Addressed Transmission Service is composed of two mechanisms: DMS and Groupcast with Retries. The latter defines three retransmission methods: Legacy multicast, Unsolicited Retries (UR) and Block ACK (BACK). Legacy multicast is the multicast mode defined in the original IEEE 802.11 standard. Unsolicited Retries specifies a number of retry attempts, N, so that a frame is transmitted $N+1$ times. In spite of increasing the frame delivery probability, this method reduces the network performance due to the retransmission of unnecessary frames. Furthermore, although the stations do not require acknowledgments, this mechanisms still suffers from scalability issues. In the Block ACK method, the AP agrees with the stations the number of consecutive unacknowledged frames. After that, the AP sends a burst of multicast frames up to that number, and requests the Block ACK to each station. Both the request and the ACKs are sent in unicast mode. Although the control traffic is reduced with regard to DMS, the scalability degree of this scheme is also limited.

III. Related Work

The lack of ACKs makes multicast frames in 802.11 to be transmitted at the basic MCS. In this regard, the channel congestion and the QoS restrictions mainly determine the data rate that is selected in most of the related proposals. In spite of achieving higher transmission rates, the performance improvement of all these works usually depends on the size of the multicast group and may suffer from scalability issues. Moreover, many of these works require significant modifications to the wireless clients stack, making them incompatible with the IEEE 802.11 standard. In this section we will summarize such works pointing out in which way our solution improves multicast communications with multiple multicast groups.

Feedback gathering from the stations can be carried out through leader–based schemes. J. Kuri *et al.* [6] and D. Dujovne *et al.* [7] seek to improve the transmission reliability by enabling ACKs for the group leader, which is selected as the receptor exhibiting the worst signal quality. However, a procedure for the leader selection is not provided. Signal–to–Noise Ratio (SNR) in combination with leader–based works have been widely used in the literature. The Auto Rate Selection Multicast mechanism [8] selects the multicast group leader during the first part of the algorithm, while in the second one the SNR obtained from the leader ACKs is considered to adapt the data rate. The SNR–based Auto Rate for Multicast algorithm [9] makes the AP periodically send beacons frames to the multicast stations with the aim of figuring out from their responses the perceived SNR level. Based on this information, the transmission rate is adapted according to the selected leader, which corresponds with the client exhibiting the worst SNR value. Lastly, the Hierarchical Auto Rate Selection Multicast mechanism [10] ensures that the clients under the worst channel conditions receive, at least, the base layer of the video, while the remaining ones also receive some enhancement layers. However, most of these approaches either require to make changes in the 802.11 standard, or they do not specify a procedure for the leader election or need to reach a trade–off between reliability and scalability.

Mathematical and analytical models have been also taken as reference to improve the performance of the multicast transmissions. M. Sun *et al.* [11] propose an analytical model to perform a multicast scheduling by gathering the channel state information and the quality of each Scalable Video Coding layer. The Batch Mode Multicast MAC scheme [12] enhances the network reliability by polling the receptors to obtain individual ACKs, which makes it not scalable to large multicast groups. The Enhanced Leader Based Protocol [13] relies on the use of multiple leaders for the ACKs handling and the Block ACK techniques. However, analytical models are usually applied on a saturated network and make assumptions that are not always met on a real–world scenario.

Research efforts on SDN–based multicasting in OpenFlow [14] networks can also be found. L. Bondan *et al.* [15] introduced a solution for multimedia multicasting based on OpenFlow which aims at calculating the best route between the multicast source and the destinations. Similarly, H. Egilmez *et al.* [16] also aim at enhancing multicast video transmissions by enabling the QoS support at the OpenFlow control layer. The reliability of the multicast traffic is also improved in ECast [17] by means of a novel packet retransmission scheme for packet loss mitigation. OpenFlow is also used by Y. Nakagawa *et al.* [18] to introduce a method for the multicast group management problem. However, these works are targeted at the wired segment of the network and are thus not applicable to the radio access segment.

SDN concepts have been also applied to a few solutions on multicast in WLANs. The work presented by

N. Soetens *et al.* [19] demonstrates how the SDN–based management improves the performance of WLANs. H. Kumar *et al.* [20] and P. Gallo *et al.* [21] provide the users with a set of controls to manage the quality of their services. Lastly, S. Tajik *et al.* [22] present a numerical analysis to improve multicast communications using SDN principles. Nevertheless, the channel occupation could be greater than the Legacy multicast one when the multicast group size increases.

Quality of Experience (QoE) aspects play an important role in multimedia applications. K. Piamrat *et al.* [23] deploy a neural network to map QoE measurements into data rates, while G. Rubino *et al.* [24] present a new hybrid objective–subjective video quality metric. Finally, although some changes in the Linux kernel are required, S. Paris *et al.* [25] also explore this problem in a real–world environment.

When the size of the multicast group grows, some approaches present scalability problems due to the number of retransmissions, the control traffic overhead or the transformation of multicast frames into unicast ones. Scalability issues are partially solved by Y. Sangenya *et al.* [26] through a protocol that improves the delay and frame loss rate of the clients by dividing and scheduling the stations into several groups. AMuSe [27] is presented as an efficient leader–based algorithm to dynamically select a subset of feedback nodes and adjust the bitrate accordingly. Nevertheless, it is assumed that the location of the devices can be estimated. The concept of assisting stations is also presented by Y. Bokyung *et al.* [28]. The AP transmits the data in unicast mode to the client exhibiting the worst channel conditions. However, since the remaining stations need to sniff the ongoing transmission with this client in order receive the multicast stream, the proposed scheme is not compatible with the 802.11 standard.

Scalability issues are exacerbated when considering several multicast groups. Although there is not much research in 802.11, this problem has been studied in WiMAX [29], [30]. P. Sendn-Raa *et al.* [29] propose a group management by comprising in the same group the clients attached to each relay station. Nevertheless, a procedure to schedule the multicast groups is not provided. F. Han *et al.* present a mathematical model [30] where the stations are divided into two groups according to the distance to the Base Station. In this way, two time slots are needed and the data rate is adjusted based on the user with the worst channel conditions in each group. The Multi–View Group Management Protocol [31] analytically intends to facilitate the 3D video transmissions in Wi–Fi multicast. To this end, each view is associated with a multicast group, in a manner that a client may subscribe to a set of views by joining a set of multicast groups.

Despite the progresses made, most of the works are not validated in real–world environments or are not compatible with the IEEE 802.11 standard. Therefore, the lack of practical approaches to address the multicast data rate adaptation in Wi–Fi networks becomes highly noticeable. Moreover, the applications performance depends on several factors such as network congestion and distribution, QoS requirements and multicast group size. As a consequence, integration between rate adaptation features and multicast retransmissions policies while ensuring a high scalability level is still an open issue.

IV. System Design

Enterprise WLANs must support a wide spectrum of services. Nonetheless, the management of both these services and of the network itself becomes more difficult as the number of devices increases. This, along with the difficulty of adding new functionalities to the Wi–Fi MAC layer, has led to the concept of SDN–based WLANs. This new paradigm addresses such limitations by introducing a fully programmable and modular network, making it possible to implement control and management tasks on top of a (logically) centralized control plane instead of implementing them as distributed applications running across the various Wi–Fi APs in the network.

OpenFlow is one of the most widely adopted options to implement the link between the data–plane and the control–plane (the socalled southbound interface). Nevertheless, its features are targeted at wired packet switched networks and are not suited for controlling wireless networks [32]. As a result, in the last years several SDN solutions for wireless and mobile networks have emerged, examples include Odin [33], CloudMAC [34] and 5G–EmPOWER [32].

The work presented in this paper has been implemented taking as a reference the 5G–EmPOWER platform [32]. Nevertheless, it should be noted that, the multicast scheme presented in this work is absolutely general and can be in principle applied to any centrally controlled enterprise WLAN. In this section, we first describe the main features of the 5G–EmPOWER platform. Then we introduce the *Transmission Policy* abstraction designed to allow an SDN controller to reconfigure a Wi–Fi AP rate adaptation policy. Finally, we show how these two abstractions can be used to implement the *SM-SDN@Play* algorithm for multicast groups management.

A. 5G–EmPOWER

5G–EmPOWER is a network operating system for wireless and mobile networks. As shown in Fig. 1, it is composed of three layers: infrastructure, control, and application. The *Infrastructure Layer* consists of a programmable 802.11 data–path (*i.e.* the 802.11 APs). This layer is made up of four main modules, namely *Rate Control Statistics*, *Transmission Policies*, *IGMP Membership* and *Multicast Addresses Management*. The first two modules are used for MCS selection, while the last ones focus on multicast groups formation. These blocks are further described in the following sections. The data–plane network elements in the *Infrastructure Layer* are in constant communication with the (logically) centralized controller situated at the *Control Layer*. Notice that the communication between the data–path is implemented using a custom built protocol. The details of this protocol are out of the scope of the paper and are omitted due to space constrains. However, a full description can be found in [35]. Finally, applications, such as *SM-SDN@Play*, run at the *Application Layer* and leverage on the global network view exposed by the controller in order to implement the network intelligence.

B. The Transmission Policy Abstraction

The fundamentals of SDN call for a clear separation between control–plane and data–plane. This in time requires to

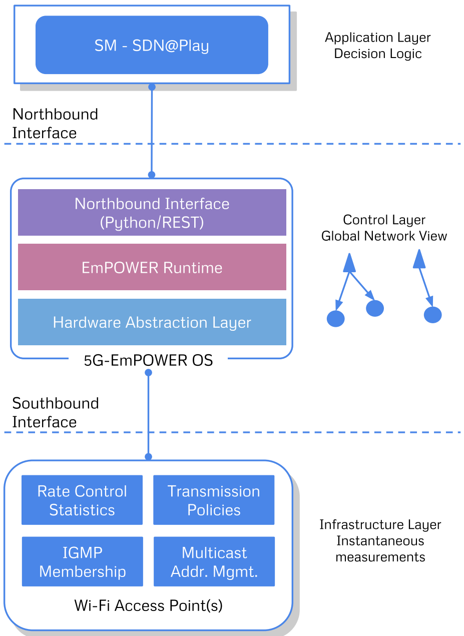

Fig. 1: 5G–EmPOWER System Architecture.

identify how network resources are exposed (and represented) to software modules written by developers and how those can affect the network state. Due to the stochastic nature of the wireless medium, the physical layer parameters that characterize the radio link between a Wi–Fi AP and a wireless client, such as transmission power, MCS, and Multiple Input Multiple Output (MIMO) configuration, must be adapted in real–time to the actual channel conditions. As a consequence, any programming abstraction for rate–adaptation in Wi–Fi networks must clearly separate fast–control operations that must happen very close to the air interface, such as rate adaptation, from operations with looser latency constrains, such as mobility management.

In this work we use the *Transmission Policy* abstraction [3]. A *Transmission Policy* is defined for each $\langle AP, client \rangle$ pair in the network and specifies the range of parameters the AP can use for its communication with that wireless client. Such parameters include:

- *MCSes*. The set of MCSes that can be used by the rate selection algorithm;
- *RTS/CTS Threshold*. The frame length above which the RTS/CTS handshake must be used;
- *No ACKs*. The AP shall not wait for ACKs if true;
- *Multicast policy*. Specifies the multicast policy, which can be Legacy, DMS, or Unsolicited Retries;
- *Unsolicited Retries Count*. Specifies the number of unsolicited retransmissions.

Table I presents three *Transmission Policy* configuration examples for unicast and multicast destination addresses. The first multicast entry (*01:00:5e:b4:21:90*) specifies the usage of Legacy as multicast mode, and 24 Mbps as transmission rate. By contrast, in the second multicast entry (*01:00:5e:40:a4:b4*), the DMS mode is selected. We remind the reader that DMS transmits each frame in unicast mode as many times as the number of receptors in the group. Therefore, the transmission rate is selected from the list of MCSes specified in the *Transmission Policy* of each receptor and the remaining parameters are not applicable.

The *Transmission Policy* configurations are manipulated by the controller via the southbound interface using a CRUD (Create, Retrieve, Update, Delete) model. Due to space constraints the details of the signaling protocol are omitted.

C. Multicast Rate Adaptation

The *SDN@Play* algorithm presented in our previous work [3] uses the unicast link delivery statistics computed at the Wi–Fi AP to calculate the MCS used for a multicast transmission. Notice how, link delivery statistics can only be computed for unicast transmissions. Therefore, *SDN@Play* alternates between the DMS and Legacy multicast modes in order to collect unicast link delivery statistics even when there are no ongoing unicast transmissions between the AP and the wireless clients. The ratio between the DMS and Legacy periods is fully configurable, hence allowing network programmers to trade reliability for channel. Fig. 2 depicts the high level operation of *SDN@Play*.

In the *first phase*, which extends over the shortest period of time, the controller sets DMS as multicast policy for a given multicast address A. This allows the APs to gather the statistical information of all the clients in a multicast group. In the *second phase*, the previous statistics are used to compute the MCS with the highest delivery probability, R_{tx}, for all the stations in the group. Then, the Legacy mode is set as *Transmission Policy* for the multicast address A, and R_{tx} is configured as single MCS for that destination.

Based on the current statistical information, the transmission rate for a certain multicast group is calculated as described below. Let us define M as the set of receptors in a multicast group and let $R(n')$ be the set of MCSes supported by the multicast receptor $n' \in M$. Moreover, let $P_r^{n'}$ be the delivery probability of the MCS index $r \in R(n')$ at the multicast receptor $n' \in M$. Accordingly, R_{valid}, the set of MCS indexes with a delivery probability higher than a given threshold r_{th} for all receptors, can be computed as follows:

$$R_{valid} = \bigcap_{n' \in M} \left\{ r \in R(n') | P_r^{n'} > r_{th} \right\} \quad (1)$$

Following from this result, the multicast transmission rate R_{tx} can be computed as follows:

$$R_{tx} = \begin{cases} max\left(R_{valid}\right) & if\ R_{valid} \neq \emptyset \\ min\left(\bigcup_{n' \in M} \left\{ \underset{r \in R(n')}{\operatorname{argmax}}(P_r^{n'}) \right\} \right) & otherwise \end{cases} \quad (2)$$

Notice how this approach ensures an appropriate data rate even for the clients with poor channel conditions. Furthermore,

TABLE I: *Transmission Policies* Configuration Examples.

Destination	Type	MCS	RTS/CTS	No ACK	Multicast	Unsolicited Retries Count
5c:e0:c5:ac:b4:a3	unicast	6, 9, 12, 18, 24, 36, 48, 54	2436	False	n.a.	n.a.
01:00:5e:b4:21:90	multicast	24	n.a.	n.a.	Legacy	n.a.
01:00:5e:40:a4:b4	multicast	n.a.	n.a.	n.a.	DMS	n.a.

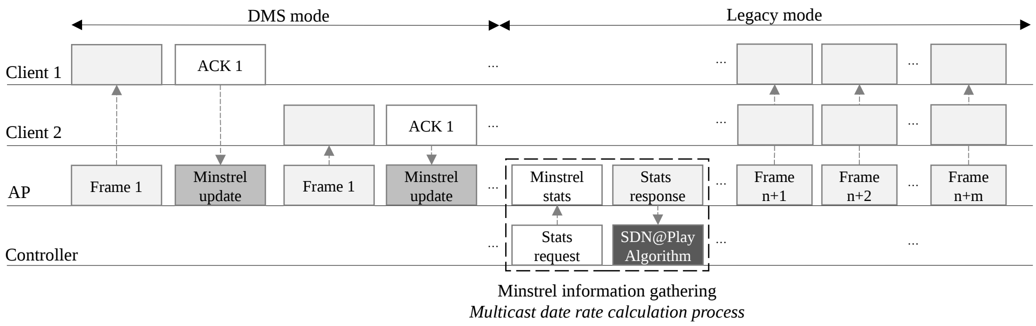

Fig. 2: *SDN@Play*'s scheme. In the *first phase DMS* is used as multicast policy allowing the link delivery statistics gathering. In the *second phase* the policy is switch to *Legacy* and the collected statistics are used to compute the optimal multicast MCS.

if the delivery probability of all MCS indexes is lower than the minimum required reliability level, r_{th}, the algorithm picks for each receptor the data rate that achieves the highest delivery probability. Then, it chooses as multicast rate the most robust one (*i.e.* the lowest) among those rates. Although gathering the link delivery statistics needs some signaling between the APs and the network controller, only a few small changes must be done at the APs and no modifications are required on the wireless clients. Consequently, *SDN@Play* is fully backward compatible with the standard IEEE 802.11 protocol.

D. Multiple Multicast Group Management

Building on the *SDN@Play* algorithm described in the previous section, the *SM-SDN@Play* multicast group management algorithm is introduced in this work. When several transmissions are targeted for multiple multicast groups, an instance of *SDN@Play* must be run for each of them to separately adapt their data rate. Consequently, the lack of coordination among the working phases of *SDN@Play* of each group with regard to the others may make the algorithm very inefficient. Especially, depending on the size of each group, the overlapping of several DMS phases may arise collisions, retranssmissions and performance issues. In other words, if the group management is not performed properly, the DMS phase of some of them may take place at the same time, which would result in a high number of simultaneous unicast transmissions (one for each receptor in each multicast group).

In order to show the importance of an efficient scheduler for the multicast groups, we will use as an example the scenario described below. Let us take 500 ms and 2500 ms for the duration of the DMS and Legacy periods of *SDN@Play*, respectively. In other words, during the first 500 ms the algorithm uses the DMS policy, while the Legacy one is used for the next 2500 ms. This is applied to all the multicast groups in the network. Consequently, in the first phase of the algorithm, the number of simultaneous unicast transmissions will increase with the number of active multicast groups. As described in Section II, DMS has serious scalability problems, which would also affect the performance of *SM-SDN@Play*.

In order to overcome the problem described above, the total length of the two phases, L, is divided into small parts, whose duration corresponds with the duration of the DMS period, dms_d. Let also leg_d be the duration of the Legacy period. Thus, we can define n as the total number of subphases and d_i as the length of each subphase $i \in L$ as follows:

$$d_i = \frac{L}{dms_d} \tag{3}$$

Accordingly, the Legacy period, leg_d, would be composed of $n-1$ consecutive subphases, and can be derived as follows:

$$leg_d = (n-1) \cdot d_i \tag{4}$$

In order to prevent the unicast transmissions (in the DMS phase) of all the multicast groups from taking place simultaneously, the DMS period of each group is set in a different subphase i. When an new multicast group is created, the controller assigns the DMS period of that group to one of the available subphases. This operation is sketched in Fig. 3. As can be observed in the situation displayed in *Step 3.1*, up to 6 multicast groups can be accommodated without overlaps in the DMS phase. This is achieved by using 500 ms and 2500 ms for the duration of the DMS and the Legacy phases, respectively, as stated in the example above.

However, as can be seen in the situation shown in *Step 3.2* in Fig. 3, it may be the case that all the slots are occupied when a new multicast group is created. In view of this, if possible, the duration of each subphase d_i for the DMS period must be recomputed. Let dms_{min} be the minimum amount of time needed to compute the link delivery statistics, and dms_{max}

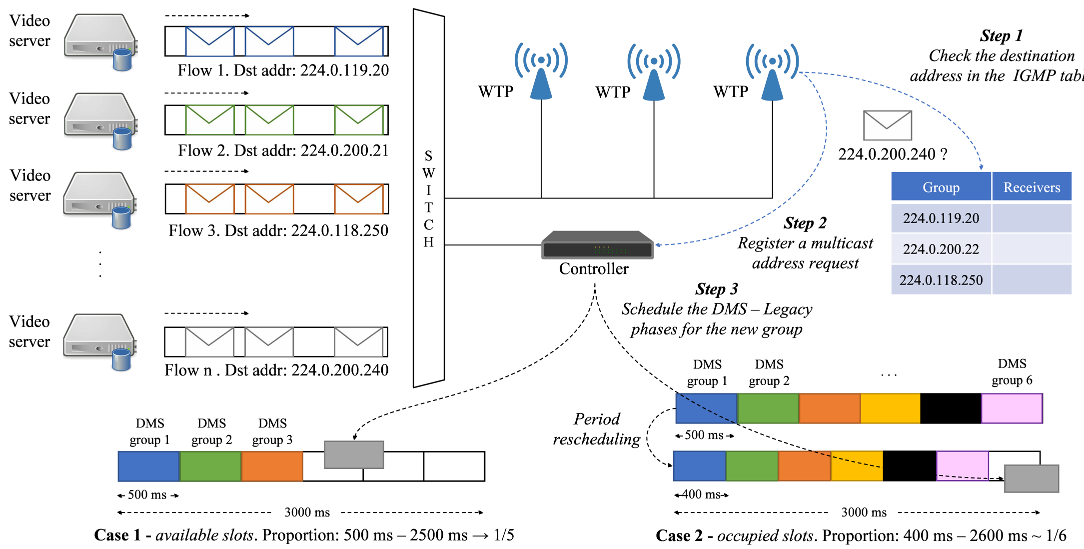

Fig. 3: *SM-SDN@Play* periods management and transmission policies coordination.

the maximum length for the DMS period to avoid causing performance degradation. Moreover, let S be the set of $s = |S|$ multicast groups in the network. Therefore, the new length for the DMS and Legacy periods can be expressed as follows:

$$dms_d = max(dms_{min}, min(dms_{max}, \lceil\frac{L}{S}\rceil)) \tag{5}$$

$$leg_d = max(L - dms_d, dms_{min}) \tag{6}$$

We would like to emphasize that, in some cases, the proportion between the policies ratio and the entire duration, L, may not be exact. In that case, the algorithm will approximate the duration d_i with the aim of not modifying the defined ratio. This phenomenon is also sketched in *Step 3.2* in Fig. 3.

Although it would be an extreme case, in the specific situation of simultaneously managing a huge number of multicast groups, and depending on the duration dms_{min}, it could happen that dms_{min} is equal to dms_d, and hence to d_i. In other words, the protocol subphases cannot be split again. In view of this, the DMS period of a new multicast group would coincide with one of the already scheduled groups. We consider this as an unlikely scenario which in any case would only result in the overlap of a few subphases with a negligible impact on the network performances.

SM-SDN@Play allows to dynamically schedule the DMS periods of the different multicast groups with the aim of avoiding collisions between the unicast transmissions of each group. This approach makes *SM-SDN@Play* suitable for managing huge multicast groups, increases the scalability level with regard to *SDN@Play* and makes it possible to maintain the network throughput.

E. Complexity Analysis

In this section, we would like to analyse the computational complexity of the *SM-SDN@Play* algorithm.

For each multicast frame to be transmitted by an AP the list of active multicast groups must be traversed. As a result, if the number of multicast groups is s, the complexity of this operation is $O(s)$. Notice however that the number of multicast groups is expected to be very small. As a result, this operation complexity can be considered constant.

The complexity of scheduling a new multicast group is also essentially constant. In fact, if a free DMS slot is available, then *SM-SDN@Play* simply assigns the new multicast group to a free slot. Conversely, if a free DMS slot is not available then *SM-SDN@Play* must recompute a new length for both the DMS and the Legacy periods. However, since this operation does not depend on the number of multicast groups nor on the number of active multicast receptors, the complexity of scheduling a new multicast group can be considered constant. Finally, if the periods have been recomputed, the algorithm must iterate through the list of multicast groups to assign the new periods to each of them. Consequently, in the worst case, the computational complexity of recomputing the groups periods is $O(s)$.

In order to compute the list of valid rates R_{valid}, the *SM-SDN@Play* algorithm must first traverse the list of receptors M and for each of them it must then traverse the list of supported transmission rates R. In the worst case the length of this list is mr where m is the number of receptors in the group and r is the number of transmission rates. Such list must then be traversed again in order to find the actual multicast transmission rate R_{tx}. As a result, the overall computational complexity for this operation is $O((mr)^2)$. Notice how this operation is performed once for each multicast group at the end of the group DMS period.

TABLE II: Minstrel Retry Chain Configuration.

Rate	Look–around		Normal transmission
	Random < Best	Random > Best	
$r0$	Best rate	Random rate	Best rate
$r1$	Random rate	Best rate	Second best rate
$r2$	Best probability	Best probability	Best probability
$r3$	Base rate	Base rate	Base rate

V. Implementation details

A. Statistics gathering

The 5G–EmPOWER platform provides a full set of programming primitives for the network management trough a Python–based SDK [32]. These primitives can be used in *polling* or *trigger* mode. The *polling* mode allows the controller to periodically poll the APs for specific information, while in the *trigger* mode this information is sent by the APs to the controller when the firing condition is verified.

In this work, the *polling*–based primitives presented by E. Coronado *et al.* [3] are used to collect the rate adaptation algorithm statistics for a given multicast receptor. This information includes, for each supported MCS, the Exponentially Weighted Moving Average (EWMA) of the frame delivery probability, the expected throughput, and the number of successful and failed transmissions in the last observation window. This primitive is used by *SM-SDN@Play* to periodically gather and update the link delivery statistics of all the receptors in a multicast group. This information is updated by the MCS selection algorithm implemented by the AP. Therefore, no extra computation complexity is added to the APs.

B. Data–path Implementation

APs are composed of one OpenvSwitch [36] instance for the wired backhaul and one Click modular router [37] instance for the 802.11 data–path implementation. In this work, Click is used to handle the clients/APs frame exchange, while the remaining network intelligence is managed by the controller. The controller communicates with Click via the southbound interface through a persistent TCP connection.

The MCS selection mechanism is implemented in Click using the Minstrel algorithm [38]. Minstrel follows a multi–rate retry chain model where four rate–count pairs, $r0/c0$, $r1/c1$, $r2/c2$ and $r3/c3$ are defined, as shown in Table II. They specify the rate that must be used to transmit a given number of retry attempts. If a frame is successfully transmitted, the remaining part of the retry chain is ignored. Otherwise, the next pair is used until the frame is properly transmitted or is finally dropped. To adapt to channel conditions, the statistics are recomputed every 500 ms. Minstrel spends the 90% of the time using the collected link delivery statistics to configure the retry chain, while in the remaining 10% of the time, other MCSes are randomly selected to gather new statistics.

For a multicast address, Minstrel will use the first MCS in the list if the retransmission mode is set to Legacy. If the policy is set to DMS, the entry is ignored and the policy associated to each receptor is used instead. Finally, if the Unsolicited Retries mechanism is selected, the frame is sent N times at the specified rate.

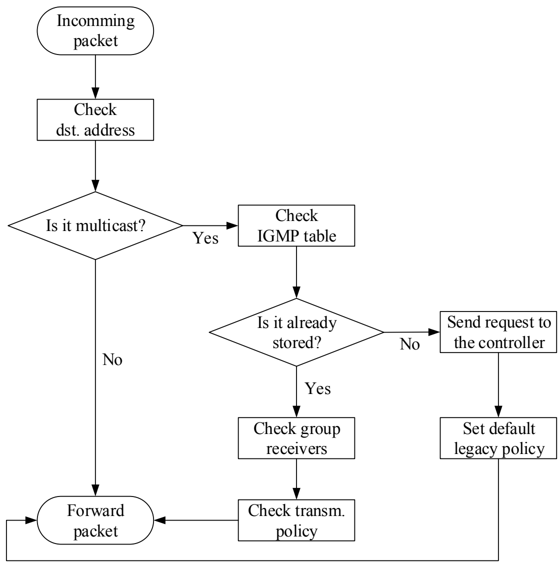

Fig. 4: *SM-SDN@Play* AP flowchart.

C. Transmission Policy Abstraction

The *Transmission Policy* abstraction is exposed to the programmers to configure the delivery features of a destination address through the `tx_policy` property of a *Resource Block* object. A *Resource Block* is the minimum allocation block in the network and is defined as a 2–tuple $\langle f, b \rangle$, where f and b are, respectively, the center frequency and the band type. Therefore, the each AP has as many *Resource Block*s as the number of installed Wi–Fi interfaces.

The *Transmission Policy* configuration only requires to specify the information for the MCS and multicast policy. The following example shows the configuration needed to set the DMS retransmission policy for the *01:00:5e:00:00:fb* address:

```
txp = block.tx_policies['01:00:5e:00:00:fb']
txp.mcast = TX_MCAST_DMS
```

In a similar manner, the `tx_policy` can be reset to the Legacy mode, for which the new multicast rate is also defined:

```
txp = block.tx_policies['01:00:5e:00:00:fb']
txp.mcast = TX_MCAST_LEGACY
txp.mcs = [24]
```

This solution is directly extensible to *SM-SDN@Play* given that it easily allows the specification of a different policy for each multicast group without introducing extra complexity.

D. Multicast Groups Management Abstraction

In this work, a *Multicast Group Management* abstraction is introduced to properly handle the stations requests to join or leave a certain multicast group. To achieve this goal, both the APs and the network controller are involved. However, APs merely forward the information to the controller, which is in charge of making the corresponding decisions.

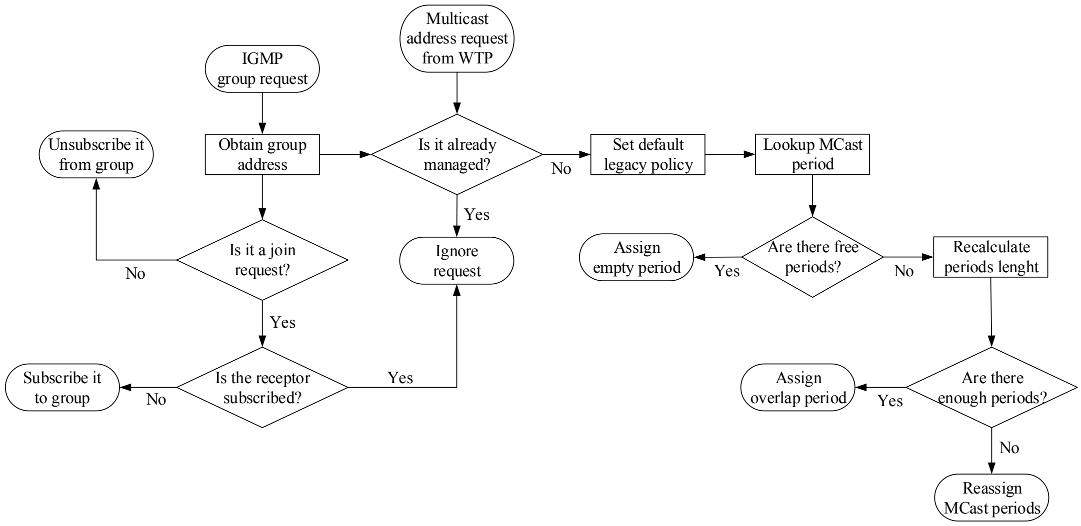

Fig. 5: *SM-SDN@Play* controller flowchart.

When an AP receives a multicast frame, it must check if there is already defined a forwarding rule for that multicast address. The flowchart followed by an AP is shown in Fig. 4. When the frame comes from an station that wants to join a multicast group, an Internet Group Management Protocol (IGMP) frame is also received. This management frame stores the multicast address and the IGMP request type, which mainly corresponds with *join* and *leave* requests. To this end, the *IGMP table* object is defined, as depicted in *Step 1* in Fig. 3. This structure includes the multicast addresses in use and the receptors of each group. On the one hand, if the group is already registered in the table, it means that it is already managed by the controller. Therefore, the receptors subscribed to the group are directly obtained and the frame is forwarded using the *Transmission Policy* defined for that address. On the other hand, if none of the entries corresponds with the group address, the request is sent to the controller, as depicted in *Step 2* in Fig. 3. While this request is being processed by the controller in *Step 3* to schedule the DMS phase of the new group, the frame is transmitted using the Legacy policy.

At the controller side two types of inputs can be distinguished, as can be seen in Fig. 5. The controller may detect a multicast transmission to which there are no clients subscribed yet or receive IGMP requests from the AP for a group inclusion or exclusion of a certain client.

When a new multicast address request is received, the Legacy multicast *Transmission Policy* is temporarily specified. Then, the controller must look for an available DMS period for this group, as described in Subsection IV-D. However, if all the slots are occupied, the protocol periods must be recalculated. If after this procedure, there are still free slots, the multicast group is scheduled in the first available one. Otherwise, it means that there is a huge number of multicast groups and this one must be scheduled in conjunction with another one.

The controller can also receive IGMP requests. On the one hand, if the multicast group specified in the request is already managed, the controller checks only the request type. The request could come from a client already subscribed to a group or from a new one. Depending on this fact, the controller will register it as a group member or ignore the request. On the other hand, a client could send a request for a multicast transmission that has not started yet. Then, in addition to the previous procedure, the operations described above for a new multicast address scheduling must be also performed.

VI. Performance evaluation

The performance evaluation presented in this section has been carried out from two points of view to show the scalability level of *SM-SDN@Play* and how efficient it is in managing multiple simultaneous multicast applications. This evaluation is performed in a real–world scenario, and establishes a comparison between our proposal, and the Legacy multicast and DMS schemes defined in the IEEE 802.11 standard. In the next subsection we will describe the characteristics of the scenarios. Then, an in–depth analysis of the results obtained during the measurements campaign will be shown.

A. Evaluation Methodology

The testbed used for the evaluation is displayed in Fig. 6 and is composed of an AP, the 5G–EmPOWER controller, a video server and a set of multicast receptors (MRx). All these devices, apart from the APs, are Dell-branded laptops powered by an Intel i7 CPU, equipped with 8GB RAM memory modules and running Ubuntu 16.04.01.

The AP is built upon a PCEngines ALIX 2D (x86) board, to which a Wi–Fi card based on the Atheros AR9220 chipset is connected. This AP uses the OpenWRT Operating System (15.05.01 version) and runs a Click instance for the 802.11

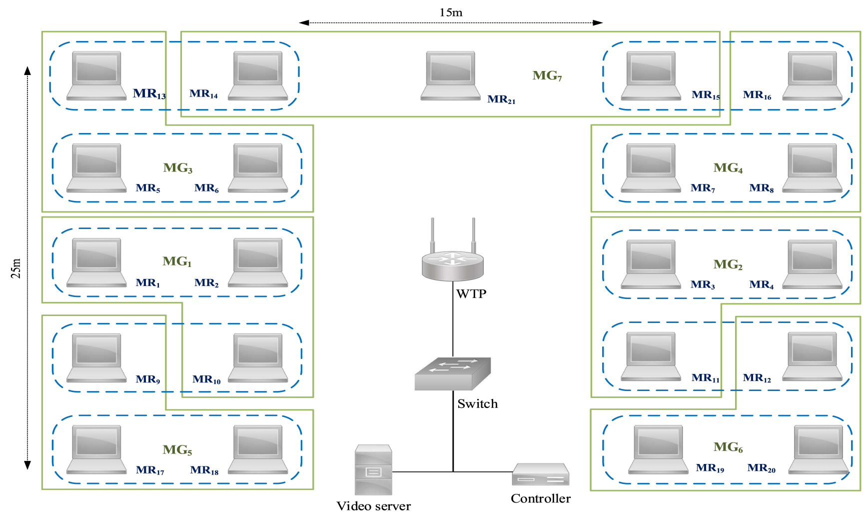

Fig. 6: Testbed deployment layout. Groups marked in blue color corresponds with the *Scalability Analysis*, whereas green markings are related to the *Multiple Groups Analysis*.

data–path. The multicast clients are widely distributed over the network coverage and are divided into groups according to the analysis type. Notice that in Fig. 6 the clients distribution corresponds to the *Scalability Analysis* when they are surrounded by blue marks. By contrast, the clients distribution in green corresponds to the *Multiple Groups Analysis*.

In the *Scalability Analysis*, a variable number of multicast receptors, ranging from 2 to 20 in steps of two stations, has been considered. The server generates and transmits a video streaming application that is delivered to the multicast receptors. This application consists on one minute video sequence with a resolution of 1920×1080 encoded using the High Efficiency Video Coding Standard (HEVC) [39] and transmitted using FFmpeg [40]. This video has been encoded making use of two different compression levels, resulting in 1.2 Mbps and 6.2 Mbps bitrate transmissions. This allows us to test how the network performance is determined by different traffic loads. Due to space limitation it was not possible to report on the *SM-SDN@Play* performance using different videos and/or compression schemes.

In the *Multiple Groups Analysis* a variable number of multicast groups is considered. This number ranges from 1 to 7 groups, each of them being made up of three receptors. The same one minute video sequence encoded for the *Scalability Analysis* is used. However, in this case, the video server transmits this video at 1.2 Mbps as many times as the number of multicast groups. Moreover, since the effect of using different bitrates has been already shown in the previous analysis and those results can be equally applied to this one, it is omitted in this test due to space constraints. Notice how we decided not to change the number of receptors involved in the experiment given that the goal of this section is to demonstrate the scalability of *SM-SDN@Play* for an increasing number of multicast groups. The scalability of the scheme for an increasing number of receptors was already studied in [3].

These scenarios have been considered for both analyses: Legacy multicast, DMS and *SM-SDN@Play*. The tests are performed in the 5.2 GHz band using the 802.11a physical layer. We remind the reader that Legacy multicast transmissions are carried out at the basic rate. Hence, due to the selected physical layer, Legacy transmissions will be sent at 6 Mbps. For *SM-SDN@Play* the duration ratio between the DMS and the Legacy phases has been set to $(500, 2500)$ ms, respectively. In order to show the analysis outcomes, we have selected as metrics the normalized throughput, the channel occupancy ratio and the percentage of retransmitted frames. The link delivery statistics have been cleared after each test. The multicast application is the only transmission that takes place in the network. This ensures that, in *SM-SDN@Play*, statistical data from the Minstrel algorithm can be only gathered during the DMS period. Finally, it should be noted that the experiments have been conducted within a 95% confidence interval and repeated 10 times to avoid possible fluctuations.

B. Experimental Results

1) Scalability Analysis: To ensure a proper QoS level, a high throughput must be achieved in video applications. Fig. 7 shows the average normalized throughput for the multicast schemes with an increasing number of receptors transmitting a video application at 1.2 Mbps. The performance of the Legacy multicast and *SM-SDN@Play* schemes remains practically constant in the $96 - 100\%$ interval. Conversely, the performance of DMS is highly damaged when increasing the number of receptors. Although at the beginning its performance is similar to the one achieved by the other schemes, it is slightly below 90% from 8 to 12 receptors. Moreover, it is highly degraded from the point in which 14 clients are considered, which shows the serious scalability issues of DMS.

Although Legacy multicast provides good delivery throughput with low bitrates, using a basic rate for all the transmis-

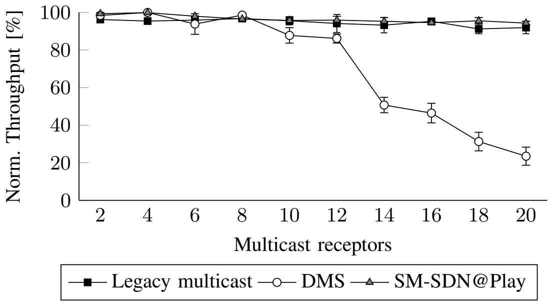

Fig. 7: Normalized throughput for an increasing number of multicast receptors using a video transmission at 1.2 Mbps.

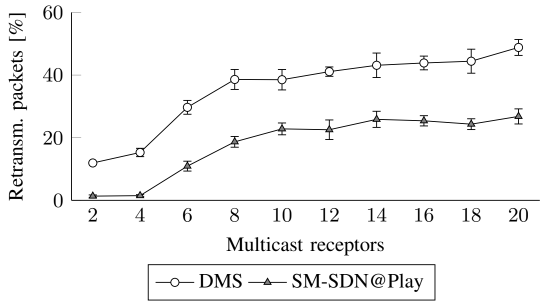

Fig. 9: Retransmitted packets for an increasing number of multicast receptors using a video application at 1.2 Mbps.

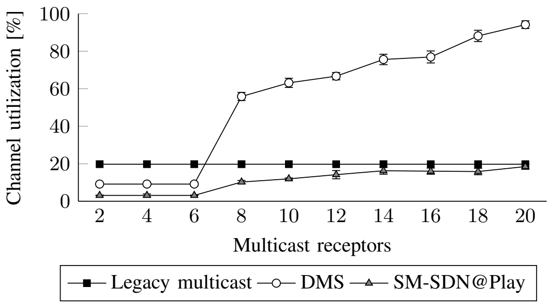

Fig. 8: Channel utilization for an increasing number of multicast receptors using a transmission at 1.2 Mbps.

sions results in a high channel utilization. In Fig. 8 it can be observed how this ratio is around 20% for a video streaming at 1.2 Mbps. In the DMS case, the channel utilization increases with the number of receptors. This issue is due not only to the increasing number of simultaneous unicast transmissions but also to the growing percentage of retransmissions. In the case of the application at 1.2 Mbps, the channel utilization becomes higher from the moment in which the network is made up of 8 receptors until the end of the measurements, when this ratio reaches 90%. By contrast, the channel occupancy ratio of *SM-SDN@Play* remains below the one achieved by the standard schemes in all the cases. The use of higher MCS indexes with regard to Legacy makes it possible to reduce the period of time that the channel is busy. Moreover, given that it only uses DMS in the smallest phase of the algorithm, the channel occupancy ratio is also lower than the DMS one. Specifically, this ratio is under 10% until the half of the test and is below 20% even for 20 receptors. These results show that, although the channel utilization of *SDN@Play Mobile* increases with the number of receptors, this growth is far lower than the DMS one and it does not raise scalability problems.

The enormous number of simultaneous transmissions sent in DMS causes an increase in the retransmission ratio. This effect is shown in Fig. 9, where it is appreciable how this ratio is over 50% when using a wide range of receptors. In contrast, using the DMS policy in only a small part of its protocol makes the retransmission ratio of *SM-SDN@Play* not exceed 20%. In Fig. 10 the behavior of this ratio over time for a 1.2 Mbps transmission is reported. Notice how when using *SM-SDN@Play*, the retransmitted packets ratio remains constant over the test. Nevertheless, a great amount of information is retransmitted by DMS, which becomes even higher when the network is saturated. Specifically, in Fig. 10 can be seen that this figure arises from the half of the test when holding 10 receptors and it practically reaches 50% from the beginning of the transmission for 20 receptors. We would like to stress that since in Legacy multicast packets are neither acknowledged nor retransmitted, the retransmission ratio analysis is omitted for this scheme in all the experiments.

Fig. 11 displays the rates distribution used by DMS and *SM-SDN@Play* for a 1.2 Mbps service. Notice that the information of the Legacy multicast scheme is omitted as it always uses the 6 Mbps basic rate. DMS uses higher MCS indexes than *SM-SDN@Play* until the number of retransmissions arises from 14 receptors. The use of acknowledgments and the constant update of the link delivery statistics make this possible for DMS. Furthermore, in order to achieve a high reliability level, the MCS index is decreased in *SM-SDN@Play* if the success delivery probability does not exceed a given threshold. In the case of *SM-SDN@Play*, the percentage of transmissions in which high data rates, such as 54 and 48 Mbps, are used is as high as 66% with a 95% CI [61.55% - 72.04%].

This analysis has shown how *SM-SDN@Play* improves the efficiency of the multicast video transmissions. Due to the lower channel utilization, our proposal may allow a greater amount of information to be simultaneously transmitted. In order to show this effect, the multicast schemes are also evaluated using a bitrate of 6.2 Mbps with the aim of checking how it compromises their performance and efficiency.

The outcomes for a 6.2 Mbps multicast application present some differences with regard to the previous analysis. First, DMS impairs the network throughput in an earlier stage. As can be seen in Fig. 12, this value is below 30% with only 6 receptors. Meanwhile, although the performance of Legacy multicast keeps constant, it is not able to achieve a normalized throughput higher than 70%. By contrast, *SM-SDN@Play* is only slightly degraded with respect to the previous test and

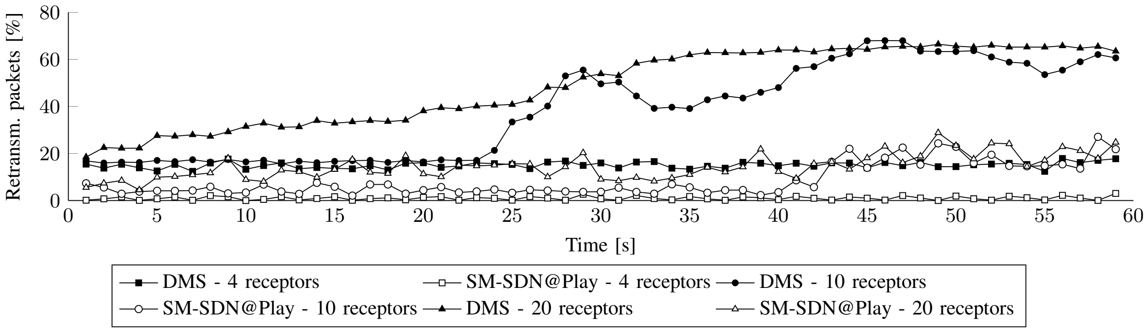

Fig. 10: Retransmitted packets over time using a multicast video application at 1.2 Mbps.

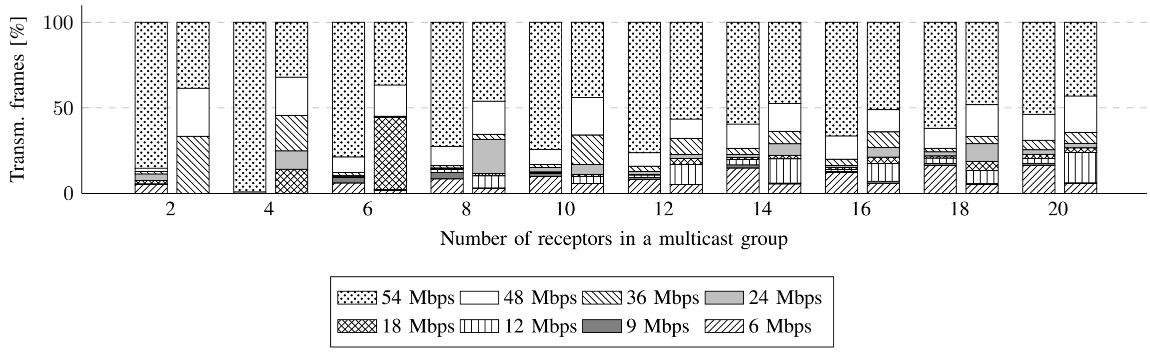

Fig. 11: Rates distribution corresponding to the DMS and *SM-SDN@Play* schemes in a multicast transmission at 1.2 mbps with an increasing number of receptors.

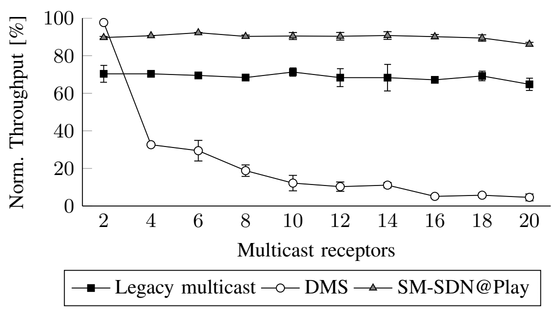

Fig. 12: Average normalized throughput for an increasing number of receptors using a video transmission at 6.2 Mbps.

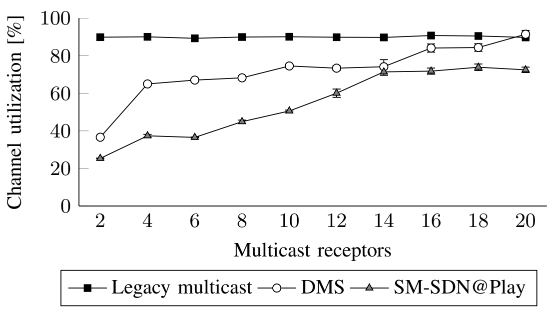

Fig. 13: Channel utilization for an increasing number of multicast receptors using a transmission at 6.2 Mbps.

outperforms the other schemes with a normalized throughput higher than 90%, regardless of the number of receptors.

As mentioned above, the basic rate used by Legacy multicast makes the channel be busy for long periods of time. The channel utilization shown when analyzing a 1.2 Mbps transmission significantly arises until reaching a value close to 90%, as plotted in Fig. 13. This proves that despite its performance, this scheme is not suitable for applications with a high bitrate and impairs the performance of other transmissions in the network. Similarly, the DMS problem is exacerbated in this scenario, in which it makes the network become saturated in an earlier point. When the video bitrate is increased, the channel utilization of *SM-SDN@Play* also arises with regard to the first scenario. However, this ratio allows the video to be delivered without loosing a significant part of the information.

The retransmissions issue becomes even worse when DMS needs to handle a 6.2 Mbps multicast service. In Fig. 14 can be seen that this ratio arises from the half of the test when

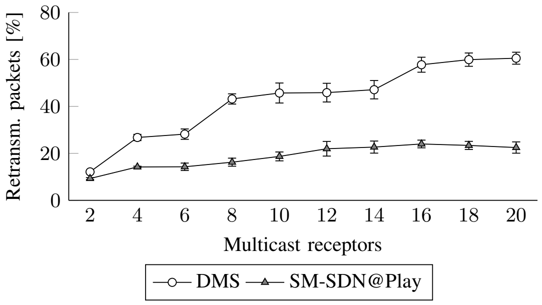

Fig. 14: Retransmitted packets for an increasing number of multicast receptors using a video application at 6.2 Mbps.

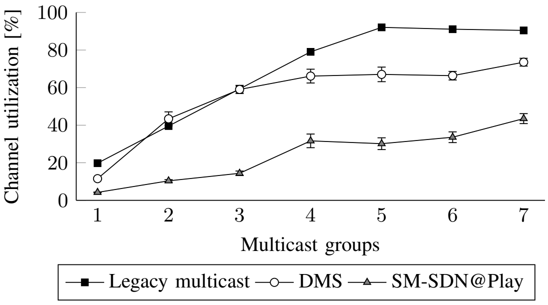

Fig. 16: Channel utilization for a transmission at 1.2 Mbps targeted at multiple multicast groups.

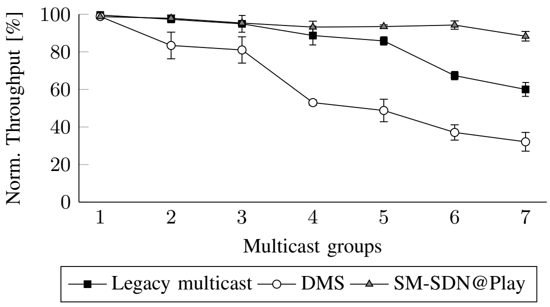

Fig. 15: Average normalized throughput for a video transmission at 1.2 Mbps targeted at multiple multicast groups.

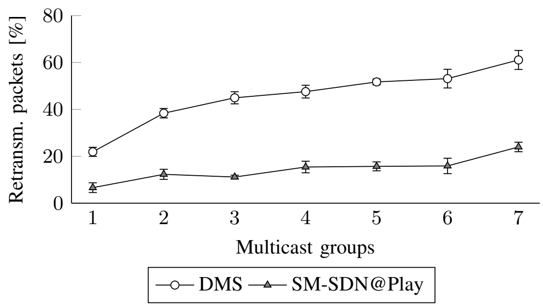

Fig. 17: Retransmitted packets for an increasing number of multicast groups using a video application at 1.2 Mbps.

holding 10 receptors and it practically reaches 70% from the beginning of the transmission for 20 receptors. By contrast, small differences can be found for *SM-SDN@Play*, whose retransmission ratio stands at 20%, irrespective of the bitrate.

2) Multiple Groups Analysis: After studying the scalability level of the proposal, a similar analysis is performed to evaluate its efficiency when managing multiple multicast groups.

Fig. 15 reports the average normalized throughput of the evaluated schemes upon an increasing number of multicast groups. This shows how the performance of Legacy multicast is highly degraded with regard to the case of a single multicast group. We remind the reader that, for a 1.2 Mbps application, the channel utilization of this scheme is around 20%, which makes it be practically saturated when 4 groups are managed. Hence, a throughput fall can be appreciated from this point for the Legacy mechanism. The performance of DMS is similar to the one provided in the previous analysis. However, it is also slightly reduced due to the increase in the simultaneous transmissions and the need to forward them accordingly. Conversely, the normalized throughput of *SM-SDN@Play* remains practically constant and similar to the figure obtained in the single group analysis ($96-100\%$) until practically the end of the measurements, where the performance is slightly impaired by the amount of traffic in the network.

Closely connected to the previous metric, Fig. 16 plots the channel utilization of each scheme. In contrast with the first analysis, where the channel occupancy ratio remained constant for the Legacy multicast scheme, in this case it proportionally arises with the number of multicast groups. The ratio achieved by DMS is similar to the one obtained in the single group analysis, until the AP is completely saturated and is not able to forward on time all the frames. At this point, the period of time that the channel is busy by *SM-SDN@Play* falls far short of the remaining schemes. In fact, the channel occupancy ratio is only slightly risen with regard to the *Scalability Analysis* and it only reaches a 40% utilization ratio when managing 7 simultaneous multicast transmissions.

The retransmissions distribution of DMS and *SM-SDN@Play* is depicted in Fig. 17. This view is almost equal to the one observed in the first analysis of the evaluation for both schemes. The only small difference can be seen for *SM-SDN@Play* when the network holds 7 multicast groups, when the percentage of retransmitted packets lies minimally above 20%.

Finally, Fig. 18 displays the distribution of the MCSes used by DMS and *SM-SDN@Play*. We remind the reader that, once again, Legacy multicast rates distribution is omitted. Figures obtained for DMS are practically the same as presented above given that, regardless of the multicast group a receptor belongs to, the unicast transmission for each of them is forwarded independently from the remainder. However, it is worthy to highlight that in *SM-SDN@Play* the percentage of the frames

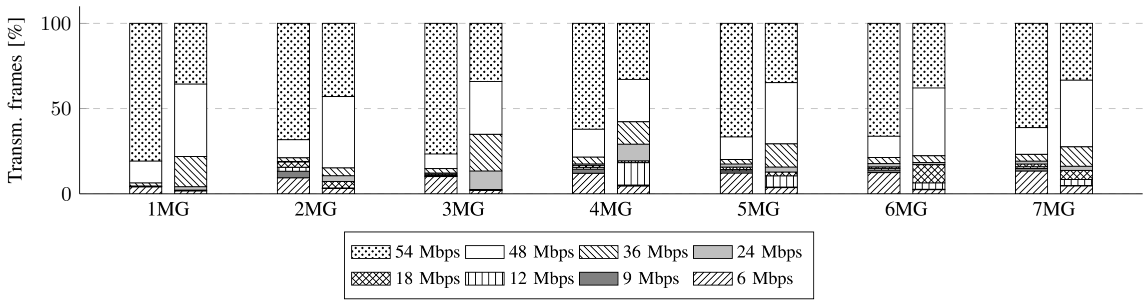

Fig. 18: Rates distribution corresponding to the DMS and *SM-SDN@Play* schemes based on a multicast video transmission at 1.2 Mbps for an increasing number of multicast groups.

that is transmitted using high data rates (both 48 and 54 Mbps) is, in some cases, higher than in the *Scalability Analysis*, reaching it up 83% with a 95% CI [80.22% - 86.29%]. This is due the fact that the data rate of each group is independently calculated only considering the receptors in that group, which allows it to provide more accurate results.

VII. Conclusions

In this work we have proposed *SM-SDN@Play* as a novel solution for multicast group management in SDN–based WLANs. *SM-SDN@Play* is fully backward compatible with the 802.11 standard and does not require any change to the wireless clients. Only minimal changes to the APs are needed.

The performance of *SM-SDN@Play* has been evaluated in a real–world scenario implemented over the 5G–EmPOWER platform and compared with the one achieved by the standard DMS and Legacy multicast schemes. The results prove that, in contrast with the standard mechanisms, our proposal scales properly with respect to both the number of receptors in a multicast group and the number of multicast groups.

A particularly important open issue regards the security and performance isolation properties of the *SM-SDN@Play* scheme. Further studies are needed here in order to properly assess the impact of a misbehaving multicast stream on the other groups. Moreover, we also plan to extend *SM-SDN@Play* in order to make use of Scalable Video Coding and investigate the video quality layers prioritization according to the channel status of the multicast groups.

Acknowledgements

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CHAPTER 3

Conclusions and Future Work

3.1. Conclusions

Wireless technologies have come to stay, fueled by the digital world in a global culture that expects instant access to information. The explosive increase in Wi-Fi enabled devices and the changes in the behavior of the users have spawned a greater need for constant connectivity. In this respect, the versatility provided by 802.11 networks, together with their reliability, high speed and performance, are making WLANs available not only in residential environments, but also in a large range of scenarios such as offices, airports, conferences and public venues.

Wireless networks are evolving rapidly to provide more flexible approaches. However, in addition to offering high-density Wi-Fi, there is an incessant demand for multimedia facilities due to the rapid penetration of digital content across multiple devices. In this regard, high definition video streaming services have generated considerable interest. Therefore, accommodating these latency sensitive applications requires new and innovative techniques in network management to deliver revolutionary network services and enhance the end-user experience.

In this Doctoral Thesis it has been demonstrated that QoS in WLANs is subject to many constraints such as the number of users per AP and specific application requirements. This landscape is complicated in the particular case of multimedia communications with the need for providing mobility support. Although it has attracted significant research interest, an analysis of the literature has shown that accomplishing these quality requirements is not trivial when aiming at designing standard compliant solutions. In view of this, in this Thesis, the QoS provisioning problem has been tackled from two main angles. On the one hand, we have focused on the MAC layer level with the purpose of improving the QoS differentiation provided by IEEE 802.11e. On the other hand, we have leveraged the hardware abstractions provided by the SDN paradigm to address the limitations derived from the ossification of traditional network architectures.

In the first part of the Thesis, we have seen that the EDCA function can only deliver high performance in scenarios with a low traffic load. However, voice and video services are highly impaired in the opposite scenarios, especially in the presence of legacy stations without QoS support. From the simulation results we can conclude that this issue is mainly caused by the number of collisions between voice and video streams given that EDCA uses static values for the medium access parameters regardless of the network conditions. Even though the 802.11 standard does not specify any procedure for selecting these values, they can be periodically updated through beacon frames. However, given that this process must be performed in real-time, the solutions introduced for this purpose must be as simple as possible.

In order to meet these restrictions, we have relied on machine learning algorithms to identify the factors that can provide a good approximation of the network status while avoiding excessive complexity. This information is later used for designing a simple predictive algorithm capable of adapting the medium access parameters in EDCA over time. This algorithm is implemented at the APs in an incremental way to tune the AIFSN and CW parameters using only the channel utilization statistics as a basis for the decision. The results obtained via simulation showed that this approach makes it possible to reduce the collisions and retransmissions in the network. As a consequence, this fact leads to improvements in the QoS provided, which is directly shown by a performance increase of 20% on average. In fact, this improvement increases to 30% in the absence of stations unable to use QoS features.

This proposal can be applied directly to commercial devices since changes in the 802.11 protocol are not required. However, maintaining compatibility with the standard restricts the scope of further enhancements. This limitation can be overcome through the network management simplification provided by SDN by clearly decoupling the data-plane from the logic of the control-plane. SDN is a widely adopted technology in the wired domain. By contrast, the inherent complexity of wireless networks, involving aspects such as client mobility and load balancing, has led to this topic not receiving the same research interest. However, network component virtualization allows for more flexible and faster innovations, and this has recently started to attract sustained research attention in the wireless portion of the network with a view to tackling challenging operations in the Wi-Fi architecture.

In the second part of the Thesis we have shown how network programmability makes it possible to address specific problems in QoS provisioning in the delivery of multimedia content over WLANs. More specifically, in several real-world scenarios we have studied two principal aspects that have been particularly difficult to deal with in the current monolithic Wi-Fi architectures: network resource allocation and multicast services distribution.

Resource allocation is a challenging operation due to varying channel quality and interference. Moreover, load imbalance may highly degrade network performance and cause unfairness. The efficient assignment of network resources can enable enhancements in the use of the wireless medium and, as a consequence, in the quality of experience of the user. To solve this problem, we propose a joint channel selection and load balancing algorithm based on the SDN paradigm. This algorithm coordinates the operations of several APs with the aim of estimating the user distribution that delivers the highest performance and minimizes resource usage. As a matter of fact, experimental results have shown how the algorithm is able to outperform the RSSI-based user association schemes by up to 25% and reduce resource utilization by up to 30%. Furthermore, high performance is also ensured for mobile users thanks to the *LVAP* abstraction which, by means of a per-client AP, enables seamless handovers even across different channels and/or bands.

Multicast services in WLANs lack feedback mechanisms and are transmitted at the lowest MCS, thus causing performance issues and high resource consumption. To address these problems, the IEEE 802.11aa amendment introduces a set of multicast policies. However, each policy is only suitable for a specific scenario and the amendment does not define any procedure to combine them or to dynamically select the appropriate policy depending on the scenario considered. In this context, we take advantage of the capabilities of SDN to design a programming abstraction named *Transmission Policy* to allow the SDN controller to specify the MCS and the multicast retransmission policy to be used for a particular multicast group.

The *Transmission Policy* abstraction has laid the basis for designing an algorithm capable of dynamically recomputing the most suitable MCS for each group. As a result, channel utilization is reduced by up to 80% with respect to the standard schemes without diminishing the performance, which also allows other transmissions to take place simultaneously and improves the user experience. The capabilities of this solution are extended to account for roaming users and to orchestrate the operations of various multicast sessions that are simultaneously delivered. This approach enables support for seamless mobility in multicast environments, hence improving the performance for mobile users and providing a similar quality to the one experienced by static users. Furthermore, as opposed to the standard solutions that show serious scalability issues, the performance of the algorithm is close to 100% when delivering several multicast transmissions at the same time.

In summary, we can conclude that the QoS capabilities in 802.11 have a critical role in the quality perceived by the user when it comes to voice and video services. In this Doctoral Thesis we have addressed the QoS provisioning problem from different perspectives. On the one hand, the performance of delay sensitive traffic can be improved by exploiting artificial intelligence techniques with the aim of identifying traffic patterns and reacting to changes in an autonomous fashion. On the other hand, the flexibility provided by the Software Defined Networking paradigm opens up new research challenges in future Wi-Fi networks. Decoupling network control from the forwarding operations at the APs makes it possible to overcome the problems derived from the ossified traditional architecture and introduce faster standard compliant innovations that are directly applicable to commercial devices.

3.2. Future Directions

The work presented in this Doctoral Thesis encourages new research directions for future work. This Thesis places particular emphasis on designing solutions compatible with Wi-Fi compliant devices and their direct applicability in the industry. However, the work proposed to improve the capabilities of EDCA is implemented via simulation. Accordingly, in the short term, we intend to support these features for real network cards. The algorithms presented in this work base their decisions on the channel occupancy of each type of traffic. In this regard, we have seen that through SDN abstractions we are able to obtain information about the overall channel utilization without distinguishing between different applications. Therefore, extending the scope of these abstractions would allow the algorithms to have the data they require. Moreover, to further simplify the design, the algorithms could be placed at the application layer instead of running on the APs, hence retrieving the information from the SDN controller. As a result, the complexity of the devices would be reduced and the SDN approach would enable more efficient decisions that can be coordinated across the APs.

Furthermore, apart from the proposal mentioned above in which IEEE 802.11g [85] was considered, the remaining works presented in this dissertation have taken into account the IEEE 802.11a physical layer [86]. However, they were adapted to contemplate the capabilities available in IEEE 802.11n [87]. On this basis, another possible extension in the short term is to migrate all the proposed solutions to IEEE 802.11ac [88] with the aim of supporting a high number of devices, meeting the increasing bandwidth demand and guaranteeing the user experience. Furthermore, backward compatibility with the previous amendments must be ensured. This same approach can be followed to cover the forthcoming IEEE 802.11ax radio access [89].

Enabling more advanced QoS features has become mandatory in future Wi-Fi networks. This fact is motivated by the emergence of 5G networks, characterized by supporting diverse and dynamic services. Conversely, WLANs are typically designed in such a way that different physical network portions correspond to different types of users or services. However, these portions are usually rigid, do not consider the requirements of the applications and cannot be dynamically reconfigured since it would require deep changes in both the network architecture and the 802.11 stack. In this context, the SDN paradigm makes it possible to break these restrictions and accommodate multiple logical networks on the same physical architecture. This approach, named network slicing, enables a service-oriented vision in which each slice is characterized by different settings and dynamic control policies.

Based on the slicing concept, in our last work [80] we have introduced a programmable end-to-end slicing framework for WLANs. The purpose behind this proposal is to enable the flexible definition and management of several wireless segments while ensuring functional isolation and efficient resource utilization. Experimental tests have shown that this framework is able to offer high performance and slice customization. In this work, a hypervisor must schedule the operations of the slices and divide the resources of each slice across the users. In the current implementation, an Airtime Deficit Round Robin (ADRR) policy is used for both tasks. In this respect, other more sophisticated approaches could be introduced for this job to prioritize the user operations in a more effective way. Moreover, traffic aggregation functionality could be added as a parameter of each slice with the aim of allowing it to apply any type of aggregation mechanism defined in 802.11 for the traffic of a certain slice.

In the long term, we intend to reinforce the QoS capabilities by means of flexible network control and management. To this end, high scalability and resilience are desired in radio access networks. Despite the advantages provided by the centralized control plane, most of the operations are undertaken on this layer. Consequently, it represents the network core, and becomes a critical point of failure as well as a potential bottleneck [90]. In this regard, more than one controller may be needed for meeting high QoS constraints and ensuring high performance. This problem could be addressed by distributing multiple SDN controllers across the network. In this way, each of them must manage the set of APs and users in each network segment and report to the remaining controllers on the collected information with the aim of having a complete view of the network. In this sense, messages from other controllers could be used as a heartbeat. Accordingly, if no information is received from one controller for several rounds, its functionality should be adopted by another available one. Hence, a procedure for selecting the appropriated controller in each case must also be investigated.

Finally, the range could be extended by introducing Network Function Virtualization (NFV). This paradigm allows for virtualizing most of the network functionality and its agile deployment in the cloud. As a result, new network features can be dynamically added and the network capabilities can scale to meet an increasing resource demand. Furthermore, it makes it possible to control user devices in different networks. Based on this, the scope could be expanded to introduce cooperation between various radio access networks, i.e. between Wi-Fi and 5G interfaces. This advantage could be used for a two-fold purpose. On the one hand, mobility could be improved by transparently switching between both radio access networks to ensure high signal quality and performance. On the other hand, traffic could be offloaded to the other network segment for load balancing purposes. This offloading approach could be limited to the same network slice, if available, to ensure the same fairness level in the resource assignment regardless of the radio access mode. As a consequence, the latency for delay sensitive services would be reduced, hence contributing to delivering a better QoE.

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