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iVoIP: an intelligent bandwidth management scheme for VoIP in WLANs

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Abstract Voice over Internet Protocol (VoIP) has been widely used by many mobile consumer devices in IEEE 802.11 wireless local area networks (WLAN) due to its low cost and convenience. However, delays of all VoIP flows dramatically increase when network capacity is approached. Additionally, unfair traffic distribution between downlink and uplink flows in WLANs impacts the perceived VoIP quality. This paper proposes an intelligent bandwidth management scheme for VoIP services (iVoIP) that improves bandwidth utilization and provides fair downlink–uplink channel access. iVoIP is a cross-layer solution which includes two components: (1) *iVoIP-Admission Control*, which protects the quality of existing flows and increases the utilization of wireless network resources; (2) *iVoIP-Fairness* scheme, which balances the channel access opportunity between access point (AP) and wireless stations. *iVoIP-Admission Control* limits the number of VoIP flows based on an estimation of VoIP capacity. *iVoIP-Fairness* implements a contention window adaptation scheme at AP which uses stereotypes and considers several major quality of service parameters to balance the network access of downlink and uplink flows, respectively. Extensive simulations and real tests have been performed, demonstrating that iVoIP has both very good VoIP capacity estimation and admission control results. Additionally, iVoIP improves the downlink/uplink fairness level in terms of throughput, delay, loss, and VoIP quality.

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1 Introduction

IEEE 802.11 wireless local area networks (WLAN) have been widely deployed as the last-mile Internet access in homes, universities and enterprises [1]. Along with the significant growth of WLAN connections, the popularity of multimedia delivery services is also increasing [2], including web-browsing, e-mails, on-line games, video streaming, and social networking. Meanwhile, popular VoIP software products, such as Skype[1] and Google Talk,[2] have been supported by the majority of mobile consumer devices and have attracted millions of users [3]. An interesting investigation shows that more than 50 % of voice calls originate from indoor WLANs [4].

Figure 1 shows how multiple mobile devices can be connected to 802.11 WLANs via access points (APs). Mobile VoIP users in different WLANs communicate through a remote VoIP server. Unfortunately, the original 802.11 protocol does not support quality of service (QoS) provisioning for real time applications such as VoIP. Consequently, many solutions have been proposed to provide QoS support for multimedia services and some focus on VoIP applications in wireless networks [5–7].

Admission control and downlink/uplink fairness are two critical issues when delivering VoIP services over IEEE 802.11 networks. First, it is imperative to have an efficient admission control mechanism for VoIP services in order to provide a consistent level of QoS, especially in terms of

[1] Skype, Available [Online] http://www.skype.com.

[2] Google Talk, Available [Online] http://www.google.com/talk.

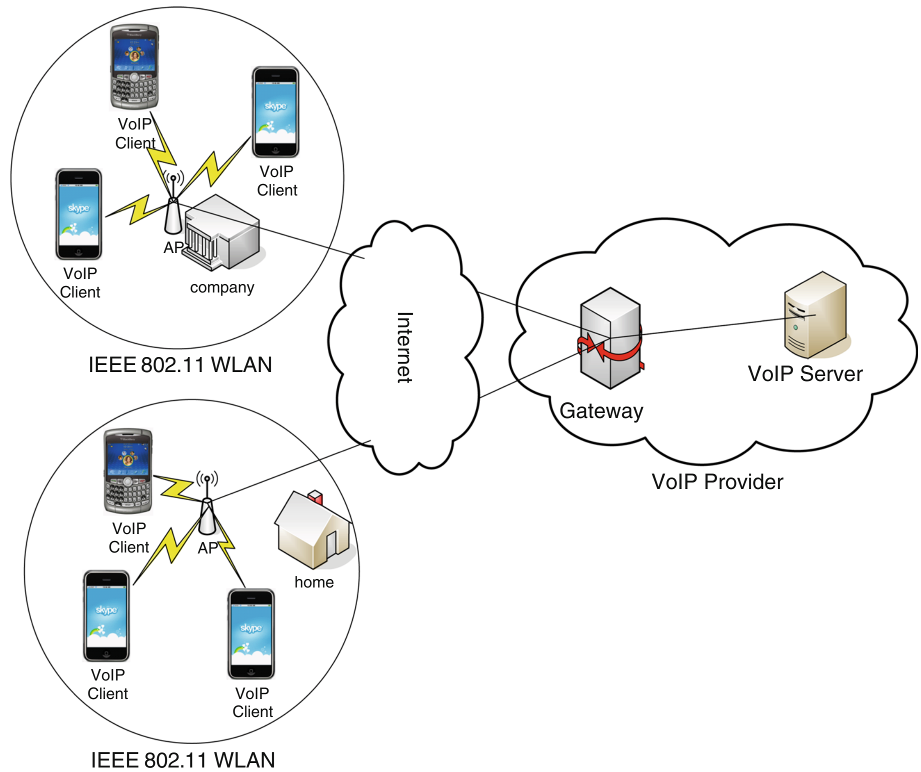


Fig. 1 Scenario of IEEE 802.11-based VoIP applications

delay, to existing traffic. Most of the current admission control techniques are based on either bandwidth estimation [8–11] or application level QoS requirements [12–14]. Secondly, the QoS level of VoIP services is also affected by the unfair traffic distribution between downlink and uplink in the IEEE 802.11 WLAN [15]. In original 802.11 protocols in infrastructure mode, the downlink flows obtain less channel access opportunity than the uplink flows, as AP has only one MAC queue and one backoff access entity. IEEE 802.11e [16] standard updates the MAC layer of the original 802.11 standard by using four access categories (ACs) that use separate queues and backoff instances. However, multiple downlink flows belonging to an AC of the AP have the same priority with the uplink flows belonging to the same AC of the stations. Therefore, the downlink/uplink unfairness problem might still be caused by the flows from the same AC. Consequently, there is a need for a fair traffic distribution between the AP (at the core of the traffic management) and wireless stations.

Many fairness-oriented solutions have been developed focusing on modifying the IEEE 802.11 MAC settings including Contention Windows (CW) [17–19], Transmission Opportunity (TXOP) [20], and Arbitration Inter Frame Space (AIFS) [21, 22]. However, these fairness-based solutions do not take into account the QoS level of both downlink and uplink traffic and do not consider upper layer parameters. For instance, in order to fairly balance QoS levels, it is necessary to make use of the values of several QoS-related parameters in conjunction such as throughput, delay, and loss, since they are all critical for the VoIP traffic. Additionally, none of the previous research works provide both admission control and downlink/uplink fairness support for VoIP services in IEEE 802.11 networks.

This paper proposes a novel intelligent bandwidth management solution (iVoIP) for VoIP services in IEEE 802.11 WLANs. iVoIP guarantees desired QoS levels by introducing a new *bandwidth estimation-based admission control mechanism*. Additionally, a novel *contention window (CW) adaptation scheme* is designed to achieve QoS fairness between downlink and uplink. This scheme relies on a stereotypes-based algorithm, which utilizes the ratio between the major QoS parameter values (i.e. throughput, delay, and loss) measured for downlink and uplink traffic, respectively. Stereotypes for managing groups of users were first introduced by Rich [23] in the Grundy system and they are still widely used by many QoS-oriented adaptive solutions [24, 25].

This paper is structured as follows. Section 2 discusses the related works on admission control and fairness schemes in WLANs. Section 3 presents the principle of iVoIP and the architecture of the iVoIP system. Sections 4 and 5 introduce the experimental setup and present results analysis, respectively. Conclusions are drawn in Sect. 6.

2 Related works

This section briefly summarizes related works regarding admission control and downlink/uplink fairness techniques in wireless networks.

2.1 Admission control techniques

The purpose of admission control in wireless networks is to restrict the traffic load in order to maintain QoS at certain level for the existing flows. The core mechanism of admission control is the admission decision policy.

An admission control algorithm based on the IEEE 802.11e EDCA [16] protocol is proposed in [9] by taking into account the dynamic network conditions. A new flow is admitted only if it will not cause the overall admitted flows used bandwidth to exceed the wireless network capacity, which is estimated based on a 802.11 MAC analytical model [26]. Another admission control scheme, *Traffic Stream-Admission Control* (*TS-AC*) [10], was developed to maximize the wireless network utilization. When a new VoIP call request comes, *TS-AC* first exploits the flow's characteristics and then compares the flow's requested bandwidth to the measured unused bandwidth. The call is admitted if the comparison's outcome is positive, otherwise, it is rejected. In [11], a novel call admission control scheme with a polling-based scheduling policy for CBR traffic is proposed in IEEE 802.11e wireless LAN. The proposed transmit-permission policy for HCF controlled channel access (HCCA) protocol can predict the maximum delay for each packet and derive sufficient conditions so that all the CBR sources satisfy their time constraints to ensure QoS levels. Simulations are conducted and results show that the proposed scheme provides a high throughput with respect to the system load. In [12], a new IEEE 802.11e flow-based admission control scheme is developed to control and adjust channel access parameters with the channel condition variations. Each wireless station classifies the arriving request and calculates the admission control parameters for the flow based on its maximum tolerable collision rate (CR_{max}). When CR_{max} is less than the current channel collision rate, the required delay and dropping rate of this flow cannot be satisfied under the current channel conditions, and therefore, the flow request should be rejected. In [13], a novel admission control scheme is proposed to improve VoIP quality in multiservice wireless cellular networks. It considers delays as a resource shared by all system users and regularly measures VoIP packet delays. The admission control scheme verifies if the current VoIP delays added to the estimated resource demand of the incoming session are higher or lower than the admission threshold for that type of service. If higher, the access of the incoming session is blocked, otherwise, it is allowed. A recent call admission control scheme [14] probes the network to determine if a VoIP flow can be supported with acceptable QoS. The probing procedure utilizes Internet Control Message Protocol (ICMP) echo messages (ping) to measure Round-Trip Time (RTT), jitter, and packet loss. The call is then admitted if the measured RTT, jitter, and loss do not exceed predetermined admission thresholds.

In conclusion, performance of admission control schemes for wireless networks largely depends on two factors: (1) accurate analysis of the wireless network conditions; (2) QoS levels required by applications.

2.2 Downlink/uplink fairness techniques

Fairness issue between 802.11 downlink and uplink flows refers to fair channel access between AP and wireless stations. Current fairness-oriented schemes in 802.11 wireless networks focus on adapting MAC parameters, i.e. minimum or maximum CW, TXOP, AIFS, etc.

In [17], fairness between downlink and uplink flows in IEEE 802.11 networks is considered, where uplink flows dominate over downlink flows in terms of channel usage. The proposed scheme dynamically controls the minimum size of contention window (CW_{min}) at AP according to a computed optimal ratio between the packet transmission rate of downlink and uplink flows. Packet rate is defined as a function of the numbers of uplink and downlink flows, as well as the minimum contention window sizes at the AP and wireless terminals. Hirantha Sithira Abeysekera et al. [17] firstly provides a simplified analysis of the packet rate ratio R, which does not strongly depend on the number of uplink flows (N_U). This is then supported by the mean field approximation analysis. Secondly, an explicit formula for the optimal CW_{min} at the AP was derived, assuming $N_U = 1$, and using it as a quasi-optimal CW_{min} at the AP for all $N_U \geq 1$. In [19], VoIP capacity is increased through adaptive frame aggregation and downlink/uplink bandwidth fair distribution. In particular, the solution uses minimum contention window (CW_{min}) adaptation, which can precisely control the bandwidth distribution among wireless stations. The bandwidth share ratio is inversely proportional to CW_{min} ratio. The CW_{min} for the AP is set to the 1/(k − 1) of that for actively transmitting wireless stations. Different from existing TXOP-based fairness scheme, the solution in [20] dynamically controls the contention window and TXOP according to the packet error rate and the number of stations. In [20], fairness levels between downlink and uplink flows are improved by controlling both TXOP limit and CW_{min} size. The principle of the scheme relies on the TXOP differentiation approach, which allows AP to send multiple frames per one channel access. In error-free environments, fairness between downlink and uplink is achieved by setting AP's TXOP limit to the time required for n_d (the number of downlink stations) frame transmissions. When channel error is large, increasing AP's TXOP limit is not efficient. Therefore, AP sets its TXOP limit for transmitting n_d frames, and dynamically adjusts CW_{min} according to channel error rate. Chou et al. [21] proposes a distributed solution to control the channel usage of each stations in IEEE 802.11e networks, where the AIFS is prioritized for different service classes. The principle is to modulate the distribution of station's AIFS and analyze the threshold of AIFS that can achieve fair channel access between downlink and uplink. Leith et al. [22] improves TCP fairness in 802.11e WLANs in terms of asymmetry between TCP downlink and uplink flows. IEEE 802.11e AIFS and CW_{min} parameters are used to assign higher priority to AP. A small value of AIFS and CW_{min} results in near strict prioritization of TCP ACKs at the AP. A larger value of CW_{min} is set at each wireless station in order to reduce contention between competing TCP ACKs. Additionally, 802.11e TXOP mechanism provides a fine grained way for prioritizing TCP downlink data packets.

IEEE 802.11e EDCA improves DCF by using access categories to support priorities between different traffic classes. Most current APs adopt EDCA since it is distributed and easier to implement. The idea of TXOP is introduced by the IEEE 802.11e HCF Controlled Channel Access (HCCA) protocol. HCCA allows wireless stations to send multiple contention-free packets and assigns higher channel access opportunity in comparison with EDCA. Nevertheless, HCCA is not widely deployed in ad-hoc networks due to its centralized approach. However, to the best of our knowledge, none of the previous research works consider fairness between downlink and uplink traffic in 802.11 networks in terms of QoS levels, as proposed in this paper.

3 iVoIP system architecture

iVoIP performs admission control based on bandwidth availability and improves fairness according to QoS levels between downlink and uplink traffic. Figure 2 illustrates the TCP/IP protocol stack model-based architecture of the iVoIP system. It consists of three components: (1) Model-based Bandwidth Estimation (MBE) [27, 28], which is located at application layer and estimates the available bandwidth resources of IEEE 802.11 network; (2) iVoIP-Admission Control, which is located at application layer and limits the amount of VoIP traffic admitted into an IEEE 802.11 network; (3) iVoIP-Fairness, which is deployed at MAC layer and balances the downlink/uplink channel access opportunity. *iVoIP-Admission Control* module receives feedback (i.e. available bandwidth) from MBE and *iVoIP-Fairness* module receives information (i.e. QoS levels) from the MAC layer. VoIP traffic is delivered using the Real-time Transport Protocol (RTP) [29] and feedback information is collected and delivered using the mechanism provided by IEEE 802.21 [30] framework. Details of each major component of the iVoIP system are introduced next.

The decision-making procedure of iVoIP is shown in Fig. 3. When a new VoIP $flow_i$ requests joining the wireless LAN, the iVoIP-Admission Control module estimates the overall available bandwidth and decides whether to admit or not the new flow in order to avoid network overload and congestion. If $flow_i$ is admitted to the wireless network, then iVoIP-Fairness module is used to assign proper channel access opportunity to $flow_i$, and in meanwhile, provide fair access between existing downlink and uplink flows. This is done by using the stereotypes-based resource allocation and adjusting the contention window size of $flow_i$. Detailed description of the system modules will be provided in the following sections.

3.1 Model-based bandwidth estimation (MBE)

MBE [28] estimates the available bandwidth based on novel TCP and UDP throughput models for IEEE 802.11 WLANs. *MBE* is an application layer solution and uses an *IEEE 802.21 Media Independent Handover* (*MIH*) *Function* module to monitor the transmission-related information such as packet loss, round trip delay, and packet size. *MBE* estimates the bandwidth in three steps as follows.

First, *MBE* combines the TCP throughput model [31] and the IEEE 802.11 DCF [32] model, in order to consider both TCP congestion control and wireless characteristics. The bandwidth achieved by a TCP flow (B^{TCP}) is given in Eq. (1), where b is the number of packets acknowledged by a received ACK, P_{retr}^{TCP} is the probability of packet retransmission, $MRTT$ is the transport layer round-trip time between sender and receiver, and MSS denotes the maximum segment size. T_o is the timeout value to trigger retransmission.

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

Unlike TCP, the UDP protocol does not support packet retransmissions and therefore the UDP over WLAN throughput model should consider this. Hence, the terms

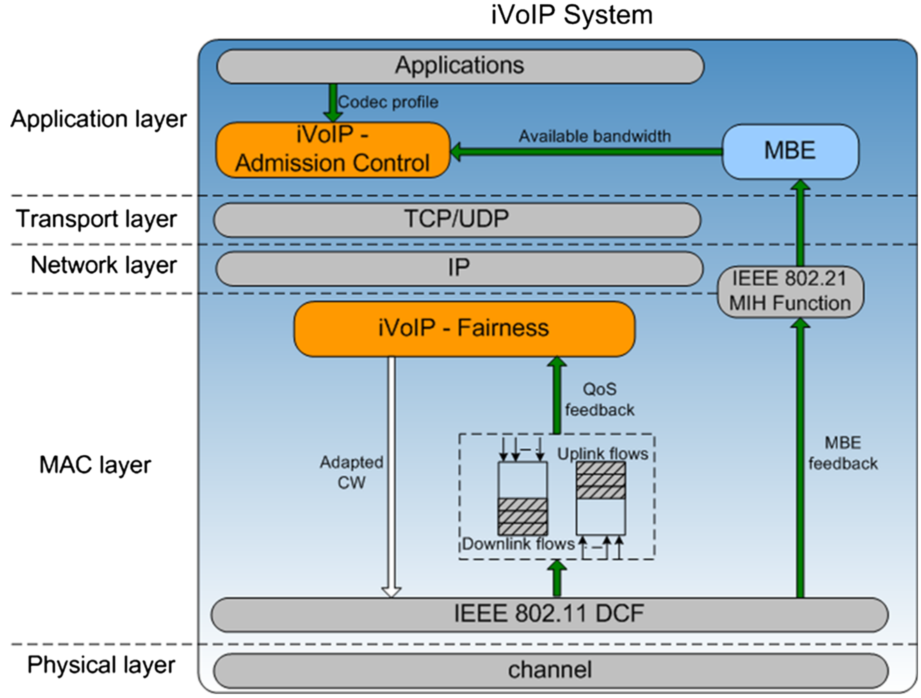


Fig. 2 Block architecture of iVoIP system

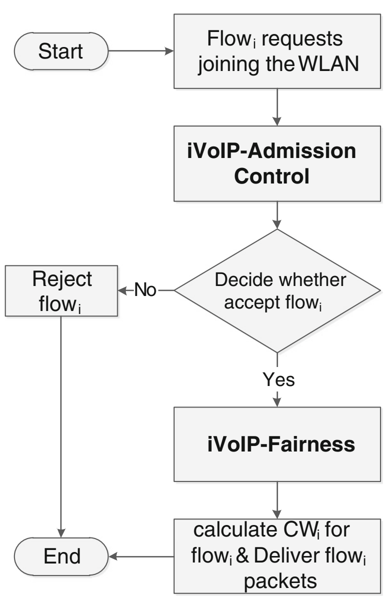


Fig. 3 System algorithm of iVoIP

P_{retr} and *MRTT* defined in Eq. (1) which consider TCP fast retransmission and timeout respectively should be removed in *MBE*'s UDP version. According to [28], the probabilities of retransmission $\left(P_{retr}^{UDP}\right)$ and successful transmission when UDP traffic runs over IEEE 802.11 networks can be given as shown in Eqs. (2) and (3), respectively, where P_{DCF} is the transmission loss probability and P_{drop} refers to the drop probability:

$$P_{retr}^{UDP} = P_{DCF} + P_{drop} \tag{2}$$

$$P_{succ}^{UDP} = 1 - P_{retr}^{UDP} \tag{3}$$

The average delay, *Delay_UDP*, for successfully transmitting the individual UDP packet could be written as in Eq. (4). T_{retr}^{UDP} and T_{succ}^{UDP} represent the average delay for retransmission and successfully transmission. Details for the derivation of $T_{retr}^{UDP}$and T_{succ}^{UDP} are given in [28].

$$Delay\_UDP = (1 - P_{tr}) \times \sigma + P_{retr}^{UDP} \times T_{retr}^{UDP} + P_{succ}^{UDP} \times T_{succ}^{UDP} \tag{4}$$

Next, *MBE* estimates the throughput of a UDP flow using Eq. (5), which analyses the packet transmission probability and delay. The parameter *Payload* is the total information transmitted during time period from T_0 to T_1, and *Delay_UDP* denotes the average delay for successfully transmitted individual UDP packets.

$$B^{UDP} = \frac{\int_{T_0}^{T_1} \frac{Payload}{Delay\_UDP} dt}{T_1 - T_0} \tag{5}$$

Finally, *MBE* derives a formula predicting the achievable bandwidth when TCP and UDP flows co-exist in 802.11 networks, as shown in Eq. (6). The parameter w is the bandwidth weight factor since TCP and UDP have different bandwidth requirements.

$$\sum B^{\mathrm{TCP}+UDP} = w \times \sum B^{\mathrm{UDP}} + (1 - w) \times \sum B^{TCP} \tag{6}$$

MBE has been modeled, implemented, and tested through simulations and real life testing. The results show that *MBE* performs very well in conditions with variable packet size, dynamic wireless link rates and different channel noise [28].

3.2 iVoIP-Admission Control

The principle of the *iVoIP-Admission Control* is to limit the number of VoIP calls based on the predicted VoIP network capacity so that the QoS of existing VoIP flows will not be degraded, while maintaining high network bandwidth resource utilization. The VoIP transport capacity is defined as the number of VoIP flows which can be supported by a wireless network for a given average QoS level. The admission policy of the proposed *iVoIP-Admission Control* scheme is as follows. When a new VoIP call intends to be sent over an IEEE 802.11 network already carrying N VoIP flows, if the throughput of the $N + 1$ flows exceeds the predicted VoIP capacity, then the new VoIP call is rejected; otherwise, it is accepted. *iVoIP-Admission Control* is deployed at the application layer of the iVoIP system and utilizes two types of information: (1) network available bandwidth as it is estimated by *MBE*; (2) VoIP flow information, as it is known, at the codecs including codec bit-rate, packetization interval and protocol header size.

The VoIP codecs information depends on the application deployed, but can be precisely gathered. Consequently, an accurate estimation of the VoIP capacity is critical for an efficient admission control. There are two steps performed by the *iVoIP-Admission Control* to compute the VoIP capacity.

In step one, the bandwidth required by an individual VoIP flow is computed as shown in Eq. (7). The parameter *bitrate* is the voice packet coded bitrate, *pktIntvl* represents the voice packet interval and *headerSize* is the packet header size.

$$Individual\ Flow\ Bandwidth_i = bitrate_i + \frac{headerSize_i}{pktIntvl_i} \quad (7)$$

In step two, the VoIP capacity is computed using Eq. (8). The *Available Bandwidth* refers to the overall network bandwidth resources which can be used for the VoIP flows and is estimated by *MBE* using Eq. (6). The *Individual Flow Bandwidth* is the bandwidth required by each VoIP flow which is given by Eq. (7),

$$VoIP\ capacity = \frac{Available\ Bandwidth}{\sum_{i=1}^{N} Individual\ Flow\ Bandwidth_i} \quad (8)$$

For example, we consider delivery of VoIP flows which use the ITU-T R.G.711 encoding standard [33]. Table 1 presents the specifications of the widely used VoIP codec. For instance, ITU-T G.711 codec uses a voice sampling rate of 8,000 samples per second and adopts non-uniform quantization with 8 bits to represent each sample, resulting in a content bit-rate of 64 kbps. There is a direct relationship between the inter-packet delivery interval and voice packet size. Higher values of inter-packet delivery interval lead to larger voice packet size. G.711 selects 20 ms and 160 bytes as default values for the inter-packet interval and voice packet size, respectively. According to [34] and [35], large voice packet size might result in higher VoIP capacity but lower tolerance to packet loss and jitter, in comparison with the case when shorter voice packet sizes are used. Most VoIP packets are delivered over RTP/UDP/IP, so 40 bytes packet headers are often added to the payload. For G.711 VoIP services, when the inter-packet interval equals 20 ms, VoIP flow bit-rate is 64 kbps and header size is 40 bytes, if the available bandwidth is 3 Mbps, the VoIP capacity is 31 flows computed according to Eqs. (7) and (8). iVoIP scheme was designed to use the widely deployed constant bit-rate VoIP codec such as ITU-T G.711, ITU-T G.729 [36], iLBC [37], etc. According to Eq. (7), three parameters are needed as input: bit-rate, packet interval and packet header payload. In the case of iLBC, the bit-rate is constant for a certain interval (20 or 30 ms). Take 20 ms interval for instance, the bit-rate and packet header payload are 15.2 kbps and 38 bytes, respectively. Therefore, iVoIP can be adapted to iLBC codec straightforwardly.

3.3 iVoIP-Fairness

The proposed MAC-layer *iVoIP-Fairness* scheme provides QoS-based fair channel access between downlink and uplink VoIP traffic. It takes the QoS parameters (i.e. throughput, delay and loss) as input and adapts the contention window (CW) size at the AP using a stereotypes-based adaptive solution. Details of the principle of the stereotypes-based adaptation are presented next.

3.3.1 Stereotypes-based data structure

The stereotypes-based structure is designed to store information about streams and provide this data to the CW adaptation scheme. Each stereotype (*Th*) is defined for a subgroup of streams and 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 represent actions to be performed. Each feature F_i is associated to 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*). A similar structure is defined for each suggestion S_j, which has also associated the linguistic terms $LS_j = (LS_{j1}, LS_{j2}, \ldots, LS_{jr}, \ldots, LF_{js})$ and

Table 1 Voice standard specifications

Codec	Sampling rate (kHz)	Bitrate (kbps)	Inter-packet interval (ms)	Voice packet size (bytes)
ITU-T G.711	8	64	20	160
ITU-T G.729	8	8	20	20
iLBC	8	15.2	20	38

Table 2 Group of features for a stereotype

Features	(Linguistic term, probability)
F_1	$(LF_{11}, PF_{11}), (LF_{12}, PF_{12}), \ldots, (LF_{1q}, PF_{1q})$
F_2	$(LF_{21}, PF_{21}), (LF_{22}, PF_{22}), \ldots, (LF_{2q}, PF_{2q})$
…	…
F_m	$(LF_{m1}, PF_{m1}), (LF_{m2}, PF_{m2}), \ldots, (LF_{mq}, PF_{mq})$

Table 3 Group of suggestions for a stereotype

Suggestions	(Linguistic term, probability)
S_1	$(LS_{11}, PS_{11}), (LS_{12}, PS_{12}), \ldots, (LS_{1q}, PS_{1q})$
S_2	$(LS_{21}, PS_{21}), (LS_{22}, PS_{22}), \ldots, (LS_{2q}, PS_{2q})$
…	…
S_m	$(LS_{m1}, PS_{m1}), (LS_{m2}, PS_{m2}), \ldots, (LS_{mq}, PS_{mq})$

probabilistic values PS_{jr}. Tables 2 and 3 present the group of features and suggestions for a stereotype.

The Poisson distribution represents the probability of a number of events occurring during a time period and is used to determine the probability associated with the linguistic terms. The occurrences of events are independent from one-another. Equation (9) shows the Poisson distribution function where u is the shape parameter (from 0 to 15) 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. 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 been used by [38] for network parameters modeling, and resulted in very good results. The maximum value of the normal distribution is close to 0.15 ($x = 7$, $u = 7$) and the minimum value is 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. Figure 3 shows the example when iVoIP uses five stereotypes: High Priority (HP), Medium–High Priority (MHP), Medium Priority (MP), Medium–Low Priority (MLP), and Low Priority (LP). 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. As shown in the Fig. 4, the peak value of the Poisson function increases when u_k gets closer to zero.

$$pois(x, u) = \frac{u^x \times \exp(-u)}{x!} \tag{9}$$

Considering that feature F_i has a list of linguistic terms, where the list length is q, the probabilistic values for each term PF_{ij} are computed as in Eq. (10). The value i is the index of the feature and j is the index of the linguistic term.

$$\begin{aligned} &PF_{ij} = Average(pois(x_j, u)) \\ &\quad x_j \in [[15/q] \times (j-1), [15/q] \times j] \end{aligned} \tag{10}$$

The stereotype-based structure is initialized and updated using the following three stage process:

Stage 1: User Classification

The purpose of the stereotype-based classification is to determine the stereotype class the stream belongs to and with what probability. Equation (11) presents the format of data associated to a data stream D, where F_i is the name of the ith feature and LF_iV_i represents the linguistic value of the ith feature.

$$D = ((F_1, LFV_1), (F_2, LFV_2), \ldots, (F_m, LFV_m)) \tag{11}$$

A degree of match between the stream and each stereotype class is computed as in Eq. (12) based on the probability theory.

$$\begin{aligned} M(Th) = p(Th|F_1 = LFV_1, \ldots, F_m = LFV_m) = p(Th|F_1 \\ = LFV_1) \times \cdots \times p(Th|F_m = LFV_m) \end{aligned} \tag{12}$$

The computation of each factor is computed using the Bayes rule, as shown in Eq. (13). It is considered that all the stereotypes have the same probability distribution, therefore, $p(Th)$ is the reciprocal of the number of stereotype classes.

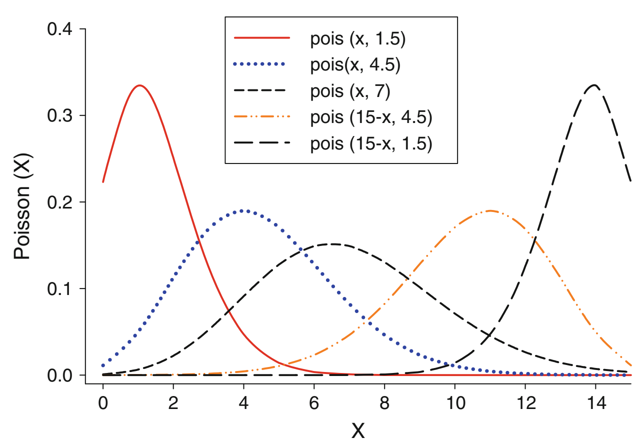


Fig. 4 Poisson distribution for five stereotypes

$$p(Th|F_i = LFV_i) = \frac{p(F_i = LFV_i|Th) \times p(Th)}{p(F_i = LFV_i)} = \frac{PFV_i \times p(Th)}{p(F_i = LFV_i)} \quad (13)$$

Stage 2: Suggestion Determination

After the stream is associated with stereotype classes, the suggestion determination is performed to determine the best CW size to be allocated to the AP.

First, for each stereotype, the strength of each suggestion has to be computed by considering the probability for the stream D to belong to this stereotype class, as in Eq. (14), where S_i is the name of the ith suggestion and LSV_i represents the linguistic term of the ith suggestion.

$$p(S_i = LSV_i|Th) = p(S_i = LSV_i|Th) \times M(Th) \quad (14)$$

Second, the combination of the suggestions of each stereotype class is computed by Eq. (15), based on the probabilistic theory. $P(E_n)$ is the probability of an event E_n to appear and $p(E_1\ \&\ E_2\ \&\ \ldots\ \&\ E_n)$ represents the probability for events $E_1\ \&\ E_2\ \&\ \ldots\ \&\ E_n$ to appear simultaneously.

$$p(E_1\ \&\ E_2\ \&\ \ldots E_n) = p(E_n) + [1 - p(E_n)] \times P(E_1\ \&\ E_2\ \&\ \ldots E_{n-1}) \quad (15)$$

Stage 3: Update

The stereotypes are updated using a four-step algorithm:

- Track the user related parameters regarding each feature (F_i) of the stereotype. For each stereotype (Th), re-calculate the probability (PF_{ip}) value that is associated with the linguistic term LF_{ip}.
- Repeat the *User Classification* process described before, in order to determine the stereotype class the current stream belongs to and with what probability.
- Repeat the *Suggestion Determination* process to produce weighted suggestions based on the probability with which the stream belongs to each stereotype classes.
- Merge the suggestions found with the previously determined suggestions to determine the adaptation approach.

3.3.2 CW adaptation using the stereotype-based structure

Five stereotype classes are used to represent five fairness levels between downlink and uplink: "Bad", "Poor", "Normal", "Good", and "Excellent". Theoretically, there can be any number of fairness levels. In this paper, five fairness levels are selected in order to correspond to the five *Mean Opinion Score* levels in the ITU-T Recommendations P.800 [39]. Three QoS performance parameters, $Throughput_{down/up}$, $Delay_{down/up}$, and $LossRate_{down/up}$ are modeled as stereotype features, representing throughput ratio, delay ratio and loss ratio between the downlink and uplink communication channels, respectively. The computation of the downlink/uplink ratio for each QoS parameters is described in the next section. The linguistic terms for the features of each stereotype are denoted using five ranges representing the possible ratios between the downlink and uplink as follows: ">1.5" (LF_{i1}), "1.2–1.4" (LF_{i2}), "0.9–1.1" (LF_{i3}), "0.6–0.8" (LF_{i4}), "<0.6" (LF_{i5}). Parameter i indicates the ith stereotype.

The initial CW size for the AP and each wireless station is selected randomly from 0 to CW_{min} (=15) according to 802.11 standards. The idea of adapting the AP's CW using the stereotype-based structure is to associate a CW size according to stereotypes suggestions. Additionally, the CW range (i.e., 15–1023, as specified in 802.11 standards) is divided into four levels representing five suggestion linguistic terms as follows: "0–15" (LS_{j1}), "16–267" (LS_{j2}), "268–519" (LS_{j3}), "520–771" (LS_{j4}), "772–1023" (LS_{j5}). LS_{j1} represents the initial CW range. LS_{j2}, LS_{j3}, LS_{j4}, and LS_{j5} are the equally divided CW ranges, each of which is set to 251. Parameter j indicates the jth stereotype. The adapted CW size of AP is then computed using the three stage process introduced in the previous section: *User Classification*, *Suggestion Determination* and *Update*. The adapted CW size of AP is then computed using the three stage process introduced in the previous section: User Classification, Suggestion Determination and Update. iVoIP divides the original CW range (i.e. 15–1023) into five levels (i.e. "0–15", "16–267", "268–519", "520–771", "772–1023") that are associated to the five stereotypes levels ("Bad", "Poor", "Normal", "Good", and "Excellent"), respectively. The five stereotypes levels are mapped to five downlink/uplink QoS-parameters (delay, throughput, delay) ratio: ">1.5", "1.2–1.4", "0.9–1.1", "0.6–0.8", "<0.6". The instant CW size of AP is determined by selecting a random the mid value from the suggested a CW interval. For instance, if the QoS-parameters ratio achieved by downlink/uplink flows falls into "0.9–1.1", the related CW interval is "268–519" and the mid value 394.

3.3.3 Computation of stereotype feature values

The proposed CW adaptation scheme collects transmission-related information at AP as well as at the wireless stations. A feedback mechanism is employed to carry the information collected at the wireless stations to the AP. The feedback information is delivered using RTP Control Protocol (RTCP) protocol [40], which allows for defining additional user defined packet types in the Receiver Report packet header. This information consists of throughput downlink/uplink ratio, delay downlink/uplink ratio, and packet loss rate downlink/uplink ratio.

Throughput downlink/uplink ratio The throughput downlink/uplink ratio is considered fair when the throughput in both directions has equal values and therefore $Throughput_{down/up}$ ratio equals one. Equation (16) illustrates how the downlink/uplink throughput ratio is computed.

$$Throughput_{down/up} = \frac{Throughput_{AP}}{\sum_{i=1}^{N} Throughput_{STAi}} \quad (16)$$

Equation (16) makes use of the throughput at the AP ($Throughput_{AP}$) and the aggregation of the throughput at the wireless stations ($Throughput_{STAi}$), where i indicates the ith wireless station and N is the number of wireless stations. Both $Throughput_{AP}$ and the overall throughput at the wireless stations can be measured at the MAC layer of the AP.

Delay downlink/uplink ratio The downlink and uplink delay distribution is fair when the two communication directions will process the same amount of traffic during a sample interval. For example if the packet size is identical, it is fair to have the AP sending N packets to the stations (downlink) and have the N wireless stations sending N packets to the AP (uplink) in the same time period.

In order to achieve delay downlink/uplink fairness, it is noted that there are three types of delay for packet transmissions via wireless: (1) *Queuing delay*; (2) *MAC delay*; (3) *Propagation delay*. Queuing delay represents the total duration of time that packets have to wait in the queues. MAC delay is caused by the contention mechanism of CSMA/CA protocols, which may also include some uplink-downlink unfairness. However queuing delay is by far the largest delay that causes delay unfairness between downlink and uplink [19, 41] and therefore MAC delay is not considered in this paper. Propagation delay is dependent on the distance and signal propagation speed only. In the VoIP system, the wireless propagation delay is the same between downlink and uplink because they use the same medium and as the packet size is the same, it does not influence the downlink/uplink fairness.

The queuing delay is determined by many factors such as queue arriving rate, queue service rate, current queue size, etc. Since there is a desire that the proposed algorithm be deployed at the AP without modifying the wireless stations, the current queue size of the ith wireless station $QSize_{STAi}$ is estimated using Eq. (17). The parameter $AvgPktSize_i$ is the average packet size received from the ith wireless station during the sampled interval. λ_{STAi} represents the arrival rate of the packets entering the queue, and depends on the VoIP encoding scheme. For instance, a 64 kbps VoIP traffic implies $\lambda_{STAi} = 64$ kbps. The VoIP encoded information is delivered to AP via the feedback mechanism. $NumRcvdPkts_i$ is the number of packets received by the AP from the ith wireless station in the sample time period. *Time* is the sampling time duration selected to investigate the queue state.

$$QSize_{STAi} = AVGPktSize_i \times (Time \times \lambda_{STAi} - NumRcvdPkts_i) \quad (17)$$

The burst arrival of packets results in a random queue size. If the sample interval is too small, it is possible that it may be no packet transmissions due to the bursty nature of traffic. Otherwise, too big interval value reduces the update frequency leading to inaccuracy in the queue size estimation. The sampling interval is selected based on the Nyquist theorem [42], given in Eq. (18), where w is the signal frequency. The purpose of computing an optimal sample interval is to alleviate the aliasing phenomenon due to the traffic busrtiness.

$$Time = SamplingInterval = 1/(2 \times w) \quad (18)$$

$$Delay_{down/up} = \frac{QDelay_{AP}}{\sum_{i=1}^{N} QDelay_{STAi}} = \frac{QSize_{AP}/\mu_{AP}}{\sum_{i=1}^{N} QSize_{STAi}/\mu_{STAi}} \quad (19)$$

The delay downlink/uplink ratio is given in Eq. (19), where $QDelay_{AP}$ and $QDelay_{STAi}$ are the average queuing delay at the AP and ith wireless station, respectively. $QSize_{AP}$ and $QSize_{STAi}$ are the amount of packets (in bits) waiting in the queue at the AP and ith wireless station, respectively. μ_{AP} and μ_{STAi} are the service rate of the queue at the AP and ith wireless station, respectively (i.e. the rate at which bits leave the queue). The values of $QSize_{AP}$, μ_{AP} and μ_{STAi} can be monitored and measured at the AP. The values of $QSize_{STAi}$ are computed using Eqs. (17) and (18).

Packet loss rate downlink/uplink ratio The end to end packet loss is caused by the following factors: (1) *Queue Drop*: Packets can be dropped at the queue depending on the queuing management algorithms adopted. For instance, in the First Come First Service (FCFS) queues such as DropTail [43], packets are dropped when the queue has filled its capacity; in Random Early Detection (RED) queues [44], packets are dropped with certain probability depending on the queue size; (2) *Channel Error*: packets can be lost due to the wireless channel error; (3) *Retransmission Limit*: when packet retransmission reaches a retry limit defined by the 802.11 MAC, the packet is dropped. (4) *Collision*: collisions occur when multiple wireless stations (uplink) attempt to transmit the packets simultaneously; packets affected by collisions are dropped. There are no collisions among the downlink flows since the AP is the unique 802.11 DCF object generating traffic in the downlink mode. The packet loss rate downlink/uplink ratio is given in Eqs. (20) and (21).

$$BERLoss_{AP} = BER \times \mu_{AP} \tag{20}$$

$$LossRate_{down/up} = \frac{QDropRate_{AP} + BERLoss_{AP} + RETRANLoss_{AP}}{\sum_{i=1}^{N} Loss_{STAi}} \tag{21}$$

The parameters $QDropRate_{AP}$, $BERLoss_{AP}$, and $RETRANLoss_{AP}$ represent the packet loss at the AP caused by queue drop, channel error and retransmission limit, respectively. $QDropRate_{AP}$ and $RETRANLoss_{AP}$ are captured at the MAC layer of the AP, and $BERLoss_{AP}$ is computed using Eq. (20), where μ_{AP} is the service rate of the AP queue. The parameter $Loss_{STAi}$ is the packet loss rate of the ith wireless station which is measured at the AP based on packet sequence numbers.

4 Experimental testing setup

The performance of the proposed *iVoIP-Admission Control* and *iVoIP-Fairness* schemes were evaluated via both real-life and simulations. Testing setup is described next, including the characteristics of the VoIP traffic used, test-bed configuration, and experimental scenarios.

4.1 Real life test-bed setup

Real life measurements were conducted using *SIPp* [45], which is an open-source SIP traffic generation tool that supports generation of multiple VoIP calls from callers to callees. The callers and callees were deployed in *HP Pavilion Entertainment* laptops and associated with *User Agent Client* (*UAC*) and *User Agent Server* (*UAS*), respectively. *UAC* and *UAS* are basic *SIPp* user agent instances that support establish and release VoIP calls. *UAC* transmitted SIP messages using RTP/UDP protocol. The voice codec used was ITU-T R.G.711 [33] with a default inter-packet interval of 20 ms. *Wireshark* [46] software was used to measure the throughput at *UAS*. A *LinksysWRV210* access point wireless router was used to support IEEE 802.11b WLAN-based network. In these experiments, PHY data rate was 11 Mbps and all the stations were stationary. RTS/CTS mechanism was disabled due to the short packet size of the VoIP service.

4.2 Simulation test-bed setup

Simulation-based tests were performed using Network Simulator NS-2.[3] The simulation topology used the wired-cum-wireless "dumbbell" topology with a 100 Mbps bandwidth and 20 ms delay wired bottleneck, as shown in Fig. 5. Each wireless station receives a single VoIP flow from a wired station through the bottleneck link and the same IEEE 802.11 AP. Constant Bit Rate (CBR) VoIP traffic was generated using the ITU-T R.G.711 codec [33], with payloads of 160 bytes/packet. The bit-rate of CBR was set to 64 kbps representing an inter packet interval of 20 ms. DropTail [41] queue with a limit of 100 packets was set to each wireless station. The RTS/CTS mechanism was disabled as voice packets are small. The MAC layer parameters were configured according to the 802.11b specifications [47], where $CW_{min} = 31$, $CW_{max} = 1023$, *DCF Interframe Space* ($DIFS$) = 50 µs, *Short Interframe Space* ($SIFS$) = 10 µs, and slot time = 20 µs. The initial *CW* size of AP was selected randomly from 0 to 31. Two additional wireless patches are deployed in the NS-2: NOAH[4] and Marco Fiore patch.[5] NOAH (No Ad-Hoc) was used for simulating the infrastructure WLAN environment, whereas Marco Fiore's patch provides a more realistic wireless network environment. As shown in Fig. 6, the scenario considers multiple mobile nodes located at random locations around AP. Darker colors area indicates areas with higher bit-rates and situated closer to the AP. According to the IEEE 802.11b specification, the data rate degrades step-wise taking values at four levels: 11, 5.5, 2, and 1 Mbps. The distance between mobile nodes and AP ranges from 30 to 120 m in order to test the effect of variable channel quality.

4.3 Experimental scenarios

In order to study the performance of both proposed *iVoIP-Admission Control* and *iVoIP-Fairness* solutions, three separate scenarios were designed.

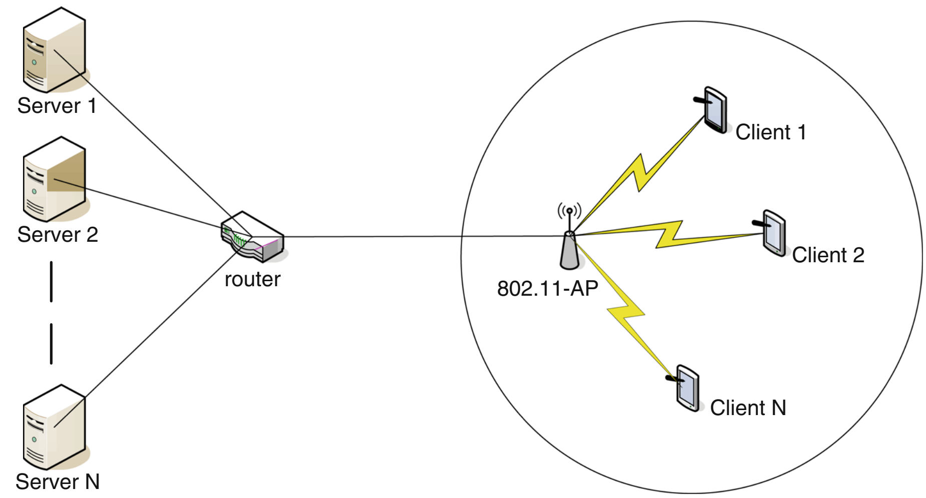
1. *Scenario 1* This scenario aims to evaluate the accuracy of the VoIP capacity provided by the *iVoIP-Admission Control*. iVoIP was deployed in the simulation test bed. The real life test-bed was used to provide measured capacity. The estimated VoIP capacity was computed using Eq. (5) and then compared with the measurement results. The SIPp UAC was configured to generate VoIP flows at a rate of 0.2 cps (calls per second). The number of VoIP flows was increased up to 50. ITU-T R.G.711 voice codec was used with the packetization interval increased from 10 to 50 ms with step of 10 ms. Two performance metrics were studied: (1) the estimated and measured VoIP capacities with variable inter-packet intervals; (2) one-way delays of VoIP flows when iVoIP and the original IEEE 802.11 protocol with no admission control mechanism were employed in turn. Notably, the VoIP capacity from the real life measurement is obtained according to ITU-T

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

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

[5] M. Fiore patch, http://www.telematica.polito.it/fiore.

Fig. 5 Test bed topology



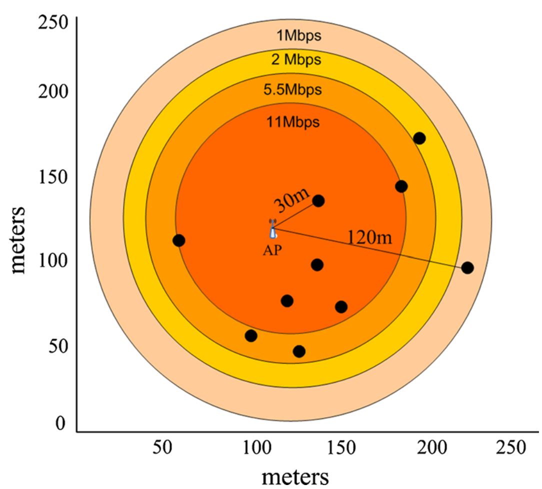


Fig. 6 Random topology in simulation

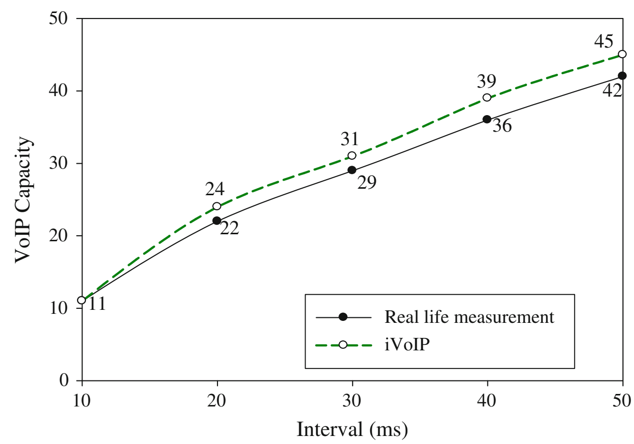


Fig. 7 VoIP capacity with variable inter-packet interval for ITU-T R.G.711

R.G.114 [48], which recommends a maximum of a 150 ms one-way delay for VoIP. Therefore, assuming the VoIP capacity from the measurement equals M, which indicates that delays experienced by all the M VoIP flows are below 150 ms.

2. *Scenario 2* This scenario investigates the performance of the iVoIP-Fairness module. iVoIP was modeled and compared against the original 802.11 protocol and a state-of-the-art fairness scheme, Dynamic-CW [17]. The number of wireless stations increased from 0 up to the VoIP capacity estimated in scenario 1. iVoIP performance was studied using Jain's fairness index [49] in terms of delay, throughput, and loss.

3. *Scenario 3* The third scenario studies the influence of iVoIP admission control mechanism and fairness scheme on VoIP quality in terms of the E-model [50] which describes VoIP quality. An experiment was conducted based on test scenario 1, where each wireless station maintains a single VoIP flow. The number of wireless stations N in this study is increased up to 30.

5 Result analysis

The three experimental scenarios in Sect. 4 were performed separately to study the performance of iVoIP and its major two contributions: the admission control scheme and the fairness algorithm. Result analyze of each experiment are presented next in details. The real-life test bed was built to measure the actual VoIP capacity in order to evaluate the performance of iVoIP admission control module. In Fig. 7 and Table 4, the measured VoIP capacities are obtained from real-life implementation and tests and while the estimated VoIP capacities are taken from simulations.

Table 4 VoIP capacity with variable inter-packet interval

Inter-packet interval (ms)	VoIP capacity	
	iVoIP	Measurement
10	11	11
20	24	22
30	31	29
40	39	36
50	45	42

5.1 iVoIP-Admission Control

This section studies the performance of VoIP capacity estimation and admission control algorithm, according to scenario 1. Figure 7 and Table 4 present VoIP capacity with variable inter-packet interval for ITU-T R.G.711 and shows that there is a good match between analytical results (iVoIP) and measurement results. Specifically, the capacity predicted by iVoIP is optimistic in comparison with the real life measurement. For instance, in the case of codec interval equals 20 ms, the VoIP capacity achieves 24 and 22 for iVoIP and measurement, respectively. This can be explained by two reasons: (1) the 150 ms delay constraints in real life measurements are not included in iVoIP (as indicated in scenario 1); (2) various real environmental factors (e.g. fading, shadowing, etc.) impact the measurements. Figure 7 also shows that larger packetization interval results in higher VoIP capacity due to higher channel efficiency (e.g. lower overhead and less interference). For instance, when the codec interval equals 40 ms, the VoIP capacity increases by 62.5 and 63.6 % for iVoIP and measurement, respectively, in comparison with that of 20 ms.

Figure 8 and Table 5 show the average one-way delay experienced by an individual VoIP flow using IEEE 802.11

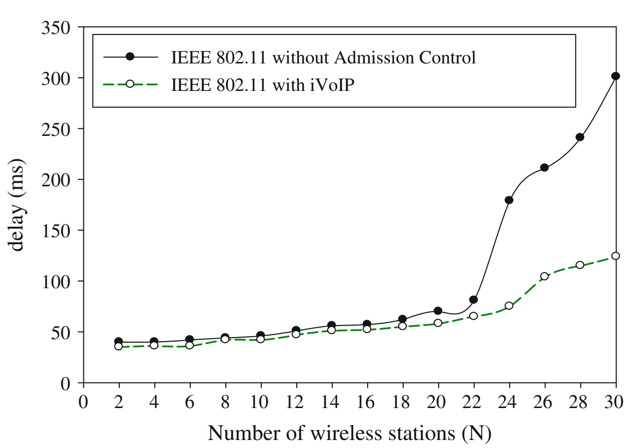


Fig. 8 One-way delay of VoIP service with increasing number of VoIP flows (codec: ITU-T R.G.711, codec interval: 20 ms)

Table 5 One way delay of VoIP service with increasing number of VoIP flows (codec: ITU-T R.G.711, codec interval: 20 ms)

N	IEEE 802.11 with iVoIP (ms)	IEEE 802.11 without admission control (ms)
2	35	40
4	36	40
6	36	42
8	42	44
10	42	46
12	47	51
14	51	56
16	52	57
18	55	62
20	58	70
22	65	81
24	75	179
26	104	211
28	115	241
30	124	301

with iVoIP admission control and classic IEEE 802.11 without admission control. The test is also based on scenario 1 except that the maximum number of VoIP flows was reduced. As indicated in Fig. 7, the VoIP capacity is below 30 when the inter-packet interval is 20 ms, therefore, the number of VoIP flows increased up to 30. In general, the delay increases as the number of VoIP flows (N) increases for both iVoIP admission control and IEEE 802.11 without admission control. In the case of iVoIP, the one-way delay remains below 150 ms which is very good. This is as all the incoming flows are rejected when N exceeds 24, which is the VoIP capacity as shown in Fig. 7. In the case of original IEEE 802.11 without admission control, the one-way delay increases dramatically when $N = 22$, which is the VoIP capacity, as indicated in Fig. 7.

5.2 iVoIP-Fairness

This section presents the experimental results based on scenario 2. Specifically, as shown in scenario 1, the VoIP capacity predicted by *iVoIP-Admission Control* module is 24 when codec interval equals 20 ms. Therefore, the number of VoIP flows is increased up to 24 in scenario 2.

In order to evaluate the performance of *iVoIP-Fairness* module, downlink and uplink fairness was measured using the Jain's fairness index [49] for all QoS parameters considered.

Let Q_D^i $(i = 1, 2, \ldots, M)$ and Q_U^j $(j = 1, 2, \ldots, N)$ represent the QoS parameters (throughput, delay, packet loss rate) of the ith downlink flow and jth uplink flow. The Jain's fairness index in terms of QoS of the downlink and uplink traffic is given in Eq. (22), where separate parameters $FI_{down/up}$ are computed for throughput, delay, and packet loss rate, respectively.

$$FI_{down/up} = \frac{\left(\sum_{i=1}^{M} Q_D^i + \sum_{j=1}^{N} Q_U^j\right)^2}{(M+N)\left(\sum_{i=1}^{M} (Q_D^i)^2 + \sum_{j=1}^{N} (Q_U^j)^2\right)} \quad (22)$$

Figures 9, 10 and 11 present the Jain's fairness index in terms of QoS parameters, delay, throughput and loss, respectively. Table 6 shows the experimental results regarding with the three figures. The results from iVoIP are compared with that of the original IEEE 802.11 protocol and Dynamic-CW. The experimental results show that the fairness level for the three schemes decreases along with increasing number of wireless stations. Moreover, the following conclusions are made: (1) the average fairness levels achieved by iVoIP are 25.8, 16, and 18.9 % higher than 802.11, in terms of delay, throughput, and loss, respectively; (2) the average fairness levels achieved by iVoIP are 21.5, 8.5, and 10.5 % higher than Dynamic-CW, in terms of delay, throughput, and loss, respectively. It is clear that, the most significantly improved QoS parameter is delay, which is critical for VoIP services.

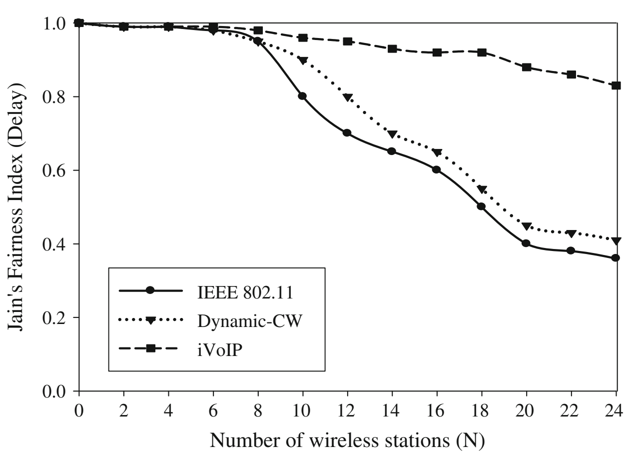


Fig. 9 Jain's fairness index in terms of delay

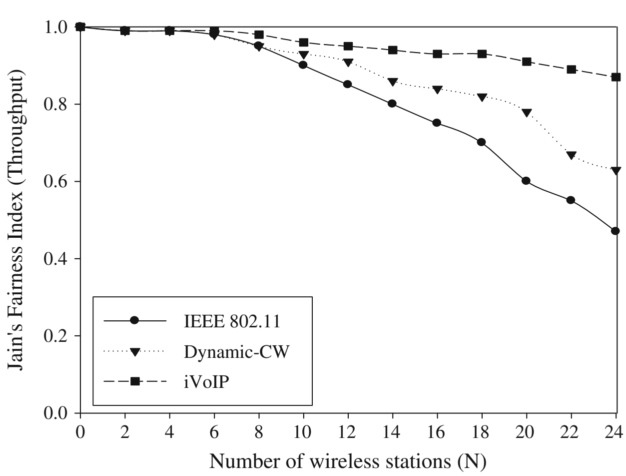


Fig. 10 Jain's fairness index in terms of throughput

5.3 VoIP quality

This section studies the effect of the proposed iVoIP admission control mechanism and fairness scheme on VoIP quality. Scenario 3 was used for the test.

ITU-T E-Model [50] is developed to evaluate the voice applications in heterogeneous circuit/packet-switched networks. A basic result of the E-Model is R-factor, which evaluates voice quality by taking into account both physical equipment impairments and perceptual effects to the equipment impairment. R-factor ranges from the worst case of 0 to the best case of 100 and has been related to Mean Opinion Score (MOS). The computation of the R-factor is simplified according to [51] and is given in Eq. (23). I_d represents the impairment caused by mouth-to-ear delay including codec delay, network delay and playout delay. I_{ef} is associated with losses due to codecs and network. Note that, Eq. (23) does not imply that I_d and I_{ef} are unrelated only that their impacts on the impairments are separable.

$$R = 94.2 - I_d - I_{ef} \quad (23)$$

Since I_d and I_{ef} are difficult to obtain in real time, R-factor is simplified as functions of delay and loss measurements only [51] for the case of G.711 codec, as shown in Eq. (24).

$$R = 94.2 - 0.24 \times d - 0.11 \times (d - 177.3) \times H \times (d - 177.3) - 30 \times \ln(1 + 15 \times e) \quad (24)$$

Parameters d and e refer to the one-way delay (in milliseconds) and the total loss probability (e ranges from 0

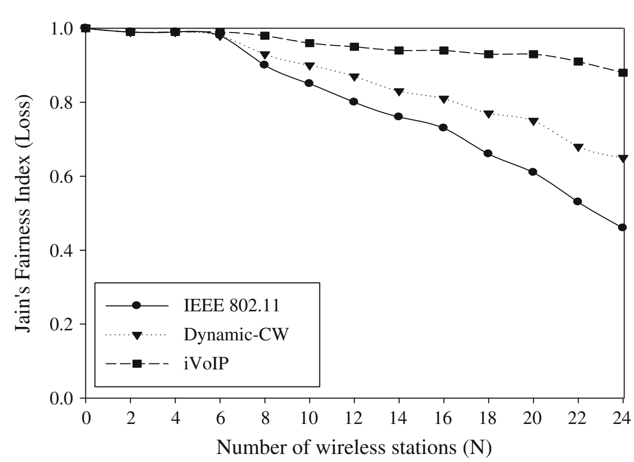


Fig. 11 Jain's fairness index in terms of packet loss rate

Table 6 Jain's fairness index achieved by 802.11, Dynamic-CW, and iVoIP in terms of delay, throughput, and loss

N	Delay			Throughput			Loss		
	iVoIP	Dynamic-CW	IEEE 802.11	iVoIP	Dynamic-CW	IEEE 802.11	iVoIP	Dynamic-CW	IEEE 802.11
2	0.99	0.99	0.99	0.99	0.99	0.99	0.99	0.99	0.99
4	0.99	0.99	0.99	0.99	0.99	0.99	0.99	0.99	0.99
6	0.99	0.98	0.98	0.99	0.98	0.98	0.99	0.98	0.98
8	0.98	0.95	0.95	0.98	0.95	0.95	0.98	0.93	0.90
10	0.96	0.90	0.80	0.96	0.93	0.90	0.96	0.90	0.85
12	0.95	0.80	0.70	0.95	0.91	0.85	0.95	0.87	0.80
14	0.93	0.70	0.65	0.94	0.86	0.80	0.94	0.83	0.76
16	0.92	0.65	0.60	0.93	0.84	0.75	0.94	0.81	0.73
18	0.92	0.55	0.50	0.93	0.82	0.70	0.93	0.77	0.66
20	0.88	0.45	0.40	0.91	0.78	0.60	0.93	0.75	0.61
22	0.86	0.43	0.38	0.89	0.67	0.55	0.91	0.68	0.53
24	0.83	0.41	0.36	0.87	0.63	0.47	0.88	0.65	0.46

to 1), respectively. $H(x)$ is an indicator function: $H(x) = 0$ when $x < 0$ and $H(x) = 1$ when $x \geq 0$. Equations (25) and (26) [51] show the formulas for the computation of both d and e.

$$d = d_{codec} + d_{playout} + d_{network} \tag{25}$$

$$e = e_{network} + (1 - e_{network}) \times e_{playout} \tag{26}$$

d_{codec} is the codec delay which equals 20 ms for ITU-R.G.711 codec. $d_{playout}$ is the playout delay caused at decoder's buffer and is set to 60 ms by default. $e_{playout}$ is the loss probability due to overflow or underflow at decoder's buffer and is set to 0.005 by default. Parameters $d_{network}$ and $e_{network}$ represent transport level delay and loss probability, respectively, which are measured in real time. Figure 12 and Table 7 show that iVoIP outperforms both IEEE 802.11 and Dynamic-CW in terms of R-factor and mean opinion score (MOS) values between downlink and uplink flows. The mapping between R-factor and MOS is given by Eq. (27) [50].

$$MOS = 1 + 0.035 \times R + R \times (R - 60) \times (100 - R) \times 7.10^{-6} \tag{27}$$

When the number of wireless stations (N) exceeds the normal VoIP capacity (i.e. $N = 22$), R-factor values achieved by downlink and uplink flows for both 802.11 and Dynamic-CW decrease dramatically. For instance, when $N = 30$, R-factor values of downlink and uplink for 802.11 is reduced by 40.5 and 35.8 %, respectively, in comparison with the case when $N = 22$. Whereas for iVoIP, there is no significant decrease in R-factor due to the admission control mechanism adopted. For instance, the number of VoIP flows will be rejected if N exceeds 24. Additionally, iVoIP provides the lowest differences in R-factor values between downlink and uplink flows, in comparison with both 802.11 and Dynamic-CW. For instance, when $N = 22$, the differences in R-factor values between downlink and uplink flows for 802.11 and Dynamic-CW are 96.7 and 83.3 % higher than that of iVoIP. This can be explained by two reasons: (1) According to Eq. (21), R-factor mainly depends on network delay and network loss; (2) iVoIP provides QoS-based fair channel access between downlink and uplink flows in terms of throughput, network delay and network loss. The contention window size at AP is adapted in order to provide fair throughput, delay, and loss between downlink and uplink flows. Therefore, low differences in R-factor values between downlink and uplink flows can be guaranteed.

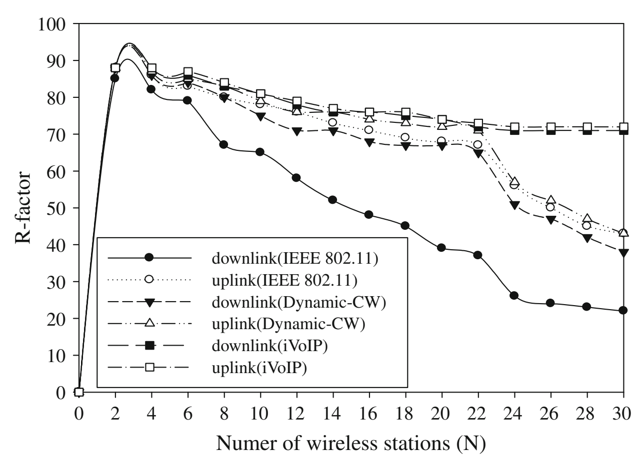


Fig. 12 R-factor values of downlink and uplink flows in 802.11, Dynamic-CW and iVoIP

Table 7 R-factor and MOS values of downlink and uplink flows in 802.11, Dynamic-CW, and iVoIP

N	iVoIP				Dynamic-CW				IEEE 802.11			
	Downlink		Uplink		Downlink		Uplink		Downlink		Uplink	
	R	MOS	R	MOS	R	MOS	R	MOS	R	MOS	R	MOS
2	88	4	88	4	88	4	88	4	85	4	88	4
4	88	4	88	4	86	4	87	4	82	4	86	4
6	86	4	87	4	84	4	85	4	79	3	83	4
8	83	4	84	4	80	4	83	4	67	2	80	4
10	81	4	81	4	75	3	79	3	65	2	78	3
12	78	3	79	3	71	3	76	3	58	1	76	3
14	76	3	77	3	71	3	76	3	52	1	73	3
16	76	3	76	3	68	2	74	3	48	1	71	3
18	75	3	76	3	67	2	73	3	45	1	69	2
20	74	3	74	3	67	2	72	3	39	1	68	2
22	72	3	73	3	65	2	71	3	37	1	67	2
24	71	3	72	3	51	1	57	1	26	1	56	1
26	71	3	72	3	47	1	52	1	24	1	50	1
28	71	3	72	3	42	1	47	1	23	1	45	1
30	71	3	72	3	38	1	43	1	22	1	43	1

6 Conclusions

This paper proposes an intelligent bandwidth management scheme for VoIP service (iVoIP) that provides both admission control and fair channel access. iVoIP includes a novel bandwidth estimation algorithm to provide quality-aware admission control and a new CW adaptation scheme which supports fair channel access between downlink and uplink VoIP flows. Comparison-based results from both simulation and real life tests show how iVoIP performs very well and outperforms state-of-the-art solutions. In particular, the following conclusions have been reached. (1) iVoIP provides good VoIP capacity estimation in IEEE 802.11 networks, i.e. 24 ITU-T G.711 VoIP flows can be supported at high QoS-level in 802.11b network; (2) the average fairness levels achieved by iVoIP are 25.8, 16, and 18.9 % higher than 802.11, in terms of delay, throughput, and loss, respectively; (3) the average fairness levels achieved by iVoIP are 21.5, 8.5, and 10.5 % higher than Dynamic-CW, in terms of delay, throughput, and loss, respectively. In conclusion, in comparison with IEEE 802.11 and Dynamic-CW, iVoIP admission control mechanism provides better VoIP quality and iVoIP fairness scheme achieves lower VoIP quality differences between downlink and uplink flows.

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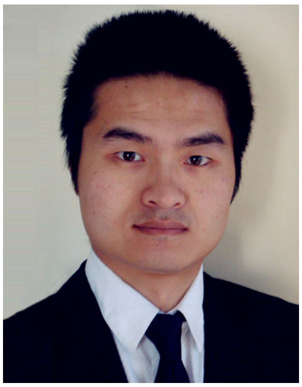

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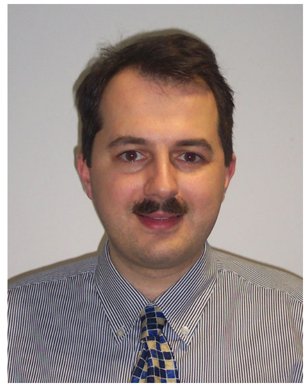

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