# PERFORMANCE OF VOIP OVER IEEE 802.11G DSSS-OFDM MODE WITH IEEE 802.11E QOS SUPPORT

Gráinne Hanley, Seán Murphy, Liam Murphy
*Department of Computer Science*
*University College Dublin, Belfield, Dublin 4*

Keywords: WLAN, IEEE 802.11e, QoS, Voice over IP, Medium Access Control.

Abstract: This paper examines, via simulation, the performance of an 802.11e MAC over an 802.11g PHY operating in DSSS-OFDM mode. The DSSS-OFDM scheme provides data rates of up to 54Mb/s as well as interoperability with 802.11b nodes. Due to the widespread use of 802.11b nodes, such interoperability is an important consideration. This paper involves a study of the number of simultaneous bidirectional G.711 VoIP calls that can be supported by such a WLAN. The results show that this mode of operation introduces a very significant overhead. The actual number of calls that can be carried is limited to 12 when using the 24Mb/s data rate and 13 when using either the 36Mb/s or 54Mb/s rates. These results demonstrate the well-known disparity between uplink and downlink performance, with the downlink imposing the limit on the number of calls that can be carried by the system in the cases studied. The results also show that when when a significant amount of lower priority traffic is introduced into the system, it can have a significant impact on VoIP call capacity despite the use of 802.11e.

## 1 INTRODUCTION

WLAN support for QoS is one issue that is receiving considerable interest at present. The IEEE 802.11e standard, which provides standardised QoS support, is in the final stages of development. Frustrated by the sluggish pace of development of the standard, the Wi-Fi Alliance, an industry forum, is attempting to promote development of WLAN QoS by offering certification for a subset of the standard's functionality which it calls *Wi-Fi Multimedia* (WMM). A significant number of vendors are already certified.

While vendors are beginning to ship systems with QoS support, there is still considerable debate within the community regarding how best to operate these systems. The standards have been written in such a way as to offer much flexibility to enable vendors to differentiate their product offerings. This flexibility, coupled with the complexity of the system, means that large differences in system performance are possible. At present, it is not clearly known how best to configure such systems. Hence, there is a need to understand how these systems behave for different parameter sets in different configurations.

One application driving the development of WLAN QoS support is *Voice over IP* (VoIP). Many enterprises and WLAN operators are very interested in providing *VoIP over WLAN* (VoIPoW) and indeed some vendors have product offerings which can address this need. However, such product offerings are typically proprietary. Standardised solutions offer many known benefits and for this reason, it is important to determine the performance of VoIPoW in a standardised 802.11e/802.11g setting.

As there are currently a very large number of 802.11b network interfaces in existence, it is interesting to see how backwards compatibility issues affect system performance for 802.11e/802.11g systems. 802.11g provides two modulation schemes with backwards compatibility mechanisms. The first is called ERP-OFDM, while the other is called DSSS-OFDM. ERP-OFDM is based on transmitting *Request To Send /Clear To Send* (RTS/CTS) signalling at a low rate such that legacy nodes know when the medium is unavailable. DSSS-OFDM is based on transmitting packet preamble and header information at an 802.11b compliant rate.

Some research has taken place to determine the performance of the ERP-OFDM scheme in certain circumstances. This paper aims to determine the perfor-



Hanley G., Murphy S. and Murphy L. (2005).
PERFORMANCE OF VOIP OVER IEEE 802.11G DSSS-OFDM MODE WITH IEEE 802.11E QOS SUPPORT.
In *Proceedings of the Second International Conference on e-Business and Telecommunication Networks*, pages 20-27
DOI: 10.5220/0001416200200027

mance of the latter – the DSSS-OFDM scheme – in a VoIP context.

This paper is structured as follows. Section II briefly outlines the operation of the 802.11e MAC and the 802.11g PHY, and Section III outlines some of the published work related to this study. In section IV, the simulation setup is described in detail. This is followed by a description of the simulations performed and a discussion of the results obtained in section V. The paper is concluded in section VI.

# 2 IEEE 802.11

The simulations studied here use an 802.11e MAC layer in association with an 802.11g PHY layer; the operation of these layers is briefly outlined below.

## 2.1 IEEE 802.11e

IEEE 802.11e (IEEE, 2005) provides for centralised and distributed QoS support at the *Medium Access Control*(MAC) layer. As there is considerably more complexity in the centralised scheme, the focus here is on the distributed approach, the *Enhanced Distributed Channel Access* (EDCA).

EDCA provides for 4 different so-called *Access Categories* (ACs). Different priorities for the 4 ACs are realised through 4 coupled *Carrier Sense Multiple Access with Collision Avoidance* (CSMA/CA) mechanisms. These mechanisms are coupled as they contend for access to the same medium, but they differ in that they are parameterised differently. More specifically, the parameters for the higher priority traffic are chosen to enable it to obtain access to the medium more quickly than the lower priority traffic. Contention between different priorities in a single station is resolved such that the higher priority traffic gains access to the medium, while the lower priority enters a backoff state.

Two of the key parameters which control how the different access categories obtain access to the medium are the *Contention Window* (CW) sizes and the *Arbitration Inter-Frame Space* (AIFS). The former controls how much random waiting, or backoff, delay should be introduced for each AC to avoid collision and the latter controls how long each AC waits after a transmission has terminated before attempting to access the medium.

The 4 ACs have been labelled *Voice* (VO), *Video* (VI), *Best Effort* (BE) and *Background* (BG). Parameter settings for each of the categories are shown in Table 1.

The values of the 802.11g parameters are shown to be relative to aCWmin and aCWmax. As stipulated in the standard, an aCWmin of 31 and an aCWmax of 1023 were used, to maintain compatibility with 802.11b systems carrying VoIP traffic.

Table 1: Default IEEE 802.11e EDCA Parameter Set

| AC | CWmin | CWmax | AIFSN |
|---|---|---|---|
| VO | (aCWmin+1)/4-1 | (aCWmin+1)/2-1 | 2 |
| VI | (aCWmin+1)/2-1 | aCWmin | 2 |
| BE | aCWmin | aCWmax | 3 |
| BG | aCWmin | aCWmax | 7 |

## 2.2 IEEE 802.11g

The 802.11g *Physical layer* (PHY) (IEEE, 2003) enhancement outlines 4 modulation schemes. Two of which are mandatory – ERP-OFDM and ERP-CCK/DSSS – and two of which are optional – ERP-PBSS and DSSS-OFDM. Of the four schemes, only ERP-OFDM and DSSS-OFDM provide data rates of up to 54Mb/s using OFDM modulation schemes, while also providing explicit support for interoperating with 802.11b nodes. Such support is necessary as 802.11b nodes are not able to detect or understand OFDM modulated signals.

The ERP-OFDM scheme is a variant of the 802.11a PHY scheme modified for use in the 2.4 GHz band. The DSSS-OFDM scheme is a hybrid modulation scheme that combines a DSSS preamble and header with an OFDM payload transmission.

ERP-OFDM uses the RTS/CTS mechanism to provide 802.11b interoperability. This mechanism gives the 802.11g nodes time to freely transmit at the higher rates. In the DSSS-OFDM scheme, all packet headers and preambles are transmitted at the lower 1Mb/s rate using 802.11b compliant DSSS modulation. Thus, 802.11b nodes know how the medium is being used even if they cannot detect OFDM payload transmission.

# 3 RELATED WORK

While WLAN performance analysis is currently a very active research area, little has been published on VoIP performance for 802.11e/802.11g systems.

Although Mangold et al. (Mangold et al., 2003) analysed the performance of the 802.11e standard, their study was performed in relation to 802.11e/802.11a. Therefore, backward compatibility with 802.11b was not a concern.

Choi and Pavon (Choi and Pavon, 2003) discussed backward compatibility of the 802.11g ERP-OFDM scheme with regards to the 802.11b standard. Their results showed that the ERP-OFDM RTS/CTS protection mechanism introduces a lot of overhead, thus greatly reducing the performance of the network when compared with a system containing only 802.11g nodes. Their results showed that the transmission time of a packet on 802.11g with a long preamble RTS/CTS exchange, was about double that of 802.11g with RTS/CTS disabled. However, Bianchi (Bianchi, 2000) showed that without the RTS/CTS mechanism, the performance of the 802.11b DCF scheme was highly dependent on the number of nodes in the system and the size of the minimum CW. Given that the DSSS-OFDM scheme does not require RTS/CTS signalling, it is interesting to see the performance levels that can be obtained using this scheme.

Garg and Kappes (Garg and Kappes, 2003) performed an analytical analysis of VoIP capacity for the 802.11b and 802.11a schemes. They developed a formula which can be used to calculate the VoIP call capacity of a WLAN network under certain assumptions. However, their paper did not discuss the use of the 802.11e QoS mechanism. In addition, although quite useful, their formula does not hold, unless there is only VoIP traffic on the system. Also, their formula is based on the assumption that there are only ever two active senders in the network, whereby the AP and one wireless node always have a packet to send. For these reasons, it was unclear that their work could accurately predict the levels of performance that can be attained by VoIP traffic on an 802.11e/g system.

An important phenomenon in these systems is the disparity between the system performance in the uplink and downlink; this has been reported in previous work by both Grilo and Nunes (Grilo and Nunes, 2002) and Casetti and Chiasserini (Casetti and Chiasserini, 2004). Both papers were however related to the 802.11e MAC layer over the 802.11b PHY layer. The same difference is apparent throughout the results in this paper but is further examined and discussed in relation to the 802.11g PHY scenario.

# 4 THE SIMULATION SCENARIO

In order to determine the performance of VoIP in an 802.11e/802.11g system, a series of simulations were performed. In these simulations, the wireless nodes were arranged with an AP in the network which formed a connection between every 802.11e node in the wireless domain and a single node in the wired domain (see Fig.1). This AP was connected to the wired network by a high capacity link with negligible delay, which was dimensioned such that it could easily carry all the traffic and hence no loss occurred on this link. All wireless nodes were within radio range

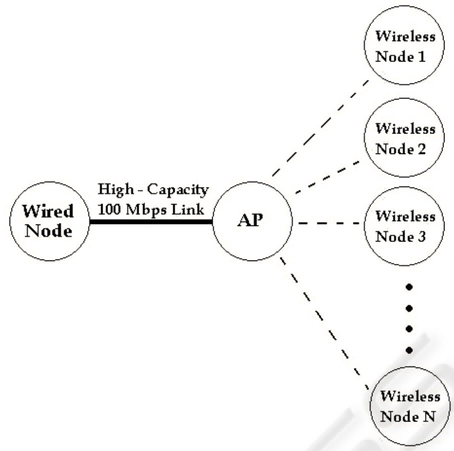


Figure 1: Network Topology

of each other, thus ensuring that no issues arose relating to hidden/exposed station problems. In addition, all nodes were situated sufficiently close together so that they were able to transmit at the highest data rate supported by 802.11g. Static routing [1] was used, so as to ensure realistic routing of the wireless traffic.

The simulations were configured such that each node used bidirectional CBR traffic sources, so as to model VoIP traffic. This was represented as *Constant Bit-Rate* (CBR) traffic, transmitted using *User Datagram Protocol* (UDP). In a similar manner to that of Yu, Choi and Lee (Yu et al., 2004), these sources were parameterised to model G.711 voice at 64kb/s with 20ms payload. The G.711 scheme was chosen as it is still commonly used, due to its simplicity, despite the availability of schemes with better compression. As in (Yu et al., 2004) a VoIP data payload size of 160 bytes was generated every 20ms, to which the 20 byte IP header, the 12 byte RTP header and the 8 byte UDP header were added. In order to avoid issues with traffic synchronisation, a low level of random noise was introduced into the packet generation process. This resulted in a source which generated a 200 byte packet approximately every 20ms, resulting in 80kb/s of traffic in total per node. This VoIP traffic was always transmitted at the highest priority level of 802.11e in accordance with the standard.

In some of the later experiments, an additional bidirectional low priority, BE traffic source was introduced for each of the bidirectional VoIP sources in the simulation. Like the VoIP traffic, this was also parameterised as CBR traffic over UDP but at a rate of 250kb/s and with a 1500 byte data payload. The aim was to see how a VoIP application would perform if there was a heavy traffic load at the lower priority.

[1] The *No Ad-Hoc Routing* (NOAH) patch for ns-2.26 was used - http://icapeople.epfl.ch/widmer/uwb/ns-2/noah.

Table 2: Extended Rate IEEE 802.11g PHY Characteristics

| Characteristic | Value |
|---|---|
| SlotTime | 20$\mu$s(long), 9$\mu$s(short) |
| SIFSTime | 10$\mu$s |
| CCATime | $<$15$\mu$s(long), $<$4$\mu$s(short) |
| aCWmin(0) | 31 |
| aCWmax | 1023 |
| Supported Rates | 1, 2, 5.5, 6, 9, 11,12, 18, 24, 36, 48, and 54Mb/s |
| Mandatory Rates | 1, 2, 5.5, 11, 6, 12, and 24Mb/s |

The IEEE 802.11g parameters (see Table 2) for the simulations using the DSSS-OFDM modulation scheme were chosen to allow backward compatibility with the IEEE 802.11b PHY. Hence, the long PLCP preamble, long slot time, and long *Clear Channel Assessment* (CCA) time were used. In accordance with the standard for DSSS-OFDM, the long PLCP preamble and long PLCP header were transmitted at 1Mb/s.

For this study, a simulation model of the IEEE 802.11e with the EDCA mechanism, developed by the TKN group in Berlin, for the Network Simulator package NS was used (Wietholter and Hoene, 2003).

Since the primary focus of this study was the VoIP traffic, a 50 packet queue limit was chosen for every node in the system, as there is no advantage in queuing VoIP packets for extended periods because they are delay limited. Furthermore, studies have indicated (Yu et al., 2004) that there is a minimal performance difference between a 50 and 100 packet queue.

Each simulation was run three times and the results were averaged over these three runs. All simulations were run for 250 seconds of simulation time, and the maximum mandatory 802.11g data rate of 24Mb/s as well as the optional higher 36Mb/s and 54Mb/s rates were studied.

# 5 RESULTS

These results are an assessment of the performance of the DSSS-OFDM scheme with regards to VoIP capacity. This analysis is firstly performed at three different data rates but in the absence of any other traffic. Then the effect of the addition of a large amount of BE (see Table 1) traffic is examined at the same three data rates.

In these experiments, loss and delay measurements were taken at the UDP layer of the protocol stack. The downlink delays are the average packet delays from the originating wired node, to the receiving wireless node. Similarly, the uplink delays are the average packet delays from the originating wireless node, to the receiving wired node.

The loss examined here represents packets which were sent by the UDP transport layer of the transmitting node, but which were never received by the UDP transport layer of the receiving node. These loss rates therefore represent the percentage of packets which are dropped due to collisions on the medium or *Interface Queue* (IFQ) overflow.

There are MAC level retransmissions of all collided packets, but those packets which exceed the retransmission threshold without being successfully received are considered as lost. For these simulations, the Short Retry Limit was set to 7 in accordance with what is recommended by 802.11e.

VoIP requires certain quality levels: ETSI studies (ETSI, 2002) indicate that a packet loss rate of 5% is at the upper bound for acceptable voice quality. Also, for this study, WLAN delays of greater than 50ms were considered to be unacceptably high.

## 5.1 Analysis of VoIP Traffic in an 802.11e/802.11g Network with DSSS-OFDM Modulation

This set of simulations was performed using the DSSS-OFDM parameters, and in the absence of any traffic other than the VoIP traffic being analysed. The simulations were run at the maximum mandatory data rate of 24Mb/s, as well as at the optional higher rates of 36Mb/s and 54Mb/s. The results show a comparison of the uplink and downlink results for the end to end delays and loss rates, the average contention window sizes used for the backoff calculation and the percentage occupancy of the IFQ.

### 5.1.1 End-to-End Delay and Loss Rates

At first, the similarity in the results for each of the three data rates, 24, 36 and 54Mb/s, seems surprising. In fact, results show that increasing the rate at which the actual data is sent has quite a small impact on overall performance levels. This can be attributed to the fact that in the DSSS-OFDM scheme, the longer PLCP data is used and is sent at a slow 1Mb/s rate for backwards compatibility. Therefore a large overhead is introduced for each packet transmission. The negative performance effects of this large overhead dominate the overall system performance and to a great degree mask the positive impact of an increased data rate.

Results show that increasing the maximum data rate of the system from 24Mb/s, to 36Mb/s, and then to 54Mb/s, does lower the delay experienced by packets (see Fig.2). However, if the average delay for 15 bidirectional VoIP calls is examined for each of the three data rates, the average downlink end-to-end delay at 24Mb/s is 111ms, at 36Mb/s is 96ms, and at 54Mb/s is 86ms. Although notable, these improvements in delay at the higher data rates are quite low given the greatly increased data rates.

The known disparity between uplink and downlink performance is quite visible in these results: the delay difference between uplink and downlink is especially apparent when there are 13, 14 and 15 calls on the network. For this region there are quite large downlink delays in comparison to minimal uplink delays. This can be attributed to the saturation of the downlink occurring prior to that of the uplink. However, it can be seen that after 15 calls the uplink delays also begin to climb. By the 17 call point, there is more uniformity to the delays as both uplink and downlink are saturated and hence delay levels begin to converge.

It is clear from the results that in the 24Mb/s scenario a large increase in loss is experienced by the downlink when there are only 13 bidirectional calls on the system. In fact, at this stage loss rates are already in excess of the 5% threshold (see Fig.3). Results show that the 24Mb/s network can support only 12 bidirectional calls, and both the 36Mb/s and 54Mb/s systems can support only 13 bidirectional calls before loss rates on the downlink reach 6% or higher.

In this scenario, the majority of the downlink loss was due to queue overflow, whereas the majority of the uplink loss was caused by retransmission failure. Loss at the AP can be explained by noting that the AP has to handle downlink traffic for *all of the wireless nodes*; if the medium is congested, then it may not obtain sufficient access to the medium for all this downlink traffic, its queue builds up and it suffers packet loss. In contrast, the individual wireless stations have much lower traffic levels and so the queues at the wireless stations rarely contain many packets. Hence, wireless stations do not encounter IFQ overflow. However, it seems that the uplink does experience a higher probability of collision than the downlink. These higher levels of collisions and retransmissions on the uplink occasionally lead to a packet being dropped as it has exceeded its maximum number of retransmission attempts without being successfully received.

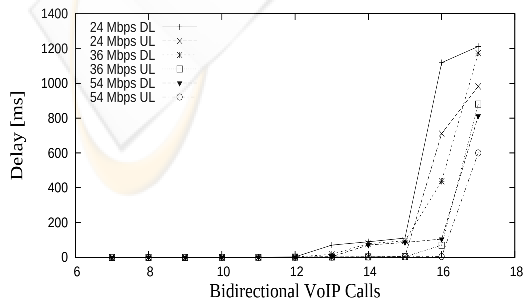


Figure 2: End-to-End Delays With Only VoIP Traffic

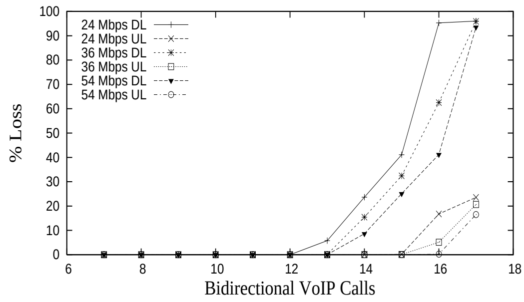


Figure 3: Loss Rates With Only VoIP Traffic

These results indicate that the use of parameters which enable backward compatibility lead to a serious reduction in the performance of VoIP applications. In previous experiments that were performed which focussed on the ERP-OFDM scheme without backward compatibility mechanisms, it was found that at 24Mb/s an ERP-OFDM 802.11g WLAN system could support approximately 48 bidirectional G.711 VoIP calls. In fact, in this scenario, the 802.11g DSSS-OFDM modulation scheme performs little better than a basic 802.11b in terms of VoIP call capacity: Garg and Kappes (Garg and Kappes, 2003) have shown that the 20ms G.711 capacity for an 11Mb/s 802.11b system is 12 calls.

### 5.1.2 Contention Window Size

For each of these simulations the *Contention Window* (CW) size that was used for any transmission or retransmission was recorded. The following graphs (see Fig.4) show the average size of the high priority CW for the scenarios which involved between 7 and 17 bidirectional VoIP calls. Surprisingly, it was noted that the average size of CW used by the AP was always smaller than that of the corresponding wireless node.

This difference is as a result of the frequency with which the AP attempts to access the medium as opposed to the frequency with which the wireless nodes attempt to gain access.

Due to the greater level of traffic at the AP, its IFQ fills much more quickly than that at any individual wireless node. This means that frequently when the AP is attempting to gain access it is not in direct contention with any of the wireless nodes. Due to this lower contention level, the AP has a lower probability of colliding with another packet and thus a lower probability of having to retransmit with an increased CW.

On the other hand, the wireless node will almost always be in direct contention with the AP when it is attempting to transmit, hence, it will have a higher potential for colliding with a packet sent by the AP.

#### 5.1.3 IFQ Occupancy

The results show a breakdown of the IFQ occupancy for high priority traffic as a percentage of the simulation time. These statistics are very informative as to the point at which high priority traffic begins to encounter notable delays in accessing the medium (see Fig.5). If queue occupancy is at a high level for a large amount of the simulation time, it is an indication that there is a lot of congestion on the network, and hence packets are encountering difficulty in accessing the medium.

The results from the 24Mb/s simulations (see Fig.5 show that the queue occupancy rates of the wireless nodes are lower than that of the AP. This behaviour is not unexpected as the amount of traffic that a single wireless node has to handle is a lot lower than the traffic levels at the AP.

### 5.2 Analysis of VoIP Traffic in the presence of Best Effort traffic

This section describes experiments that were performed to determine the impact of high levels of Best Effort CBR traffic on the system, while using DSSS-OFDM modulation.

#### 5.2.1 End-to-End Delay and Loss Rates

If the end-to-end delays for these bidirectional VoIP calls are compared with the results obtained in the absence of BE traffic (see Fig.2 and 6), a significant increase in delay is observed. In addition, for this network setup, results show a definite increase in the losses at all three data rates, compared to the voice only traffic scenario (see Fig.3 and 7). It was found that in the absence of BE traffic, the 24Mb/s network could support 12 bidirectional calls, and that both the 36Mb/s and the 54Mb/s networks could support 13 bidirectional calls before delays and loss rates on the downlink reach unacceptable levels.

Based on the loss and delay results in the presence of BE traffic, it can be seen that the 24Mb/s network can support only 9 bidirectional calls, and both the 36Mb/s and 54Mb/s systems can support only 10 bidirectional calls. After this point, loss rates on the downlink become excessive and the VoIP traffic encounters a large increase in end-to-end delay. This indicates that with this amount of additional traffic in this network scenario, the number of supportable bidirectional VoIP calls is reduced by 3, which is a significant decrease in terms of VoIP call capacity.

Due to the nature of the 802.11e mechanism such a large increase in delay seems somewhat surprising as 802.11e was designed to facilitate medium access to higher priority traffic largely at the expense of lower priority traffic. However, here it is demonstrated that this is not always the case.

The influence of background traffic is dependent upon many factors. In general, extra traffic will increase the risk of collision, which will lead to an increased number of retransmissions as well as a greater number of lost packets.

In this case, the lower priority, BE traffic packets have a data payload of 1500 bytes as opposed to the 160 bytes in the VoIP packets, and hence will occupy the medium for longer periods. Such longer transmission times will further increase the risk of collision with VoIP traffic. In fact, the transmission of a BE packet at the 24Mb/s data rate will occupy the medium for approximately 1ms as opposed to only 590$\mu$s for a VoIP packet. Therefore, during this additional 410$\mu$s delay more VoIP packets will have arrived in the AP queue. This will cause the queue at the AP to build much more quickly than when no BE traffic is present. This will lead to increased queuing delays, as well as an increase in back-off delays, for the VoIP traffic. Plus, as queue occupancy levels increase, it could eventually lead to packets being dropped due to queue overflow.

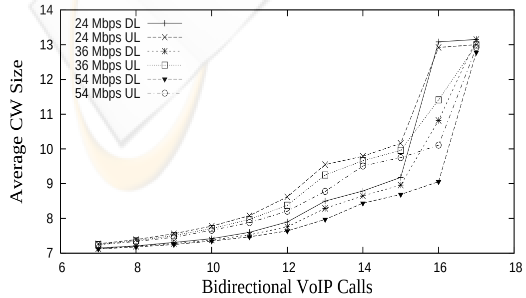


Figure 4: Contention Window Sizes With Only VoIP Traffic

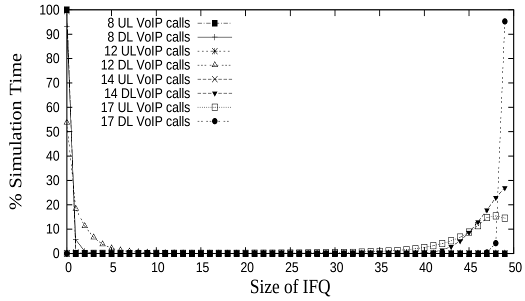


Figure 5: 24Mb/s AP and Wireless Node IFQ Sizes With Only VoIP Traffic

### 5.2.2 Packets Received After Retransmission

In order to further understand the increases in loss and delay, the levels of packets received after retransmission was examined.

In this scenario, results show that the additional BE traffic leads to a large increase in retransmissions at both the AP and at the wireless nodes. This can be clearly seen in Fig.8.

The medium capacity used when packets collide and are retransmitted is a highly inefficient use of resources. This inefficient use of resources leads to less available resources in the system, which results in lower system throughput and lower service rates on each queue in the system. This, in turn, causes queue occupancy to increase.

Due to the high traffic levels at the AP, an increase in retransmission levels has a large impact on the downlink delay in particular. Combined with its high packet arrival rate this can often also lead to queue overflow at the AP, which results in increased loss levels on the downlink.

### 5.2.3 Average Contention Window Sizes

The results again show that the AP has a consistently lower CW than the wireless nodes (see Fig.9). However, the results can be seen to show three phases in the relationship between the contention window sizes of the AP and the individual wireless nodes. Firstly, from 7 to 10 VoIP calls, then from the 10 to 15 VoIP calls and finally as the level of voice traffic reaches 16 to 17 VoIP calls, a third distinct region can be seen.

If these results are considered in association with the IFQ occupancy rates then an association between both sets of results can be seen. In the first phase, both the AP and wireless nodes have an empty IFQ for the majority of the time, that is, at least more than 50% of the time the queues are empty. However in the second phase, it can be noted from the results that the AP IFQ has reached a point whereby the IFQ contains packets for most of the simulation time. Although at this stage the wireless node queues still remain empty for most of the time. The final section corresponds to a stage when both the AP and wireless nodes queues are full for the majority of the simulation time and the CW size can be seen to increase more rapidly until the CW size of the uplink and downlink ultimately converge.

The difference in the uplink and downlink CW sizes is a reflection of the differing levels of retransmissions on the uplink and downlink since retransmissions are sent with an increased CW; as the uplink has more retransmissions, its CW is higher. Also, in this scenario, there is correlation between the IFQ occupancy and the mean CW size. As the mean CW size

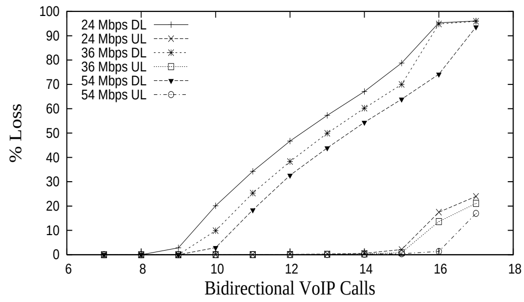


Figure 7: Loss Rates With Best Effort Traffic

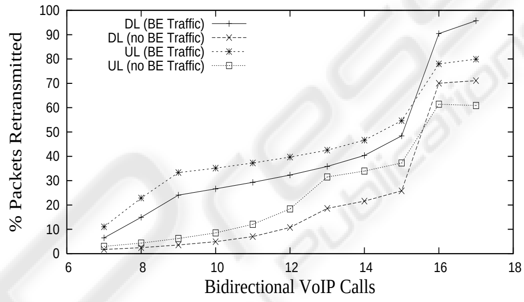


Figure 8: VoIP Packets Received After Retransmission With Best Effort Traffic at 24Mb/s

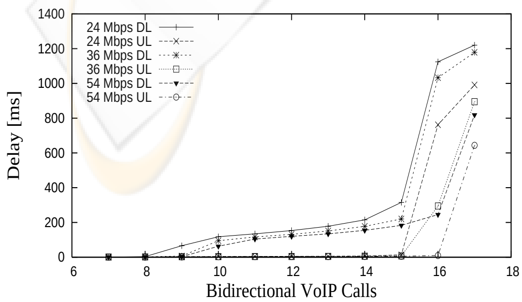


Figure 6: End-to-End Delays With Best Effort Traffic

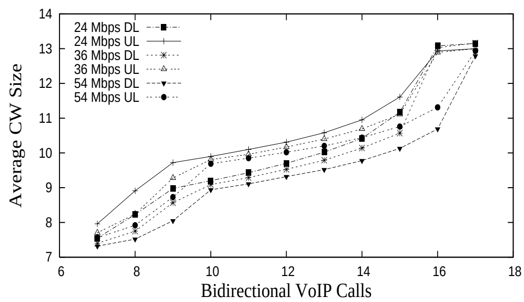


Figure 9: Contention Window Sizes With Best Effort Traffic

increases (due to retransmissions on the medium), there are increased delays and lower medium throughput resulting in increasing occupancy levels at the IFQ.

## 6 CONCLUSION

The VoIP capacity of the DSSS-OFDM modulation scheme when using parameters which provided backward compatibility, showed that the 24Mb/s network can support only 12 bidirectional calls, and both the 36Mb/s and 54Mb/s systems can support only 13 bidirectional calls. Above these capacities, loss rates on the downlink reach an unacceptably high level. The results also clearly show that, as expected, the downlink quality begins to suffer long before the performance of the uplink begins to deteriorate.

Interestingly, the average size of CW used by the AP was generally smaller than that of the wireless nodes. This was due to the different medium access requirements of the AP and the individual wireless nodes. The AP almost always had a packet waiting in its IFQ for transmission and so due to the lower levels of traffic being sent by each wireless node often it was not in direct contention with another node. In contrast, the wireless nodes were frequently in direct contention with the AP, hence the wireless nodes often had a higher probability of collision. Since collided packets are retransmitted with an increased CW size, this led to a difference between the average CW size at the AP and at the wireless nodes.

Surprisingly, results also showed that the addition of BE traffic leads to an increase in end-to-end delay, loss rates and IFQ occupancy for the VoIP traffic. Additional traffic leads to an increased risk of collision for the VoIP packet, which was further increased by the large size of the BE load. It was found that the delays resulting from the additional collisions and retransmissions cause a large increase in queuing delay and so decreased the packet service rate, particularly at the AP.

In fact, it was found that, under such conditions, the 24Mb/s network can support only 9 bidirectional calls, and both the 36 and 54Mb/s systems can support only 10 bidirectional calls before loss rates on the downlink are in excess of 10%. These unanticipated results indicate that for this network scenario and with this amount of additional traffic, the number of supportable bidirectional VoIP calls is reduced by 3 calls.

Future work involves further investigation of backward compatibility mechanisms for 802.11g, further investigation into ways to equalise the division of resources between the uplink and the downlink and investigating the optimum transmission opportunity sizes for this scenario.

## ACKNOWLEDGEMENTS

The support of the Informatics Research initiative of Enterprise Ireland is gratefully acknowledged.

## REFERENCES

Bianchi, G. (2000). “Performance Analysis Of The IEEE 802.11 Distributed Coordination Function”. *IEEE Journal on Selected Areas in Communications*, 18(3):535–547.

Casetti, C. and Chiasserini, C. F. (2004). “Improving Fairness And Throughput For Voice Traffic in 802.11e EDCA”. In *Proc. IEEE PIMRC*, Barcelona, Spain.

Choi, S. and Pavon, J. P. (2003). “802.11g CP: A Solution For IEEE 802.11g And 802.11b Inter-Working”. In *Proc. IEEE VTC*, Jeju, Korea.

ETSI (2002). *“ETSI TR 101 329-6 V2.1.1, TIPHON Release 3; End-to-End Quality of Service in TIPHON Systems; Part 6”*.

Garg, S. and Kappes, M. (2003). “Can I Add a VoIP Call?”. In *Proc. IEEE ICC*, Anchorage, Alaska, USA.

Grilo, A. and Nunes, M. (2002). “Performance Evaluation of IEEE 802.11e”. In *Proc. IEEE PIMRC*, Coimbra, Portugal.

IEEE (2003). *“Wireless LAN Medium Access Control and Physical Layer Specifications Amendment 4: Further Higher Data Rate Extension in the 2.4 GHz Band, IEEE Std 802.11g-2003”*.

IEEE (2005). *“Local and Metropolitan Area Networks - Specific Requirements Part 11: Wireless LAN Medium Access Control and Physical Layer specifications: Amendment: Medium Access Control (MAC) Quality of Service Enhancements - P802.11e/D13.0”*.

Mangold, S., Choi, S., Hiertz, G., Klein, O., and Walke, B. (2003). “Analysis of IEEE 802.11e for QoS Support in Wireless LANs”. *IEEE Wireless Communications*, 10(6):40–50.

Wietholter, S. and Hoene, C. (2003). “Design And Verification of an IEEE 802.11e EDCF Simulation Model in ns-2.26: TKN-03-019”. Technical report, TKN, Berlin.

Yu, J., Choi, S., and Lee, J. (2004). “Enhancement of VoIP Over IEEE 802.11 WLAN Via Dual Queue Strategy”. In *Proc. IEEE ICC*, Paris, France.