A Prioritized Adaptive Scheme for Multimedia Services over IEEE 802.11 WLANs

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***Abstract*—IEEE 802.11e protocol enables QoS differentiation between different traffic types, but requires MAC layer support and assigns traffic with static priority. This paper proposes an intelligent Prioritized Adaptive Scheme (iPAS) to provide QoS differentiation for heterogeneous multimedia delivery over wireless networks. iPAS assigns dynamic priorities to various streams and determines their bandwidth share by employing a probabilistic approach-which makes use of stereotypes. Unlike existing QoS differentiation solutions, the priority level of individual streams in iPAS is variable and considers service types and network delivery QoS parameters (i.e. delay, jitter, and packet loss rate). A bandwidth estimation technique is adopted to provide network conditions and the IEEE 802.21 framework is used to enable control information exchange between network components without modifying existing MAC protocol. Simulations and real life tests demonstrate how better results are obtained when employing iPAS than when either IEEE 802.11 DCF or 802.11e EDCA mechanisms are used. The iPAS key performance benefits are as follows: 1) better fairness in bandwidth allocation; 2) higher throughput than 802.11 DCF and 802.11e EDCA with up to 38% and 20%, respectively; 3) enables definite throughput and delay differentiation between streams.**

***Index Terms*—Multimedia delivery, QoS differentiation, IEEE 802.11, IEEE 802.21.**

I. Introduction

LATELY, IEEE 802.11 WLANs have been used widely to deliver multimedia content including video, voice, text and other data [1]. At the same time, the number and type of devices receiving multimedia content over the 802.11 links has increased significantly. These wireless networking-enabled devices, mostly mobile and handled, are highly heterogeneous in terms of processing capabilities, screen resolution, battery power, memory, etc.

Delivering multimedia content to heterogeneous devices over a variable networking environment while maintaining high quality levels involves many technical challenges [2], [3]. Fig. 1 presents a common scenario inspired from the home WLAN. A single IEEE 802.11 wireless router provides broadband services to multiple devices. In order to support such high quality, different multimedia services have various delivery-related QoS requirements. For instance, real-time video traffic needs large bandwidth and is less tolerable to delay and jitter, in comparison with any best-effort service. Furthermore, while delivering the same multimedia content, devices with high resolution and large battery power levels should benefit from larger bandwidth than those allocated to devices with lower resolution and reduced battery power levels. Additionally, wireless multimedia delivery solutions should consider the dynamic nature of the wireless channels, which impact the quality of multimedia applications.

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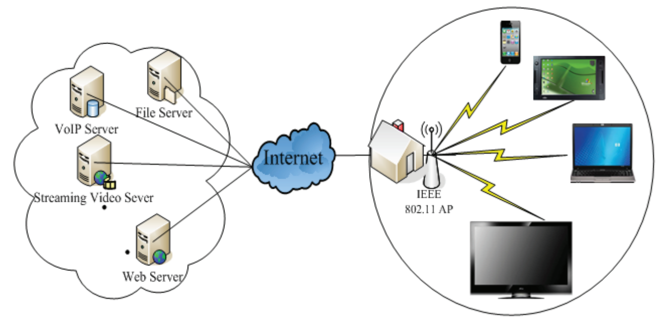


Fig. 1. Home wireless environments with heterogeneous devices.

The original IEEE 802.11 protocol adopts the CSMA/CA mechanism to manage the wireless channel access [4]. However, the 802.11 standard is only designed for best effort service and incorporates limited QoS differentiation support with regard to multimedia applications and mobile devices. The IEEE 802.11e has been developed to overcome the QoS problem of traditional 802.11 networks, introducing support for differentiation between four different classes of traffic: voice, video, best effort and background [5]. However, when the 802.11e channel is occupied by high priority traffic, the low priority traffic might suffer from starvation due to the low channel access opportunity. Nevertheless, 802.11e cannot provide QoS differentiation between various devices, nor between services belonging to the same class (i.e. video streams). Recently, several solutions have been developed to optimize the original 802.11e, including [6], [7], and [8]. Other QoS-oriented solutions like TCP Friendly Rate Control (TFRC) [9], Quality Oriented Adaptive Scheme (QOAS) [10], [11], Region of Interest Adaptive Scheme (ROIAS) [12], Partial Reliable-Stream Control Transmission Protocol (PR-SCTP) [13], Quality-aware Adaptive Concurrent Multipath Data Transfer (CMT-QA) [14], etc. are proposed at different layers of the protocol stack. However, none of the above solutions provide support for both high QoS provisioning and QoS differentiation for delivering multimedia services to heterogeneous devices. Additionally, these solutions lack wireless network conditions awareness (i.e. interference, collisions, link rate adaptation).

This paper proposes **the intelligent Prioritized Adaptive Scheme (iPAS)**, which provides QoS differentiation between multiple streams during wireless multimedia delivery. iPAS assigns dynamic priorities to various streams and determines their bandwidth share by employing a stereotypes-based approach. The priority level of individual stream is variable and depends on stream-related characteristics (i.e. device resolution, battery left, and application type) and network delivery-related QoS parameters (i.e. delay, jitter, and loss). iPAS, first introduced in [15], is incorporated into the IEEE 802.21 framework [16], [17], [18], [19], supporting both network information gathering and dissemination.

The structure of the paper is as follows. Section II introduces the related works on QoS provisioning and QoS differentiation solutions, mathematical theory for resource management, and the IEEE 802.21 framework. Section III presents the details of the iPAS architecture and Section IV describes the principle of the stereotypes-based resource management. Simulation and real life test-bed setup and analysis results are presented in Section V, Section VI, and Section VII, respectively. Finally, Section VIII concludes the paper.

II. Related Work

This section introduces the related works with regard to QoS provisioning and differentiation, resource management and IEEE 802.21 framework.

A. QoS Provisioning and Differentiation-based Solutions

Current solutions for QoS provisioning or QoS differentiation mainly focus on packet adjustment-based techniques. Extensive approaches have been designed at different OSI layers:

(a) Application Layer. Adaptive schemes at the application layer adjust the transmission rate to best suit the available network conditions. Early research works like *Rate Adaptation Protocol (RAP)* [20] requires the sender to adapt the transmission rate based on an *Additive Increase Multiplicative Decrease (AIMD)* rate adaptation scheme. If congestion occurs, the transmission rate is reduced and this increases the probability of packets arriving at the destination. *TCP-friendly Rate Control (TFRC)* [9] determines the transmission rate by combining packet loss and round-trip time. *Quality Oriented Adaptation Scheme (QOAS)* [10], [21] adapts the transmission rate at sender based on receiver's estimation of the perceived quality. [22] proposes a QoS-aware service management framework at application layer. The framework enhances services of resource-demanding applications with QoS-awareness, in resources-limited scenarios.

(b) Transport Layer. The Internet Engineering Task Force (IETF) developed a novel transport layer protocol referred to as *Partial Reliable-Stream Control Transmission Protocol (PR-SCTP)* [13]. It is an unreliable service mode extension of SCTP which differentiates retransmissions based on a reliability level that could be set dynamically. When a certain pre-defined threshold is reached, the sender abandons packet retransmission and sends the next incoming packet from the application layer. The reliability level could be set based on different data types or end stream requirements.

(c) Network Layer. IETF developed two network layer-based frameworks providing QoS differentiation: Integrated Services (IntServ) [23] and Differentiated Services (DiffServ) [24]. The IntServ framework provides per-flow QoS provisioning. Sufficient resources must be reserved at each network router on the end-to-end path. Applications are able to choose among multiple QoS levels for their data packets. Unlike the IntServ architecture where each stream notifies its QoS requirements to the network, DiffServ requires each router setup with identical traffic categories to provide service differentiation. Each packet transmitted in DiffServ is marked in its header and differentiated treatment is performed per traffic class.

(d) Link Layer. Many QoS-oriented solutions are designed at the link layer. Earlier solutions focused on queue scheduling modifications including [25], [26], [27], [28]. Recently, many research works improve the Medium Access Control (MAC) protocols [29], [30], [31], [32]. The principal idea of these schemes is to relate differentiated services with dynamic MAC parameters such as Contention Window (CW), Inter Frame Space (IFS), etc. IEEE 802.11e is developed in order to overcome the lack of QoS control of original 802.11 protocols. 802.11e classifies multimedia traffic using the concept of Traffic Categories (TC). Each category maintains its own set of QoS-related parameters, i.e., CW and Arbitration Inter Frame Space (AIFS) values. Higher priority categories obtain better services by setting lower CW sizes and AIFS values. 802.11e has performance issues such as low priority traffic class starvation in heavy traffic load and [6], [7], [8] propose solutions to solve these problems. However, current link layer–based solutions need to modify existing MAC layer scheduling algorithms in order to provide differentiated QoS. In comparison, iPAS is a middleware solution which reduces the implementation cost.

(e) Cross-layer. Cross layer techniques ensure QoS provisioning by combining the information obtained from different OSI layers. Xiao, et al. [33] propose an application-MAC cross layer adaptive scheme for MPEG-4 video, *Prioritized MPEG-4 Frame Transmission (PMFT)*. In PMFT, I-frames were assigned higher priority than B and P frames by giving more retransmission attempts, lower CW size and lower *AIFS*. In the case when I frames get lost, the following P and B frames will be discarded. Similar with [33], the scheme proposed in [34] assigns video packets of different importance different MAC retry limits, thus allowing different channel access opportunity. The authors of [35] propose a novel cross-layer framework in wireless multimedia sensor networks in order to maximize the capacity of the network without affecting the multimedia quality. This is achieved by employing Wyner-Ziv lossy distributed source coding with variable group of pictures size and exploiting multipath routing for multimedia delivery. Other researchers describe in [36] a cross-layer solution to adapt multimedia applications according to both client QoS requirements and network resource constraints. The approach is employed at three layers: 1) in the application layer, the requirement levels are changed based on measured delay; 2) in the middleware layer, a priority adaptor is employed to adjust the service classes for applications using feedback control theory; 3) in the network layer, the service differentiation scheduler assigns different network resources (i.e. bandwidth) to different service classes.

B. Mathematical Theories for Resource Management

The basic idea of Resource Management (RM) schemes is to deal with the uncertainty events in order to allocate resources distributed across a heterogeneous environment. Recently, several mathematical-based solutions were considered to optimize the resource allocation including: stereotypes [37], fuzzy logic [38], game theory, overlay network, etc.

Stereotypes for managing groups were first introduced by Rich in the Grundy system [37] and they are still widely used by many QoS-oriented adaptive solutions [39], [40], [41], [42], [43]. Stereotypes make a powerful probabilistic analyzing tool for dealing with a wide range of uncertain events, and can be useful especially in the case of variable wireless environments.

Fuzzy logic is a mathematical theory that attempts to imitate the human decision logic. The fuzzy set theory adopted by fuzzy logic can be used to represent subjective events. The fuzzy set theory uses the concept of grade of membership, i.e., how much a factor is in the set. In contrast with traditional logic theory like two-valued logic: true or false, Fuzzy logic forms a multiple-valued logic. The truth value of fuzzy logic variables ranges between 0 and 1, indicating different levels of true. Although fuzzy logic-based resource management has been applied in many areas [44], [45], it was not fully embraced mostly due to the concerns regarding the validity of the methods used or the preference towards probability for rigorous mathematical description of uncertainty.

Game theory applies to a regulated circumstance where a player's success is based on the choices of others. Wang, et. al. [46], [47] propose resource allocation solutions using the game theoretic approach. They build a multidimensional game model

using quantified manufacturing data. By solving the game model, the Nash equilibrium solution is regarded as an optimal resource allocation strategy. The desired equilibrium makes each product achieve optimal production status and simultaneously improve the resource utilization efficiency. A team of researchers propose in [48] the game theoretic-based Multipath Multimedia Dynamic Source Routing (g-MMDSR), a multipath routing protocol to improve video streaming services, and enable dynamic selection of the routing paths. Considering that each video source node intends to transmit a set of frames through several paths, each node plays a routing game to distribute the video content while aiming to achieve its own best performance. By solving the game model, there is an optimal routing through which the competing nodes share the common resources in a more satisfactory and efficient way.

Overlay networks have been widely deployed to provide end-to-end QoS support without modifying the existing network architecture. The overlay topology adopts a routing-based service differentiation policy to ensure the application's QoS requirements. Egashira, et al. [49] develop a resource allocation approach using a market-based mechanism in the overlay networks. The idea is based on the fact that there is

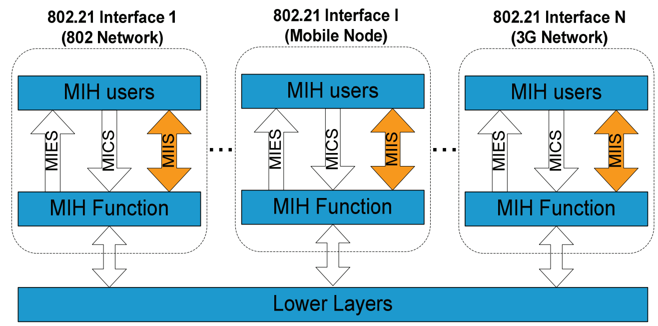


Fig. 2. Communications between different 802.21interfaces.

certain price for the network resources and each application provider aims to buy resources needed to satisfy its QoS requirements with minimum cost. The proposed architecture allows an application provider to buy resources from other application providers when the network resources are unavailable or expensive.

C. IEEE 802.21 Framework

The IEEE 802.21 framework [19] has been released to improve user experience of mobile stations by supporting handover between heterogeneous technologies including Wi-Fi, WiMAX and 3G. IEEE 802.21 provides a mechanism that allows interaction between lower layers and network layer without dealing with any specific technology. [16] focuses on IEEE 802.21 Media Independent Information Service (MIIS) architecture and studies how it can enhance the mobile node's (MN) experience in different handover scenarios. Both the proposed solution (iPAS) and [16] deploy the MIIS module in NS-2 simulator and utilize 802.21 MIIS for improving user experience without causing extra cost. The difference is that [16] focuses on handover issues in telecom networks while iPAS is designed for resource management in WLAN networks.The 802.21 MIIS is used to store information and maintain connections when switching between different networks. A general approach for deploying the 802.21 framework in a simulation environment is presented in [16].

Fig. 2 shows a logical diagram of the architecture of the different 802.21-enabled entities. Three types of 802.21 interfaces are presented, i.e., 802 network, mobile node, and 3G network. It can be observed from the figure that all the 802.21-compliant nodes have the same structure. IEEE 802.21 defines the MIHF that facilitates both mobile station and network initiated handovers. Each node maintains a set of MIH users, typically the mobile management protocols, that use the MIHF functionality to control and gain handover-related information. The MIHF encompasses three types of communication services which are the core of the specification, as shown in Fig. 2:

- *MIH Event Services (MIES).* The events related to handovers originate at the medium access control (MAC) layer, physical layer or MIHF layer at the mobile nodes or the network nodes. These events are detected from the lower layers and reported to the MIH users.

- *MIH Command Services (MICS).* MICS provides a set of commands to allow the MIH users to control the information from the lower layers.
- *MIH Information Services (MIIS).* MIIS presents a framework whereby the MIHF is able to acquire network and terminal information, such as network type, service provider identifier, QoS information, data rate, channel characteristics, vendor specifications, etc. MIIS specifies a standard format for this information, such as Extensible Markup Language (XML) or Type Length Value (TLV). They are transmitted through MIIS using query/response or broadcast/multicast mechanisms.

These services are independent of each other and provide a unified interface for the upper layers. The MIHF logically resides between the link layer and the network layer.

In summary, the proposed solution, iPAS, is developed to provide differentiated QoS using the cross-layer technique introduced in Section A. Unlike existing research works, iPAS takes into account both stream characteristics and network conditions by adopting the stereotypes-based structure that is introduced in Section B. The main novelty of iPAS, in comparison with the state-of-the-art literature, is the flexible differentiated service and increased network utilization.

III. iPAS System Architecture

Fig. 3 presents the iPAS system architecture which consists of two main blocks: *iPAS server* and *iPAS client*. The iPAS server is responsible with managing bandwidth resources using a stereotype-based resource allocation mechanism and a bandwidth estimation scheme (MBE). The iPAS client collects information about stream preferences, which is sent as feedback to the iPAS server.

The multimedia traffic is delivered using Real-time Transport Protocol (RTP) [50] and the feedback is transmitted using Real-time Control Protocol (RTCP) [50]. *The Server and Client Communication Agent* situated at the two communicating sides of the system, respectively, establish and manage the communication link. IEEE 802.21 MIH Function and MIH User modules are utilized to gather feedback information from lower and upper layers of the multimedia gateway system, respectively. Fig. 3 also indicates the feedback exchanged between different system components. MBE feedback consists of information regarding packet size, number of clients, MAC layer loss and round trip time. iPAS feedback consists of data regarding the service type, screen characteristics, power consumption level, data delivery delay, delay jitter and packet loss. The details of each major component of the iPAS system are presented next.

A. Model-based Bandwidth Estimation (MBE)

A novel *Model-based Bandwidth Estimation (MBE)* algorithm for IEEE 802.11 networks was proposed in [51], [52]. MBE considers traffic carried by the two basic, yet most widely used transport layer protocols, TCP and UDP.

In the first step, *MBE* relies on a novel TCP model for wireless data communications, which extends an existing TCP throughput model [52] by considering the IEEE 802.11 WLAN characteristics (transmission error, contention, and retry attempts) [54]. The proposed TCP over WLAN throughput model is given by equation (1), which uses the Padhye's TCP model [53]. *MBE* updates two parameters, round-trip time and loss to take into account both TCP congestion control and wireless channel characteristics. The achievable bandwidth B for each TCP connection is described in (1), where b is the number of packets acknowledged by a received ACK, P_{retr}^{TCP} denotes the probability of packet retransmission, *MRTT* is the transport layer round-trip time between sender and receiver, and MSS means the maximum segment size. T_0 is the timeout value used by the congestion control. (See equation (1) at the top of the next page.)

In step two, *MBE* approximates the UDP throughput by analyzing the UDP packet transmission probability and the IEEE 802.11 channel delay. The maximum achievable bandwidth for UDP traffic over 802.11 WLANs is given in (2), where *Payload* is the total information transmitted during one time period from T_0 to T_1, and *Delay_UDP* denotes the average delay for successfully transmitted individual UDP packets. The core parameter is *Delay_UDP*, which was derived to take into account the IEEE 802.11 MAC layer retransmission mechanism. *Delay_UDP* includes delay due to the application process, propagation transmission, MAC layer delay, and time out delay caused by readmission. Equation (2) presents the UDP bandwidth estimation formula.

$$B^{UDP} = \frac{\int_{T_0}^{T_1} \frac{Payload}{Delay\_UDP} dt}{T_0 - T_1} \tag{2}$$

Additionally, *MBE* derives a formula predicting the achievable bandwidth when TCP and UDP co-exist in 802.11 networks, as shown in (3). The parameter w is the bandwidth weight factor since TCP and UDP have different bandwidth requirements. The computation of w is given in [51].

$$\sum B^{TCP+UDP} = w \times \sum B^{UDP} + (1-w) \times \sum B^{TCP} \tag{3}$$

Both extensive simulations and real tests performed demonstrate that *MBE* performs very well in conditions with variable packet size, dynamic wireless link rates and different channel noise [51].

B. iPAS

There are three steps in the functionality of iPAS: 1) Assign priority to each stream; 2) Allocate certain bandwidth share (in percentage) for each stream with a probability value through the stereotypes-based algorithm; 3) Allocate specific bandwidth amounts to the streams by combining the stream's bandwidth share and the estimated available bandwidth according to the **Model based Bandwidth Estimation (MBE)**.

The communication between the iPAS server and the iPAS client applications makes use of a control communication link which is created when the client sends a request to the server. This link is used for the exchange of control messages including feedback information. Subsequently, an additional data communication link is established between the server and client allowing for multimedia data transmissions to be performed.

Server Communication Agent (SCA) and **Client Communication Agent (CCA)** situated at both sides of the

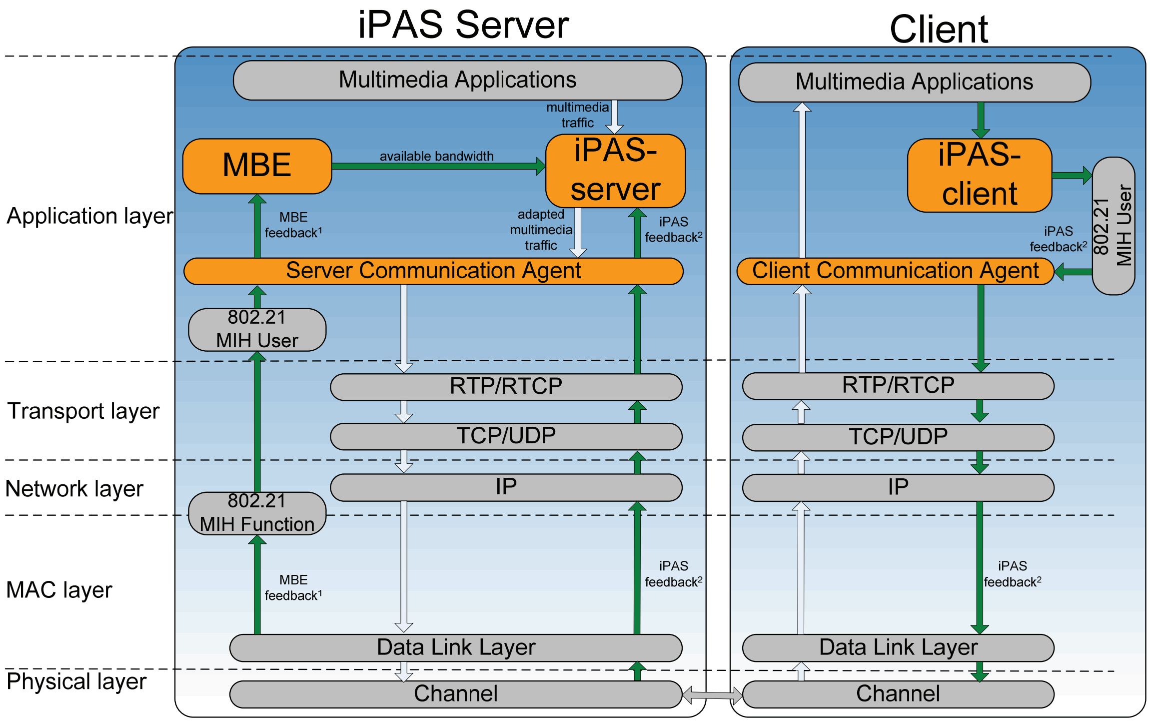


Fig. 3. The block structure of the iPAS-based multimedia delivery system.

$$B^{TCP} = \frac{MSS}{MRIT \times \sqrt{\frac{2bP_{retr}^{TCP}}{3}} + T_0 \times \min\left(1, 3\sqrt{\frac{3bP_{retr}^{TCP}}{8}}\right) \times P_{retr}^{TCP} \times \left(1 + 32P_{retr}^{TCP^2}\right)} \quad (1)$$

communication link are responsible with establishing and maintaining the double control-data links. CCA manages the receiver buffer and forwards feedback messages from the **Feedback Controller (FC)** component to the 802.21 interface. SCA manages the sender buffer and forwards the feedback messages from the 802.21 interface to the **MBE** and **Stereotypes-based Bandwidth Allocation (SBA)** components. iPAS makes uses of the IEEE 802.21 framework (i.e. MIIS function) to transmit the control signals. In 802.21 MIH terminologies, an MIH User is any application that resides at layer 3 or above in TCP/IP protocol stack and has access to the MIH services. In the iPAS system, the MIH User instances are configured at application layers in both iPAS server and client. The role of the MIH User instances are to retrieve MAC layer information (i.e. MAC layer delay caused by retransmission and contention) and upper layer information (i.e. device resolution, service type) using MIIS function of MIHF module. Since 802.21 is a standard protocol for communication between MAC and upper layers, there is no need to modify existing protocols. The multimedia data traffic uses RTP/ UDP or TCP protocols.

SBA, located at the iPAS server, is the center piece of the iPAS system. SBA is responsible for determining each stream's priority level and suggesting a proportional bandwidth share. Two types of control information, estimated bandwidth from MBE and feedback information from SCA, are utilized by SBA for analysis. MBE estimates the available bandwidth using the equations (1)–(3) based on the feedback information (loss and transmitted data size) sent from SCA. The details of the stereotypes-based process for bandwidth allocation are presented in the next section.

FC, located at the iPAS client, gathers feedback-related parameters from client applications and CCA and sends the formatted feedback messages to SCA. Two types of feedback related parameters are processed by FC: 1) *Stream characteristics related parameters* such as the application type, device resolution, and device power left. They are initialized by the client application process when sending the first request and updated whenever there is a change. 2) *Delivery QoS-related parameters* such as delay, jitter, and packet loss rate, which are extracted from the CCA's receiver buffer. The computation of the instant delay takes into consideration packet timestamps as suggested in [55], the calculation of the instant jitter is based on the computed delay as shown in [56], and the measurement of the instant packet loss rate is done by analyzing the packets' sequence numbers as presented in [57]. These measured values are monitored by the FC and sent to the iPAS server as feedback messages. The instant values of the QoS parameters (delay, jitter, packet loss rate) are computed each time the multimedia packet arrives at the client. To alleviate the fluctuation of these QoS parameters values, average values $AVGQoS_{delay}$, $AVGQoS_{jitter}$, and $AVGQoS_{loss}$ are considered for each one of the QoS parameters: delay, jitter, packet loss rate, respectively. The incremental computation of the estimated average values is suggested in [58] and is given in (4) as an example for the delay parameter. α is an update factor

$(\alpha = 0.9)$. *AVGQoS*$_{delay}$, *AVGQoS*$_{jitter}$, and *AVGQoS*$_{loss}$ are initialized with the first QoS parameter value available and updated at the end of each monitoring interval.

$$AVGQoS_{param} = AVGQoS'_{param} \times \alpha + QoS_{param} \times (1-\alpha) \quad (4)$$

IV. Stereotypes-Based Bandwidth Allocation

An optimal bandwidth resource allocation scheme must be able to deal with uncertain and imprecise information related to the wireless channel and streams. At the same time, the bandwidth resource management mechanism must be flexible enough in order to adapt to network fluctuations and device characteristics changes. **SBA** satisfies the above requirements by using a stereotypes-based resource allocation process.

Stereotypes are defined as stream classes (groups) described by a set of features, which include attributes. Each stream will belong to every stereotype group with a certain probability depending on stream features. These features include delay, jitter, packet loss rate, service, device resolution, and battery power left. iPAS utilizes the stereotypes to build stream profiles and then suggest a proper bandwidth share for each stream. The bandwidth share of one stream will be suggested by combining the features based on the probabilities.

A. Principle of Stereotype-based Resource Allocation

SBA uses five stereotypes classes *(Th)*: *High Priority (HP)*, *Medium to High Priority (MHP)*, *Medium Priority (MP)*, *Medium to Low Priority (MLP)*, *Low Priority (LP)*. Each *Stream*$_i$ belongs to one of the five stereotypes with a certain probability. Each stereotype class *Th* consists of two components: a group of features $F = (F_1, F_2, \ldots, F_i, \ldots, F_m)$ describing the stereotype and a group of suggestions $S = (S_1, S_2, \ldots, S_j, \ldots, S_n)$ that should be performed to determine stream's bandwidth. Each feature F_i has associated a list of linguistic terms $LF_i = (LF_{i1}, LF_{i2}, \ldots, LF_{ip}, \ldots, LF_{iq})$. Each linguistic term LF_{ip} has a numeric value PF_{ip} between 0 and 1, representing the probability that the feature F_i equals the linguistic term LF_{ip} for this stereotype *Th*. The probability PF_{ip} indicates the degree of match between stream's characteristics and the stereotype. A similar structure is defined for each suggestion S_j, which has also associated the linguistic terms $LS_j = (LS_{j1}, LS_{j2}, \ldots, LS_{jp}, \ldots, LF_{iq})$ probabilistic values PS_{jp}.

The Poisson distribution is used to determine the probability associated with the linguistic terms. The Poisson distribution represents the probability of a given number of events occurring in a fixed interval of time. The occurrence of each event is independent of time of the last event. Equation (5) shows the Poisson distribution function where u is the shape parameter and indicates the mean and the variance of the distribution during a time interval. The integer value $x (x = 0, 1, 2, \ldots, n)$ represents a particular event.

$$pois\,(x, u) = \frac{u^x \times e^{-U}}{x!} \quad (5)$$

By analyzing the shape of the Poisson function, a near normal distribution is obtained for $u = 7$ across the $[0, 15]$ interval. The selected value of u has also been used and validated in [36] for network parameter modeling. The maximum value of the normal distribution close to 0.15 $(x = 7, u = 7)$ and the minimum value close to $(0\ x = 0$ or $x = 15,\ u = 7)$. Consequently, the interval $[0, 15]$ is considered for the computation of the Poisson function for all the stereotypes. It is noticed that each stereotype associates one Poisson distribution with a mean value u_k which is obtained by dividing the interval $[0, 15]$ in five equal segments and considering their middle value.

Considering feature F_i has a list of linguistic terms, where the length is q, the probabilistic values for each term PF_{ij} are computed as in (6), (7), and (8). The value i implies the index of feature and j is the index of linguistic term in feature i,

$$PF_{ij} = Average\,(pois\,(x_j, u)) \quad (6)$$

$$x_j \in [step \times (j-1)\,, step \times j] \quad (7)$$

$$step = \left\lceil \frac{15}{q} \right\rceil \quad (8)$$

1) Stream Classification

The purpose of the stream classification is to determine the stereotype classes the stream belongs to and with what probability. iPAS describes a stream with the format shown in equation (9), where F_i is the name of the ith feature and LF_iK_i represents the linguistic term of the i_{th} feature.

$$U = ((F_1, LF_1K_1)\,, (F_2, LF_2K_2)\,, \ldots, (F_m, LF_mK_m)) \quad (9)$$

A degree of match between a stream and each stereotype is computed in (10) based on the probability theory.

$$\begin{aligned} M\,(Th) &= p\,(Th|F_1 = LF_1K_1, \ldots, F_m = LF_mK_m) \\ &= p\,(Th|F_1 = LF_1K_1) \times \ldots \times p\,(Th|F_m = LF_mK_m) \end{aligned} \quad (10)$$

The computation of each factor is performed using the Bayes rule, as shown in equation (11). In the equation, $P(Th)$ represents the probability that a certain stream belongs to one stereotype class and $P(F_i = LF_iK_i)$ indicates the probability that feature i is equal to LF_1K_1. It is considered that there are equal priority-probabilities $(P(Th) = 1/Number\ of\ Stereotypes)$ for all stereotype classes. This assumption is made as a delivered flow could belong to any stereotype class (no supplementary information on the traffic is available).

$$\begin{aligned} P(Th|F_i = LF_iK_i) &= \frac{p\,(F_i = LF_iK_i|Th) \times p\,(Th)}{p(F_i = LF_iK_i)} \\ &= \frac{PF_iK_i \times p\,(Th)}{P(F_i = LF_iK_i)} \end{aligned} \quad (11)$$

2) Suggestion Determination

The suggestion determination procedure is performed to determine the bandwidth share for each stream.

First, for each stereotype class, the strength of each suggestion has to be re-computed by considering the probability with which the stream belongs to this class, as given in equations (12). S_i is the name of the i_{th} suggestion and LS_iK_i represents the linguistic term of the i_{th} suggestion.

$$\begin{aligned} p\,(S_i = LS_iK_i|Th) &= p\,(S_i = LS_iK_i|Th) \times M\,(Th) \\ pS_ik_i\,(Th) &= PS_ik_i\,(Th) \times M\,(Th) \end{aligned} \quad (12)$$

Second, the combination of all stereotype suggestions resulted from equation (12) can be calculated using the probabilistic theory. Equation (13) shows an example of how the strength of suggestion S_i can be calculated. S_i is equal to the linguistic term LS_iK_i in case that the stream belongs to two stereotypes classes $Th1$ and $Th2$, respectively. (See equation (13) at the top of the next page.

B. Stereotypes-based Resource Allocation for iPAS

The stereotype classes defined by iPAS include six features: *delay*, *jitter*, *loss*, *power left*, *device resolution*, and *application type*. Delay, jitter, and loss are QoS parameters of the streams. Power left and device resolution indicate the client device characteristics. The application type includes five widely used application types, a model which extends the 802.11e four class model by considering the different quality video content: VoIP, Standard-Definition Video (SD-Video), High-Definition Video (HD-Video), Best-effort Service, and Background Traffic. Each feature is divided into five levels using threshold values, as shown in Table I. These threshold values are suggested based on ITU-T Rec. G.1010 [59] and ITU-T Rec.Y.1541 [60]. Different applications have specific requirements on the QoS features. Take VoIP for example, one way delay of less than 150ms indicates excellent quality, while delay higher than 400ms causes bad perceived quality. iPAS assigns higher priority to traffic which is sensitive to delay and jitter, i.e. voice and video.

All five stereotypes have the same structure: six features and each feature consist of five linguistic term-probability pairs. For the purpose of this demonstration, the groups of features and suggestions for the medium priority stereotype *(MP)* are shown in Tables II and III. The probabilities are calculated based on equations (1)–(4).

C. Exemplification

With the stereotype classes proposed, we give an illustration of the bandwidth allocation process involving two streams: U1 and U2.The procedure includes the initialization phase (steps 1 to 3) and the update phase (step 4):

1. Collect stream related parameters regarding each feature F_i of the stereotype Th. These parameters include: delay, jitter, packet loss rate, power left, device resolution, and application type. For the purpose of demonstration, the two streams are first configured with the six features using static values. The features of the two streams are shown in equations (14) and (15).

$$U_1 = (delay, 150ms), (jitter, 50ms), (loss, 1\%), (power, 85\%) \\ (resolution, 768 \times 480), (application, VoIP) \tag{14}$$

$$U_2 = (delay, 1500ms), (jitter, 70ms), (loss, 2\%), (power, 85\%) \\ (resolution, 768 \times 480), (application, HTTP) \tag{15}$$

2. Determine the degree of match between the stream and each stereotype. This can be done using equations (9)–(11). After the normalization of the calculated values, we have probabilistic results indicating the match degree and bandwidth share suggestion, as shown in Table IV. The two streams have different probabilities to belong to the stereotype classes, i.e., U_1 belongs to the MHP and MP stereotype classes with probability of 57.05% and 42.95%, respectively and 0% to the other stereotype classes.

3. The maximum bandwidth share for U_1 and U_1 are denoted as B_1_MAX and B_2_MAX which can be calculated based on Table IV, as shown in equations (16) and (17):

$$\begin{aligned} B_1\_MAX &= 20\% \times 17.44\% + 40\% \times 41.97\% + 60\% \\ &\quad \times 29.23\% + 80\% \times 9.61\% + 100\% \times 1.75\% \\ &= 47.25\% \end{aligned} \tag{16}$$

$$\begin{aligned} B_2\_MAX &= 20\% \times 0.91\% + 40\% \times 6.39\% + 60\% \\ &\quad \times 23.41\% + 80\% \times 46\% + 100\% \times 23.29\% \\ &= 76.87\% \end{aligned} \tag{17}$$

By normalizing B_1_MAX and B_2_MAX, the bandwidth for U_1 and U_2 are 38.07% and 61.93%. Consequently, the actual amount of bandwidth is obtained based on the estimation by MBE.

4. Actual values for the QoS parameters during data transmission for each stream will be sent back regularly to SBA module in iPAS. Step 1 to step 3 are repeated to update the priority level and the bandwidth share of each stream is re-evaluated. The probabilistic values PF_ik_i associated with the linguistic values LF_ik_i are recalculated.

The above four steps present the iPAS procedure for bandwidth allocation using stereotypes. The bandwidth share of certain stream depends on six features (delay, jitter, loss, power, resolution, and application) and the available bandwidth. The same procedure can be applied for any number of streams.

Notably, our stereotype-based resource allocation model considers the same probability distribution, i.e., each stereotype class consists of six features and each feature is further divided into five levels. Different probability distributions can also be considered in this model based on the number of features and classification of feature's linguistic values.

V. Simulation Test-Bed Setup

This section describes the simulation testing setup including multimedia traffic characteristics, test-bed configuration, and evaluation metrics used.

A. Multimedia Traffic

Four types of multimedia traffic: voice, video, best-effort, and background were used for transmission which is the same as the default traffic access categories in the IEEE 802.11e. The characteristics of the four traffic classes are shown in Table V. The voice traffic uses ITU-T G.711 [61] which has been widely deployed in commercial products like Skype. Video traffic adopted the H.264 codec [62] which is one of the most popular video codec for IP-based networks. Both best-effort traffic and background traffic were generated using the Pareto distribution traffic model to mimic bursty traffic. The encoding bit-rate of voice and video data are set to typical values specified in the standard and industry, i.e., 64Kbps for voice traffic and 1000Kbps for video traffic. The bit-rate of

$$\begin{aligned} P\left(S_i = LS_i k_i | Th1\right) &= PS_i^1 K_i \\ P\left(S_i = LS_i k_i | Th2\right) &= PS_i^2 K_i \\ & P\left(S_i = LS_i k_i | Th1 \ \& \ S_i = LS_i k_i | Th2\right) \\ &= P\left(S_i = LS_i k_i | Th2\right) + \left[1 - P\left(S_i = LS_i k_i | Th2\right)\right] \times P\left(S_i = LS_i k_i | Th1\right) \\ &= PS_i^2 K_i + \left(1 - PS_i^2 K_i\right) \times PS_i^1 K_i \end{aligned} \tag{13}$$

TABLE I
CLASSIFICATION OF FEATURES IN STEREOTYPE CLASSES

	Level 1	Level2	Level3	Level4	Level5
Delay	≤150ms	(150ms~400ms]	(400ms~1s]	(1s~5s]	>5s
Jitter	≤40ms	(40ms~50ms]	(50ms~60ms]	(60ms~70ms]	>70ms
Loss	$<10^{-5}$	10^{-5}~1%	1%~2%	2%~5%	>5%
Power Left	[100%~80%]	(80%~60%]	(60%~40%]	(40%~20%]	(20%~0]
Device Resolution	≥1024x768	(1024x768~768x4 80]	(768x480~480x36 0]	(480x360~320x24 0]	≤320x240
Application Type	VoIP	HD-Video	SD-Video	Best-effort	Background

TABLE II
GROUP OF FEATURES FOR STEREOTYPE-MEDIUM PRIORITY

Feature	(Linguistic Term, Probability)
Delay (ms)	(≤150,0.062), ((150~400],0.317), ((400~1000],0.399), ((1000~5000],0.184), (>5000,0.038)
Jitter (ms)	(≤40, 0.062), ((40~50],0.317), ((50~60], 0.399),((60~70], 0.184), (>70ms,0.038)
Loss	(≤10^{-5}, 0.062), ((10^{-5}~1%],0.317), ((1%~2%], 0.399),((2%~5%], 0.184), (>5%,0.038)
Power left	([0~20%],0.062), ((20%~40%],0.317), ((40%~60%], 0.399),((60%~80%], 0.184), ((80%~100%],0.038)
Device Resolution	(≤320x240, 0.062), ((320x240~480x360],0.317), ((480x360~768x480], 0.399), ((768x480~1024x768],0.184), (>1024x768, 0.038)
Application type	(VoIP, 0.062) (HD-Video,0.317)(SD-Video,0.399)(Best-effort,0.184)(Background,0.038)

TABLE III
GROUP OF SUGGESTIONS FOR STEREOTYPE-MEDIUM PRIORITY

Suggestion	(Linguistic Term, Probability)
Bandwidth share	(0~20%, 0. 062), (20%~40%, 0. 317), (40%~60%, 0. 399), (60%~80%, 0. 184), (80%~100%, 0. 038)

the best-effort and background traffic was set to a constant value equal 128Kbps and 100Kbps, and they can be variable in real life. The overhead introduced by the underlying protocols RTP/UDP/IP and TCP/IP is 40bytes.

B. Simulation Test-bed Setup

iPAS has been evaluated by using the NS-2.33[1] network simulator. The simulation topology shown in Fig. 4 includes one IPAS server and N servers communicating with N clients, over an IEEE 802.11b wireless network. The original NS-2 simulator was updated in the following aspects:

1) NS-2 was extended to include the IEEE 802.21 MIH function based on the IEEE 802.21 specifications. The 802.21 MIH server and client were implemented as C++ objects in NS-2 in a Linux environment.

2) Two wireless update patches are deployed in NS-2 set-up: NOAH[2] and Macro Fiore patch.[3] NOAH (No Ad-Hoc) was used to support direct communication between wireless nodes and access points and Marco Fiore's patch provided a more realistic wireless network environment (a step-wise decrease in bandwidth with increasing distance from the access point).

3) IEEE 802.11e EDCA patch for NS2[4] was also imported for the purpose of result comparison.

C. Evaluation Metrics

Six evaluation metrics are used to assess the iPAS performance. Firstly, *throughput*, *delay*, and *packet loss rate* are separately evaluated to study the effectiveness of both QoS differentiation and QoS provisioning. Secondly, the *fairness* between the demand and allocated bandwidth for all the traffic is studied.

[1]Network Simulator NS-2 [Online]. Available: http://www.isi.edu/nsnam/ns

[2]NOAH NS-2 extension [Online]. Available: http://icapeople.epfl.ch/widmer/uwb/ns-2/noah/

[3]M. Fiore patch [Online]. Available: http://perso.citi.insa-lyon.fr/mfiore/data/ns2_wireless_update_patch.tgz

[4]802.11e NS-2 patch [Online]. Available: http://www.tkn.tu-berlin.de/research/802.11e_ns2/

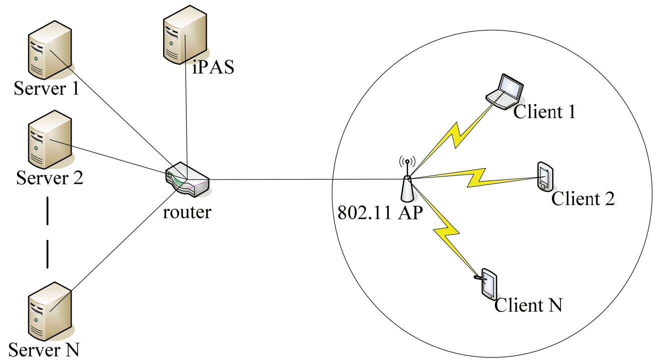


Fig. 4. Test bed topology.

TABLE IV
PROBABILISTIC RESULTS INDICATING THE MATCH DEGREE AND BANDWIDTH SHARE

Stereotype	Probability-Stream$_1$	Probability-Stream$_2$
High Priority (HP)	0%	0%
Medium High Priority (MHP)	57.05%	0%
Medium Priority (MP)	42.95%	11.82%
Medium Low Priority (MLP)	0%	88.18%
Low Priority (LP)	0%	0%
Bandwidth Share	**Probability-Stream$_1$**	**Probability-Stream$_2$**
0~20%	17.44%	0.91%
20%~40%	41.97%	6.39%
40%~60%	29.23%	23.41%
60%~80%	9.61%	46%
80%~100%	1.75%	23.29%

1) **Fairness.** In a system where streams make unequal demands for resources, one may want to measure fairness by closeness of the allocations to respective demands. Jain's fairness index [63], as shown in equations (18) and (19), was selected to indicate the fraction of demand fairness.

The value d_i is the demand of i^{th} stream and α_i is the corresponding allocation. The parameter n is the number of contending streams. The Jain's fairness index ranges between 0 and 1. For instance, a resource distribution algorithm with a fairness of 0.1 is unfair to 90% of the streams. It should be noticed that allocating bandwidth more than the demand does not make any stream user happier. A higher value of Jain's fairness index indicates a closer relationship between demand and allocation, and therefore better QoS guarantee.

$$f\left(x\right) = \frac{\left[\sum\limits_{i=1}^{n} x_i\right]^2}{n \times \sum\limits_{i=1}^{n} x_i^2}, \qquad x_i \geq 0 \tag{18}$$

$$x_i = \begin{cases} \frac{\alpha_i}{d_i} & if\ \alpha_i < d_i \\ 1 & otherwise \end{cases} \tag{19}$$

2) **Throughput**. The motivation of throughput investigation is to evaluate the QoS provisioning. The throughput was studied in two aspects: per-class throughput and aggregate throughput. The analysis of the per-class throughput achieved by stations within each traffic class (voice, video, best-effort, background) indicates the effectiveness of QoS distribution. Additionally, the aggregate throughput presents the utilization of the limited wireless channel resources.

TABLE V
CHARACTERISTICS OF FOUR TRAFFIC CLASSES

	Voice	Video	Best-effort	Back-ground
Traffic	ITU-T G.711 CBR	H.264 CBR 25fps/CIF	Pareto distribution traffic model	Pareto distribution traffic model
Underlying Protocol	RTP/ UDP/IP	RTP/ UDP/IP	TCP/IP	RTP/ UDP/IP
Encoding Bit-rate	64Kbps	1000Kbps	128kbps	100kbps
Data packet size	100 bytes	1024bytes	512bytes	512bytes

3) **Delay**. The transmission delay experienced by different stream reflects the effectiveness of the QoS differentiation and QoS provisioning. The delay represents the time duration from when data packets are sent to when they are received. Multimedia applications such voice and video are sensitive to the delay, and lower delay contributes to better perceived quality. The instantaneous delay is computed for each arrived multimedia packets, as given in equation (20), where $Time_{rcvd}$ and $Time_{sent}$ represent the time stamp when the packet is received and sent, respectively.

$$Delay = Time_{rcvd} - Time_{sent} \tag{20}$$

4) **Packet Loss Rate**. The packet loss rate in wireless networks is due to three major causes: 1) **Signal attenuation** - Packets might be dropped due to the weak signal received; 2) **Collision** - When multiple stations try to access the shared wireless channels simultaneously, collision occurs. The packets are dropped and each station increases their contention window size; 3) **Retry attempts** - When the number of retransmission for lost packets exceeds the retry threshold (this value is 7 for 802.11), the packet is dropped. Higher packet loss rate indicates a waste of bandwidth resources and degrades the received QoS. The calculation of packet loss rate, as given in equation (21), makes use of the total number of bytes sent by the server, $TotalSentBytes$, and the total number of bytes received by the client, $TotalRcvdBytes$.

$$Loss\ Rate = \frac{TotalSentBytes - TotalRcvdBytes}{TotalTxBytes} \tag{21}$$

D. **Experimental Scenario**

This scenario aims to evaluate the iPAS fairness, throughput, transmission delay, and packet loss rate. Each mobile station transmits a single traffic type. The test started by including four mobile stations and each transmitting a different traffic type (voice, video, best-effort, background). The number of mobile stations was then increased from 4 to 32 in steps of 4 in order to increase the overall offered load. For all the tests, it was configured that the number of stations transmitting each traffic type is the same. Consequently, the ratio of the number of traffic in system was set to 1:1:1:1 for voice, video, best-effort and background, respectively. To efficiently analyze the iPAS performance under variable network conditions, the normalized offered load was used. The normalized offered

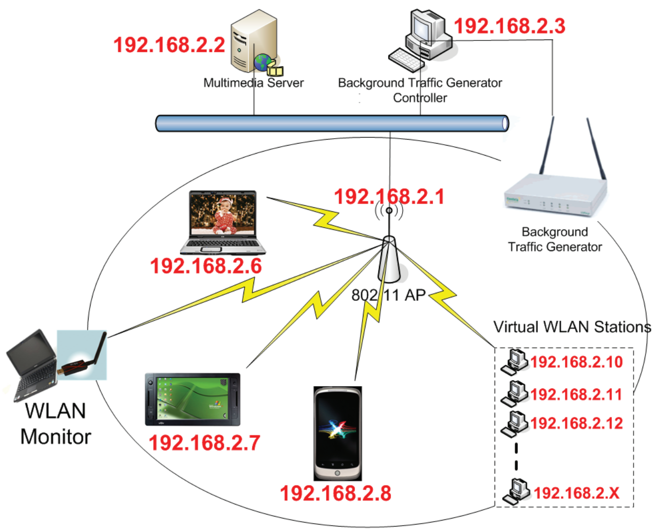


Fig. 5. Real life test-bed topology.

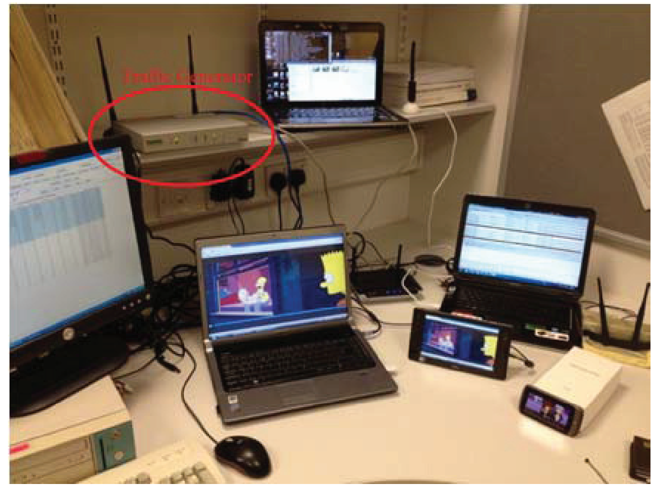


Fig. 6. Photo of the real life test-bed.

load was computed as the absolute offered load divided by the channel capacity which is determined with respect to the theoretical maximum capacity of the IEEE 802.11b mode, i.e. 7Mbps [47]. As the number of station increased, the corresponding normalized offered load has increased from 20% to 160% (the channel is overloaded). The experimental time duration was set to 150s. Specifically, the normalized offered load achieved 100% when the number of stations exceeded 20 at around 80s. In order to collect the statistics under stable conditions, all the measurements started 2s after the start of the simulations.

Moreover, the DropTail was adopted as the default queue algorithm and the queue length was set to 50. The experiments consider the situation where the active stations have always data frames to send.

VI. Real Life Test-Bed Setup

This section presents iPAS prototyping and related experimental results. The purpose of iPAS is to improve the original IEEE 802.11-based protocols by allocating bandwidth resources based on stream's priority. iPAS has been first deployed in a network simulation environment and the same topology was implemented in real life test-bed which will be described in this section. The real life test focuses on evaluating the performance of iPAS in terms of the delivered video quality. Video sequences are transmitted to three devices (laptop, tablet, and smartphone) over IEEE 802.11g network. The delivered video clips are recorded on each device.

The real life test-bed topology is shown in Fig. 5, and consists of: a multimedia server, a traffic generator (with traffic generator controller), an IEEE 802.11g wireless router, a network monitor, and an Android smartphone, a laptop, and a tablet PC. Fig. 6 further presents the photo of the test-bed based on the topology in Fig. 5. The multimedia server runs on a HP Pavillion dv3 laptop with Microsoft Windows 7 Home Edition x64, Intel Core 2 Duo T6600 at 2.2GHz and 4GB RAM. The multimedia software used on the laptop is the *Wowza Media Server*[5] which supports live or on-demand streaming to wireless devices. The traffic generator used is the LANForge-WiFIRE 802.11a/b/g from Candela Technologies,[6] which supports creating up to 32 virtual wireless stations. The traffic generator is capable to generate more than 45Mbps traffic by using various protocols such as TCP/IP, UDP/IP, etc. The WLAN monitor uses Wi-Spy DBx[7] which is capable of monitoring the interference levels and ensures iPAS system runs on non-interfered channel. A separate computer is needed to run the LANForge management software. As shown in Fig. 5, the IP address of traffic generator is 192.168.2.3. The IP address of the virtual WLAN stations starts from 192.168.2.10. Belkin N Wireless Router[8] is used to provide the local wireless network. MSU Video Quality Measurement Tool[9] software is used for assessing the objective video quality. It provides functionality for both full-reference and single-reference comparisons. A cartoon sequence, *The Simpsons Movie*, is used as it has been widely used in many other video tests. The video sequence was encoded with bit-rate of 1200Kbps, resolution of 800×448 and frame rate of 30fps.

Background traffic was introduced to the wireless network in order to evaluate the impact of network load on video transmission. The overall background traffic load was gradually increased up to around $19Mbps \sim 20Mbps$ since the practical bandwidth provided by 802.11g is in the range of 21Mbps–23Mbps. TCP and UDP traffic are used for both downlink and uplink traffic. The packet size of the downlink TCP flow ranges between 100bytes and 1472bytes with a transmission rate between 56Kbps and 1.5Mbps, which covers many widely used services. The packet size of the downlink UDP flow is 1472bytes with transmission rate 1Mbps.The uplink TCP and UDP traffic uses smaller packet size (i.e. 60bytes–640bytes) and lower transmission rate (i.e. 56Kbps–512Kbps) in comparison with that of downlink traffic.

[5]Wowza Media Server 3 [Online]. Available: http://www.wowza.com

[6]LANForge-WiFIRE, Candela Technologies [Online]. Available: http://www.candelatech.com/lanforge_v3/ct520_product.html

[7]Wi-Spy DBx [Online]. Available: http://www.metageek.net/products/wi-spy/

[8]Belkin N Wireless Router [Online]. Available: http://www.belkin.com

[9]MSU Video Quality Measurement Tool [Online]. Available: http://compression.ru/video/quality_measure/video_measurement_tool_en.html

The basement of the Electronic Engineering building at Dublin City University-Ireland was selected to deploy the wireless network for testing, as that location benefits from reduced interferences with the significant number of wireless networks operating on campus. The studied network has "iPAS" as SSID and is running on channel 6 (frequency 2.437GHz) with no other networks operating on the same or adjacent channels. The video transmission time is set to 320s. During the first 20s, there is no background traffic. From 20s to 320s, the number of background flows increases from 6 to 30 with 6 new flows added every 60s. For instance, during the 140s-200s time intervals, there are 6 TCP downlink flows, 6 UDP downlink flows, 3TCP uplink flows, and 3 UDP uplink flows. The received video sequences were recorded and compared against the original sequences using the MSU software in order to get PSNR values.

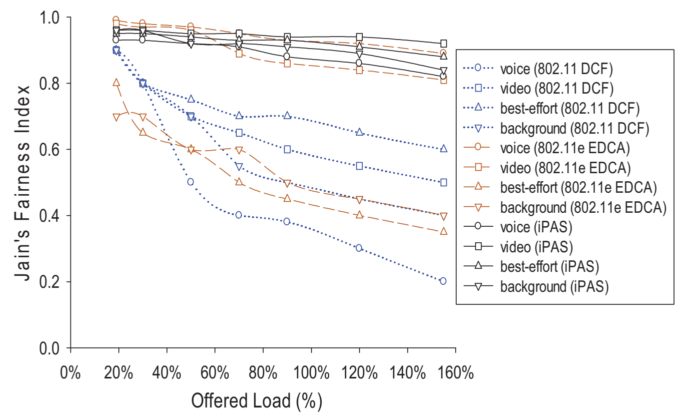


Fig. 7. Jain's fairness index for different schemes delivering voice, video, best-effort and background traffic, with increasing amount of offered load.

VII. Testing Results Analysis

This section presents the results analysis of both simulation and real life tests. The performance of iPAS in terms of fairness, throughput, delay, and packet loss are analyzed in subsections.

A. Fairness

In this simulation, the fairness performance of iPAS is compared with those of 802.11 DCF and 802.11e EDCA. Jain's fairness index is used to quantify the fairness level. The simulation experimental scenario was used.

Fig. 7 shows the fairness index for the three algorithms (iPAS, 802.11e EDCA, and 802.11 DCF) with various values of the offered load from 0% to 160%. All the stations are grouped into four traffic types: voice, video, best-effort, and background. The figure shows that the fairness is good for low amount of offered load, e.g., $F = 0.9$ (802.11 DCF), $F = 0.91$ (802.11e EDCA), and $F = 0.93$ (iPAS), when the offered load=20%. In the cases of 802.11 DCF and 802.11e EDCA, the fairness index decreases as the offered load increases. Specifically, the fairness decreases significantly for all traffic in 802.11 DCF and lower priority traffic in 802.11e EDCA. However, the fairness index of iPAS does not decrease significantly for any traffic class as the load increases. Take best-effort traffic for instance, when the offered load=100%, the fairness of 802.11 DCF and 802.11e EDCA decreases by about 21% and 40%, respectively, compared to the case of offered load =0%. In the case of iPAS, the decrease is around 5%. Additionally, iPAS achieves higher value of fairness index for certain traffic classes compared to 802.11 DCF and 802.11e EDCA for the entire range of offered load.

The rapid decrease of fairness for traffic in 802.11DCF and low priority traffic in 802.11e EDCA is mainly due to the 802.11 CSMA/CA mechanism. The Contention Window (CW) size for any traffic in 802.11 DCF and lower priority traffic (best-effort and background) in 802.11e EDCA belongs to the range of 15 to 1023 and 31 to 1023, respectively. While the higher priority traffic of 802.11e EDCA have lower CW range, i.e., 7 to 15 for voice traffic and 15 to 31 for video traffic. The increasing amount of offered load causes packet collisions, which determines the stations involved in collision enter the exponential backoff stage. The traffic with higher ranges of CW obtains less channel access opportunity and lower bandwidth achieved. Therefore, the fraction of demand fairness decreases significantly for traffic with higher ranges of CW.

In conclusion, the proposed scheme outperforms IEEE 802.11e in high network loaded conditions. IEEE 802.11e EDCA shows the best performance for video and voice services when the network loads are low (less than 60%). This is as IEEE 802.11e assigns very low contention window size for voice and video flows (i.e. 7–15 for voice and 15–31 for video), resulting in high channel access opportunity. At the same time, iPAS runs on top of the IEEE 802.11 protocol, where contention window size is defined by a higher range (i.e. 15–1023 for voice and video flows) than those of IEEE 802.11e. The lower the contention window range is, the higher the channel access opportunity is obtained. Additionally, IEEE 802.11e performed the worst for best-effort and background services due to the starvation issue. In general, IEEE 802.11 DCF provided the worst overall results as 802.11 DCF lacks service differentiation and network conditions adaptation mechanisms.

iPAS has better fairness for all traffic types with the increasing amount of the load. This can be explained by the fact that the throughput allocated to each stream is proportional to the stream priority and the wireless network condition.

B. Throughput

This test investigated the throughput achieved by multimedia traffic when 802.11 DCF, 802.11e EDCA, and iPAS were used in turn. The simulation experimental scenario was used.

Fig. 8 presents the aggregate throughput received for voice and video traffic class, and Fig. 9 shows the aggregate throughput experienced for best-effort and background traffic class. Each traffic class has the same number of flows or stations. When the total offered load is lower than 100%, (i.e., up to 20 stations), there is no significant difference between iPAS, 802.11 DCF and 802.11e EDCA for all traffic types, since there is enough bandwidth to transmit all of the traffic. It is observed that the aggregate throughput for the voice traffic class is the lowest among the four traffic

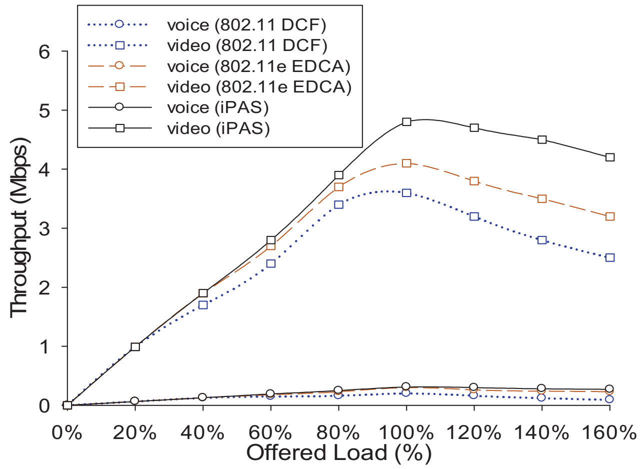


Fig. 8. Aggregate per-class throughput for different schemes delivering voice and video traffic with increasing amount of offered load.

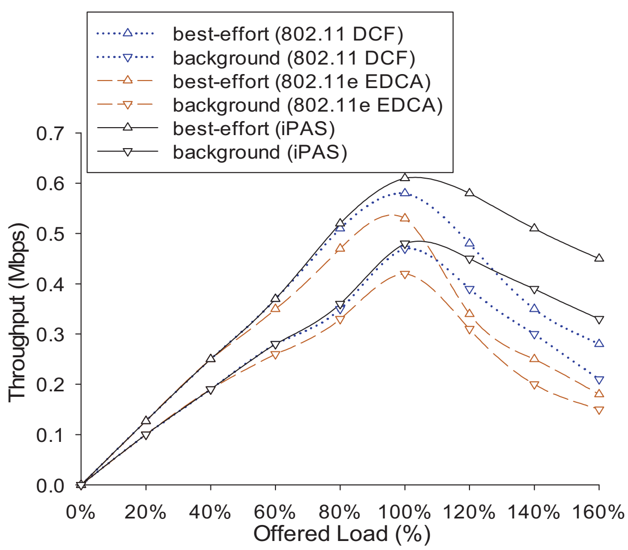


Fig. 9. Aggregate per-class throughput for different schemes delivering best-effort and background traffic with increasing amount of offered load.

classes for the three schemes due to the low bit-rate and packet size. In the case of 802.11 DCF and 802.11e EDCA, the throughput decreases significantly when the total offered load exceeds 100%. Note that the throughput of the lower priority traffic (best-effort and background) in 802.11e EDCA drops more rapidly than the higher priority traffic (voice and video) along with the increasing amount of loads. For instance, when offered load=140%, the aggregate throughput of best-effort traffic class of IEEE 802.11e EDCA decreases by about 53%, compared to the case of offered load =100%. The aggregate throughput of video traffic in EDCA decreases with around 14%. This is because the traffic with lower priority has higher values for AIFS and contention window, meaning lower opportunity to obtain the channel access. Such phenomenon observed for 802.11e EDCA is also called starvation. iPAS avoids the starvation problem of low priority traffic under high offered load. As shown in Fig. 10, the aggregate throughput of best-effort traffic class and background traffic class decreases

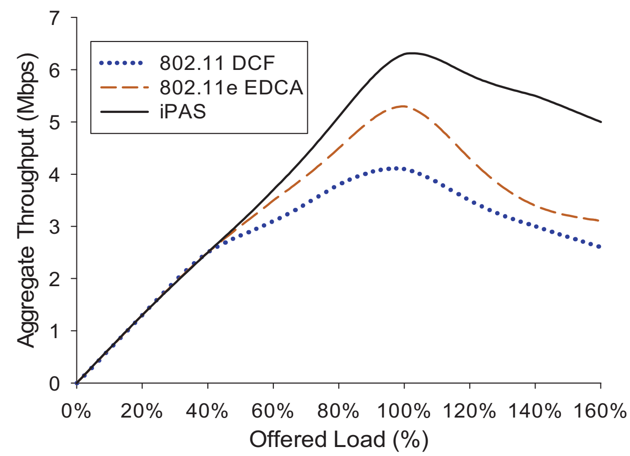


Fig. 10. Aggregate throughput for different schemes delivering voice, video, best-effort and background traffic, with increasing amount of offered load.

in a linear fashion due to the increase of the collision rate. Additionally, iPAS allocates the highest throughput for both voice and video traffic in comparison to those of 802.11 DCF and 802.11e EDCA, demonstrating good QoS provisioning for multimedia services. When the total offered load equals 100%, the available bandwidth estimated by *MBE* is around 6.5Mbps. iPAS allocates 6% to voice, 73.8% to video, 12.7% to best-effort data, and 7.5% to background traffic.

Fig. 10 shows the aggregate throughput for all traffic classes. It is observed that iPAS obtains higher aggregate throughput than both 802.11 DCF and 802.11e EDCA with increases of 38% and 20% respectively, for the entire offered load. Furthermore, under high traffic load (i.e., offered load>100%), the aggregate throughput of iPAS and 802.11 DCF decreases linearly while that of 802.11e EDCA decreases abruptly.

The high aggregate throughput of iPAS is due to the performance of the bandwidth estimation algorithm which estimates precisely the network capability.

C. Delay

This simulation compared the performance of iPAS with those of 802.11 DCF and 802.11e EDCA, with respect to the transmission delay. The simulation experimental scenario was used in this test.

To begin with, the focus of this work is on delay differentiation performance. Fig. 11 presents the average transmission delay experienced by voice, video, best-effort and background traffic in iPAS, 802.11 DCF and 802.11e EDCA. Since 802.11 DCF does not differentiate traffic based on the traffic classes, the delay of voice and video are significantly higher than those of 802.11e EDCA and iPAS. In the case of both voice and video traffic, the delay experienced by 802.11e EDCA is slightly better than that of iPAS for low traffic load. This is as the contention window sizes of voice and video traffic in 802.11e are lower than those used when iPAS was employed.

Fig. 11 also shows that, in the cases of iPAS and 802.11e-EDCA, traffic with higher priority (voice and video) experienced significantly lower delay than traffic with lower priority (best-effort and background). This phenomenon confirms that

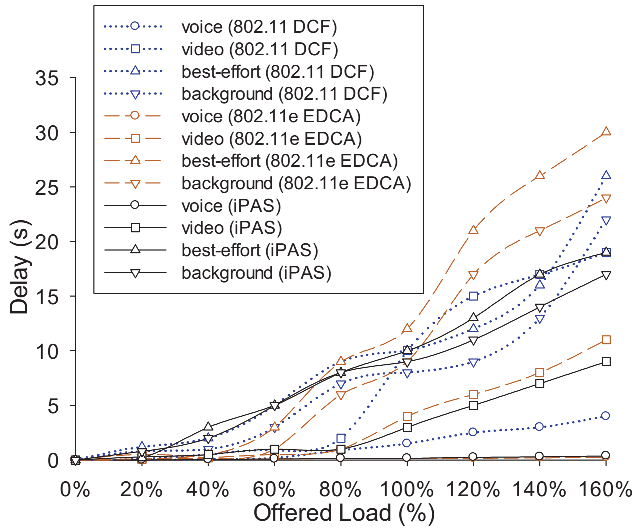


Fig. 11. Average delay for different schemes delivering voice, video, best-effort and background traffic, with increasing amount of offered load.

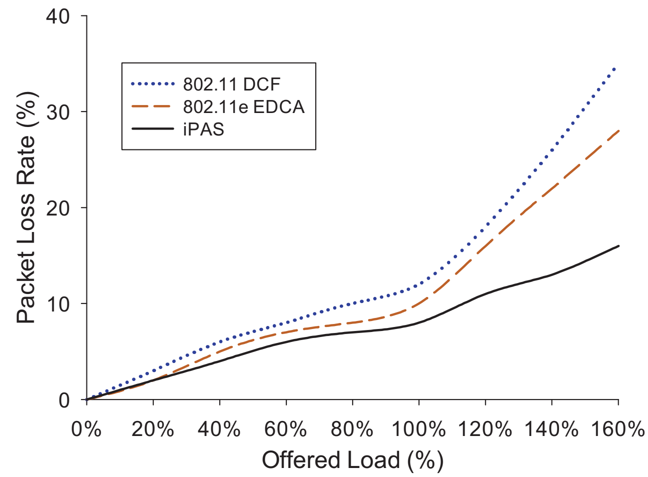


Fig. 12. Average packet loss rate for 802.11DCF, 802.11e EDCA, and iPAS.

both iPAS and 802.11e EDCA provide QoS differentiation for different traffic types.

It is observed from Fig. 11 that, although 802.11e EDCA can provide low delay for both voice and video traffic under low network load (i.e., traffic load <80%), the delay experienced by best-effort and background traffic increases dramatically under heavy offered load. In contrast, iPAS can provide low and smooth delay for the entire range of traffic load.

D. Packet Loss Rate

In this simulation, the packet loss rate for delivering voice, video, best-effort, and background traffic in iPAS, 802.11 DCF, and 802.11e EDCA was separately investigated. The simulation experimental scenario was used. Fig. 12 shows the results, where the load (X-axis) represents the overall load produced by the multimedia traffic and Y-axis represents the packet loss rate.

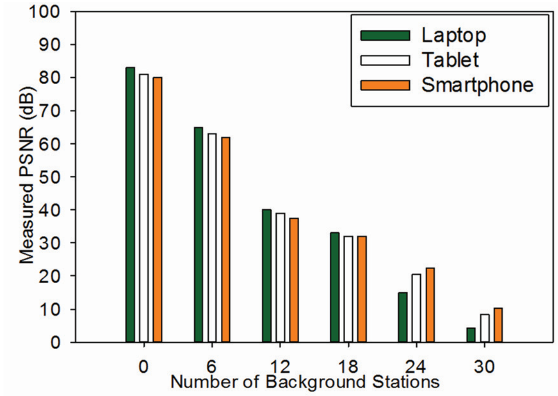
It is observed that iPAS provides the lowest packet loss for the entire range of offered load. Under low traffic load (i.e., load < 30%), the difference between the packet loss rate for the three schemes is not significant. In case of heavy traffic load (i.e., offered load>120%), the packet loss rate of 802.11 DCF and 802.11e EDCA increased significantly. For load from 100% to 160%, iPAS obtained packet loss rates lower with 18% and 34%, compared to EDCA and DCF, respectively. This is as iPAS uses the wireless channel more efficiently due to an increase in the accuracy of estimated bandwidth from MBE.

Fig. 13(a) and Fig. 13(b) present the PSNR values measured during video delivery using TCP with IEEE 802.11 and iPAS, separately. In low loaded ($N = 0$, 6, 12) and average loaded ($N = 18$) network conditions, video delivered using iPAS has higher PSNR than that of IEEE 802.11. For instance, when $N = 12$, PSNR values measured at laptop, tablet PC, and smartphone using iPAS increase by 28.6%, 23.6%, and 25.1%, compared to the case of IEEE 802.11. The reason is that iPAS can adapt the transmission rate of video traffic based on the accurate bandwidth estimation algorithm, which efficiently reduces the packet loss probability and improves the received video quality.

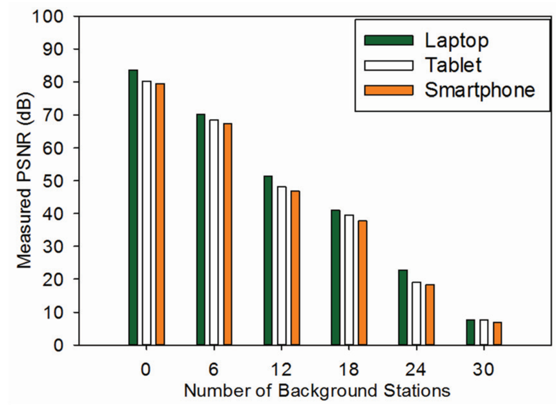
E. Video Quality

Additionally, in Fig. 13(a), under high loaded ($N = 24$) and overloaded ($N = 30$) network conditions, PSNR measured at the laptop is lower than that of tablet PC and smartphone. This can be explained by that laptop has more powerful data process capability than tablet and smartphone and requires more bandwidth to play the video. Since IEEE 802.11 allocates fair channel access for the three devices, laptop suffers more quality degradations, in comparison to the tablet PC and smartphone. Fig. 13(b) shows the PSNR measured at the three devices using TCP with iPAS. It is shown that, in comparison with the equal channel access mechanism of IEEE 802.11, iPAS over IEEE 802.11improves the PSNR at the laptop by re-allocating certain bandwidth share from the tablet PC and smartphone: 1) when $N = 24$, PSNR measured at the laptop using iPAS increases by 52% and PSNR values measured at the tablet PC and smartphone using iPAS decrease by 6.8% and 17.9%; 2) when $N = 30$, PSNR value measured at laptop using iPAS increases by 76.7% and PSNR measured at the tablet PC and smartphone using iPAS decrease by 8.3% and 33%, respectively.

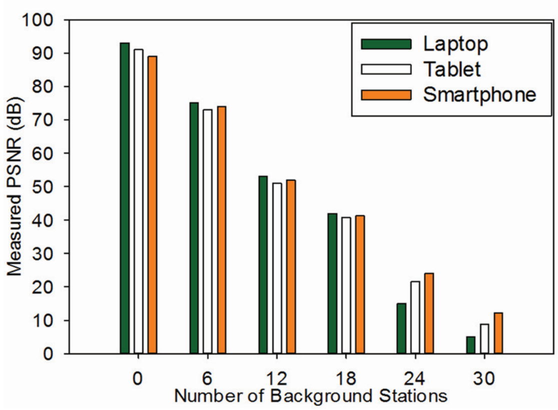
Fig. 13(c) and Fig. 13(d) illustrate the PSNR values measured when UDP video is delivered via the equal channel access mechanism of IEEE 802.11 and iPAS over IEEE 802.11, separately. It is shown that, generally, UDP traffic can result in higher PSNR than when TCP is used. The primary reason is that TCP uses flow control which causes retransmission delay and thus degraded the video quality. Also, TCP protocol has much higher overhead than that of UDP. For instance, when $N = 18$, PSNR values measured at the laptop, tablet PC, and smartphone using UDP with IEEE 802.11 increase by 27.3%, 27.5%, and 29.1%, in comparison with those of using TCP with IEEE 802.11. Also, similar with TCP traffic as shown in Fig. 13(a) and (b), UDP video traffic delivered using iPAS



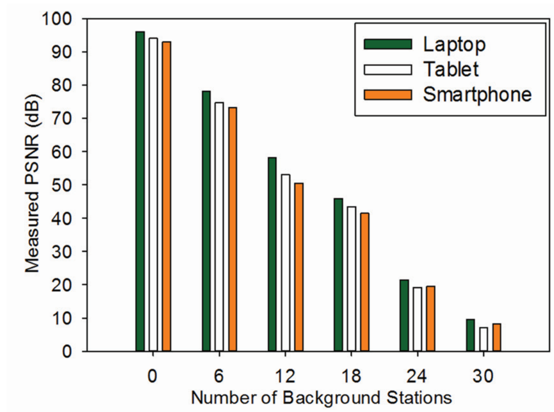
(a) PSNR achieved using TCP with 802.11



(b) PSNR achieved using TCP with iPAS



(c) PSNR achieved using UDP with 802.11



(d) PSNR achieved using UDP with iPAS

Fig. 13.

has higher PSNR than that of IEEE 802.11. Fig. 13(d) presents the PSNR measured at the three devices when delivering UDP traffic with iPAS. It is shown that, in comparison with IEEE 802.11, iPAS improves the PSNR measured at the laptop by dividing certain bandwidth share from the tablet PC and smartphone: 1) when $N = 24$, PSNR value measured at laptop using iPAS increases by 43.3% and PSNR measured at the tablet PC and smartphone using iPAS decrease by 11.1% and 18.3%; 2) when $N = 30$, PSNR value measured at laptop using iPAS increases by 88.2% and PSNR measured at the tablet PC and smartphone using iPAS decrease by 19.3% and 31.4%, respectively.

VIII. Conclusions

This paper proposes **an intelligent Prioritized Adaptive Scheme (iPAS)** to provide both QoS differentiation over IEEE 802.11 networks. iPAS algorithm assigns dynamic priorities to multimedia streams and suggests a proportional bandwidth share according to the results of a bandwidth allocation process. This process employs a novel stereotype-based bandwidth allocation solution which considers both network delivery QoS-related parameters such as delay, jitter, and packet loss rate.

The following conclusions have been drawn. 1) iPAS achieves higher and more stable fairness index than 802.11 DCF and 802.11e EDCA for all four service types with increasing network load; 2) iPAS can differentiate the bandwidth share among different streams according to the priority level. iPAS allocates higher throughput for both voice and video traffic in comparison with those of 802.11 DCF and 802.11e EDCA, demonstrating good QoS support for multimedia services. It is also observed that iPAS achieves the highest aggregate throughput for the entire range of network loads tested. The aggregate throughput of iPAS is higher than those of 802.11 DCF and 802.11e EDCA with up to 38% and 20%, respectively; 3) iPAS and 802.11e EDCA both provide delay differentiation for the four service types (i.e., voice traffic experience the lowest delay and best-effort traffic achieve the highest delay); 4) iPAS obtains the lowest packet loss rate for the entire range of network loads. In the case of heavy traffic load, packet loss rate was lower with 18% and 34%, for iPAS than the rates recorded for 802.11 DCF and 802.11e EDCA, respectively.

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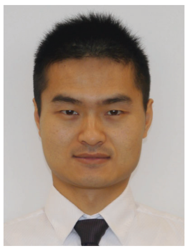

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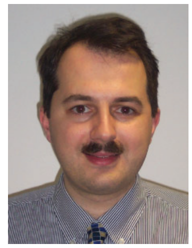

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