Enhancement of IEEE 802.11 in Handling Multiple Broadcasting Audio Data in Wireless Ad-Hoc Networks

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Audio networking is a growing field, introducing new and exiting possibilities in the professional audio industry [1]; but it also drastically changes the way audio systems will be designed, built, and used. Today's networks have enough bandwidth to transport hundreds of high quality audio channels, replacing hundreds of kilograms of cabling in conventional analog audio systems [2]. Currently there are many systems on the market that distribute audio over Ethernet but the majority of sound engineers are not using them yet. There are mainly two reasons that audio networks are not as popular as expected. First, many of the systems are based on a proprietary implementation that does not allow interoperability between different vendors. Second, wired networks, like the conventional analog audio networks, also need a cable installation. It is therefore understood that the development of a wireless digital audio distribution system would be a significant contribution in this sector. IEEE 802.11 (Wi-Fi) as the primary wireless technology in computer networking has made wireless networks widely available and inexpensive. With its most recent amendments, as well as the use of the 5-GHz ISM band, it can facilitate many high quality audio channels. However, the use of this technology has not been the choice for the audio industry so far. It is obvious that a sequence of problems related with the nature of this technology impede the use of Wi-Fi in professional live sound and studio applications [3]. Apart from the well-known drawbacks of interference and security, encountered in all wireless data transmission systems, the way that Wi-Fi arbitrates the wireless channel access is what causes the majority of the problems. In this paper we highlight the drawbacks of the IEEE 802.11 MAC algorithm in handling multiple stations broadcasting of audio data. We simulate a live audio data wireless network and test the limits of the protocol for this type of traffic. Moreover, we modify the 802.11 MAC algorithm to address the above problems. We test the amended protocol using simulation and analyze the results. We also give the directions for the future research in order for this widely accepted technology to be used in the professional audio industry.

0 INTRODUCTION

In order to use IEEE 802.11 as the wireless networking platform in a live music system we must first understand the topology, the characteristics, and the demands of such a system [4]. Generally, there are two possible cases that describe what a wireless station (WSTA) represents. In the first case, every single audio source (microphone, guitar, keyboard, etc.), and also every audio receiver (mixing console, audio processor, stage monitor, etc.), can be considered as an independent WSTA. In the second case, every individual musician is considered as a WSTA that transmits the audio data that he produces and receives the audio data he needs for his monitoring. The disadvantage of the first case is that it demands a wide number of WSATs that are only either transmitters or receivers, while the drawbacks of the second case is that many musicians produce more than one audio data stream (drum sets, stereo keyboards, etc.). This must be taken into consideration by manufacturers for potential future implementation.

Whatever the choice on how to assign WSTAs to musicians, all produced audio signals in the network must be available to all possible users. The idea of collecting all audio signals in a master WSTA, which can be a console that also works as an Access Point (AP), group them and retransmit them unicast or multicast, as we do in conventional analog systems, has serious drawbacks. Because of the half-duplex nature of the RF medium, only one station can use it at a time. All other WSTAs that have audio data to send must wait. That means that it is significantly important for the transmission to be completed as fast as possible in order to minimize delay, which in the case of live audio, is crucial. Using an AP to retransmit audio data in the same Basic Service Set (BSS), to one or multiple users, we actually occupy the wireless medium twice for the same data. Multicast transmission cannot provide any guaranteed delivery. Unicast transmission is able to provide guaranteed delivery using positive acknowledgment (AKC), but again, there is no time to retransmit lost audio data packets in a live music networking environment.

A good practice is to broadcast audio data in the whole BSS. This practice gives all WSTA the ability to receive concurrently all produced audio data and then decide at the upper layers which of those are going to be used.

What emerges from the above is an ad-hoc wireless network without AP and essentially without hidden nodes where all WSTAs are broadcasting audio data. The network size is similar to a live music stage or a studio. The only effective way to improve the performance of such a network is to achieve maximum throughput keeping the delay at acceptable levels. Defining throughput as the average rate of successful packet delivery over the network and assuming an ideal, error free channel, we can see that this is affected mainly by two parameters. The first one is the dropped packets after the number of attempted retransmissions reaches the maximum retry count, as defined by the protocol. The second parameter is the collisions that can happen in the wireless medium. The IEEE 802.11 standard implements a Carrier Sense Multiple Access mechanism with Collision Avoidance (CSMA/CA) that provides various techniques to avoid collisions. The random backoff mechanism and the RTS/CTS and CTS-to-Self protection mechanism provide significant protection from collisions but they cannot completely eliminate them. In addition, RTS/CTS cannot be implemented in broadcasting while CTS-to-Self has significant drawbacks that will be analyzed further in the following sections.

In this paper we are implementing two novel modifications of the 802.11 MAC in order to increase performance in a multi-broadcasting environment. First, we are modifying the use CTS-to-Self in order to distribute the time the network will be occupied for each transmission. This reduces the number of backoff counts and solves the problem of dropped packets due to exceeding the retransmission attempts limit.

Second, we are implementing an Exclusive Backoff Number Allocation with fairness (EBNA) algorithm that increases the contention window (CW) size using a linear increase method. Thus, we decrease the probability of collision while we keep delay at an acceptable level.

The rest of the paper is organized as follows. Section 1 reviews the 802.11 MAC process and analyzes the drawbacks of random backoff algorithm in the case of multiple broadcasting. Section 2 evaluates the ability of IEEE 802.11 to handle multiple broadcasting audio data. In Section 3, the proposed modifications of the protocol are analyzed, while in Section 4 the simulation procedure is described and the results are presented and analyzed. Finally, Section 5 gives the concluding remarks.

1 ANALYSIS AND DRAWBACKS OF IEEE 802.11 MEDIUM ACCESS MECHANISM

In this section we review the procedures of the IEEE 802.11 MAC algorithm and especially the random backoff process, which performs poorly in a multi-broadcasting live audio network.

1.1 General Description

The IEEE 802.11 MAC is mainly designed for wireless unicast communication and for unlimited number of users in the network [11]. In Distributed Coordination Function (DCF), which is its primary medium arbitration method, random backoff in conjunction with virtual and physical carrier sense provides a level of protection from collisions. The 802.11 2007 standard provides an additional protection mechanism using RTS/CTS or CTS-to-Self control frames. The last one is mainly used for Network Allocation Vector (NAV) distribution in mixed-mode environments where different 802.11 technologies coexist. Although RTS/CTS is used to address the hidden node problem, CTS-to-Self is used strictly as a protection mechanism for mixed-mode networks using data rates and modulation methods that legacy 802.11 technologies can understand. NAV is distributed by setting the duration field of the control frame with the time in microseconds required in order for the two parties to complete transmission including ACK. It is clear, however, that there is no MAC-Level recovery mechanism in broadcasting [5] and, as mentioned in the introductory section, that could not be a choice in the case of live music audio networking. In live music audio networking, the focus must be on preventing the loss of packets and the collisions instead of recovery and retransmission. NAV distribution is possible in broadcasting, only in mixed mode networks, by using the CTS-to-Self control frame [4]. CTS-to-Self is a standard CTS frame transmitted with a destination address of the transmitting station. The transmitting station cannot hear its own transmission in a half-duplex medium, but all nearby WSTAs are alerted that a broadcast frame is pending and they can also update their NAVs with the value included in the duration field of the CTS-to-Self frame. As mentioned above, the use of CTS-to-Self is strictly limited in mixed-mode environments and it is using lower data rates that reduce throughput and increase delay. The possibility of modifying the 802.11 MAC to use CTS-to-Self as a main NAV distribution method, also using high data rates, will significantly contribute to the performance of the protocol especially in broadcasting. However, as shown in Section 1.2, the use of CTS-to-Self alone cannot eliminate the collision's occurrence, which is caused by the drawbacks of 802.11 MAC random backoff mechanisms. This mechanism significantly contributes in collision avoidance but cannot totally eliminate them, especially when the

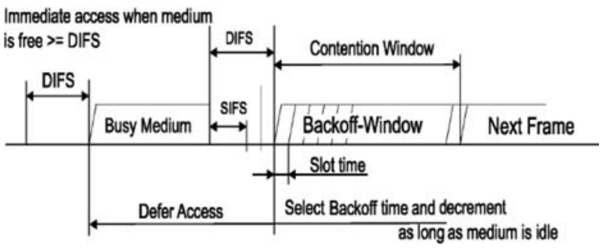

Fig. 1. IEEE 802.11 basic access method

number of WSTAs increases and there is also continuous data production, as in live music performance. In heavy data loads, there is a high likelihood that two or more WSTAs will choose the same backoff value. In this case the collision cannot be avoided regardless of the use of CTS-to-Self. For this reason an alternative EBNA algorithm can be used to overcome the random backoff algorithm drawbacks in the case where multiple broadcasting is taking place.

1.2 Analysis of the IEEE 802.11 MAC Algorithm

IEEE 802.11 MAC Layer is the lowest part of the Link Layer and it is placed between the Physical (PHY) and the Logical Link Control (LLC) sub-layer. MAC architecture is based on two basic coordination functions, Point Coordination Function (PCF) and Distributed Coordination Function (PCF). PCF is a contention free access method that provides polling intervals to allow uncontended transmission opportunities (TXOP) for participating WSTAs. This function is not used here, first because it demands the use of an AP and second, because the manufacturers never applied it to their devices. The optional Hybrid Coordination Function (HCF) that is introduced to support QoS is also outside of our interest. In a wireless audio network all data are time sensitive and belong to the same category (i.e., audio), so there is no chance to divide them in different access categories and give them different priorities. In this study the fundamental DCF contention-based access mechanism is used.

DCF's timing diagram is illustrated in Fig. 1 and its function is described as follows. A WSTA with a packet to transmit waits for the channel to become idle. When an idle period equal to DCF Inter-Frame Space (DIFS) is detected, it generates an initial backoff time value. This value indicates the period that the WSTA has to additionally defer before transmitting. The random backoff process is the most important mechanism used in IEEE 802.11 CSMA/CA to prevent collisions. CW increases exponentially for every retransmission (unique per station). Under low utilization, stations are not forced to wait very long before transmitting their frame. If the utilization of the network is high, the protocol holds stations back for longer periods of time to avoid the probability of multiple stations transmitting at the same time. When we refer to Contention-Based access, random backoff is actually the primary mechanism for contention. This value is extracted from the following formula:

$$Backoff\_Time = INT(CW\,x\,Random\,(0,\,1))\,x\,aSlotTime \qquad (1)$$

Random (0, 1) is a pseudo-random number between 0 and 1 drawn from a uniform distribution. CW is an integer within the range of values CWmin and CWmax. CWvalues $= 2^x$-1 (x starts from an integer defined by the station and goes up to 10). For example, for x = 4, $CW_4 =$ 2^4-1 = 15, $CW_5 = 31$, $CW_6 = 63$ ….. $CW_{10} = 1023$. The *aSlotTime* duration is the value of the correspondingly named PHY characteristics. The backoff timer is decremented with one slot as long as the channel is idle. When a transmission is detected, the backoff timer freezes and starts to decrease again when the channel is sensed as being idle for a DIFS. When the timer reaches zero the data packet is finally transmitted.

1.3 Drawbacks of Random Backoff in Wireless Broadcasting

There is plenty of research on the Reliable Broadcasting over wireless ad-hoc networks and many protocols have been proposed [6] [7] [8]. These protocols can be divided into four main categories according to the methods they use.

1) **Simple Flooding Methods**: Requires each node to retransmit all packets;
2) **Probability Based Methods**: Use some basic understanding of the network topology in order to assign a probability to a node to rebroadcast;
3) **Area Based Methods**: Rebroadcasting is based on the possible additional area that will be covered;
4) **Neighbor Knowledge Methods**: Maintain a state of neighbors, obtained by "Hello" messages. This stage is used in the decision to retransmit.

All the above methods require a sort of retransmission that is unsuitable for live audio networking. Reliability in audio broadcasting is reduced by the drawbacks of random backoff process, which cause channel access delay and collisions no matter the available bandwidth of the wireless technology that is used.

The IEEE 802.11 standard defines that the CW size exponentially increases for each retransmission attempt of the same packet. However, as there is no retransmission in broadcasting, the CW size always holds the CWmin value. Under high utilization due to increasing number of WSTA and/or high data production, CWmin appears to be extremely small. In this case we are facing two major problems. The first one is that it is possible for a WSTA that just completed a transmission and has a new packet to send, to choose zero as its initial backoff time and start transmitting immediately after a DIFS. As we can see from Eq. (1), backoff time is a random outcome based on a uniform distribution but its range increases proportionally with the size of CW. This consecutive transmission will give other WSTAs no chance to backoff. This problem is referred as the backoff counter consecutive freeze process (CFP), and was extensively analyzed by Xianmin Ma and Xianbo Chen [9]. They show, with their model and simulations, that the solution would be the ability to increase CW in broadcasting. The

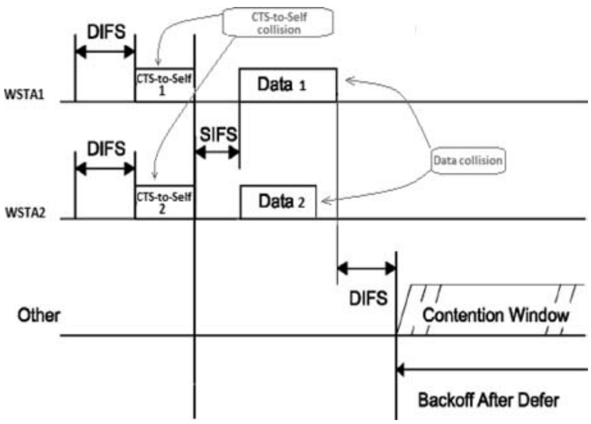

Fig. 2. CTS-to-Self and data Collisions

second and most significant problem in the case of wireless audio broadcasting is that there is a high likelihood for two or more WSTAs to choose concurrently equal backoff values. It is easy to understand that when we have fifty or more WSTAs producing continuous data and they are performing the backoff process using a CW = 15 (as in 802.11g and 802.11n) this is highly possible. In this case a collision is occurring and a data packet is lost as there is no recovery mechanism and no time for retransmission.

The use of CTS-to-Self does not make any improvement in this case as collided CTS-to-Self messages cannot be identified. As we can see in Fig. 2, two WSTAs with the same backoff time (WSTA1,2) will transmit a CTS-to-Self simultaneously. None of the two will identify the collision because CTS-toSelf1_time = CTS-to-Self2_time. After that, they will both sense the medium as idle for an SIFS and they will transmit their data causing another collision. In addition, NAV1 and NAV2 cannot be distributed to the nearby WSTAs.

2 EVALUATION OF IEEE 802.11 IN HANDLING MULTIPLE BROADCASTING AUDIO DATA

In this section an evaluation of the IEEE 802.11 protocol in handling multiple broadcasting audio data in a wireless ad-hoc network is performed. Initially, the model that was created to emulate the live music audio traffic is described. Then, the characteristics of the simulation and measuring parameters are analyzed and the graphic results are presented.

2.1 Definition of Audio Data Traffic

In order to emulate a professional live audio environment the data traffic generator in each WSTA is adjusted to generate a data payload based on a 16-bit/44.1-KHz sampling rate (PCM, no compression). That gives a bit rate of 0.67 Mbps, which gives consequently for a packet size of 2200 bytes, an interarrival time of 24.3 msec.

It is significantly important in this emulation to understand the form of live music audio. The form of the audio produced from a member of a live music band that performs a piece is totally different from Mixed Audio. Live music performances mainly produce a monophonic audio that is not continuous but contains gaps that sometimes are considerably long. Fig. 3 illustrates the waveform of a mixed song and its corresponding vocal track.

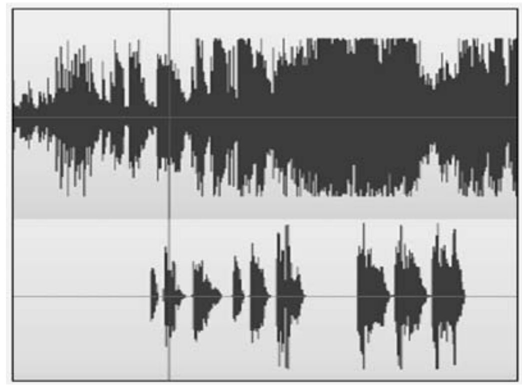

Fig. 3. Voice Track and Mixed Audio of a song

Table 1. Traffic Generation Parameters

Start Time	Normal Distribution (1, 0.01)
On-State	0.25 sec
Off-State	0.25 sec
Interarrival Time	Constant Distribution (24.3 msec)
Packet Size	2200 bytes

To emulate this specific way that live music audio is produced, we adjusted the traffic generator of each station to perform based on a tempo of "120." This practically creates a "sound" every 0.5 sec and if we assume that this sound lasts for half of this 0.5 sec period, we finally get an On-State = 0.25 sec and Off-State = 0.25 sec. We also define the start time based on a normal distribution with mean outcome 1 and variation 10 msec, to emulate the stochastic nature of musical performance.

The resulting load transmitted by each WSTA is not constant because of the normal distribution set in the start time attribute. It is approximately 383 Kbps, which is 48 Kbps higher than the generated load due to MAC overhead. All data traffic generation parameters are listed in Table 1. It is important to note here that the above model is set to create a realistic case study for comparison and cannot emulate completely the stochastic way that music is produced.

2.2 Simulation Characteristics

The network simulation platform used in this study is OPNET Modeler 17.1. The simulation is based on IEEE 802.11g PHY, with a bit rate of 54 Mbps.

The topology is based on an ad-hoc network in a single BSS, with the WSTAs located randomly in a 30×40 m surface. The number of WSTAs is gradually increased to 70 during the study. The simulation duration is 2 min. Three separate simulations have been conducted where all stations were relocated and also a different seed number has been set during the simulation execution. The presented results are the average values, in those cases where significant differences occurred.

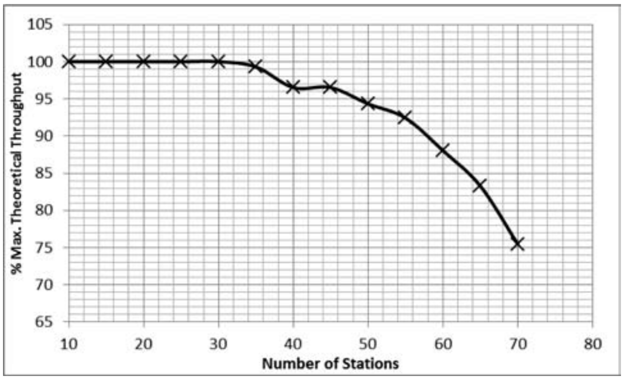

Fig. 4. Throughput of IEEE 802.11 for Broadcasting

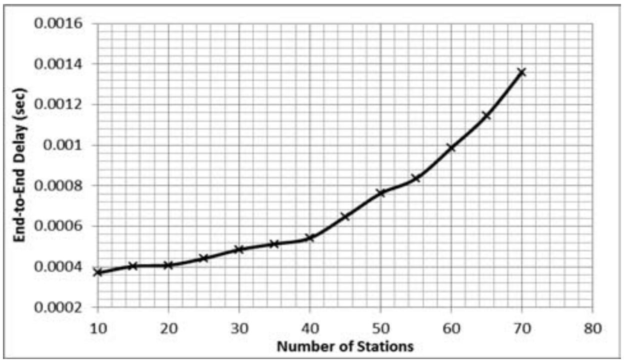

Fig. 5. Overall End-to-End Delay of IEEE 802.11 for Broadcasting

The statistics collected during simulations are *Global Throughput* and *End-to-End Delay*. Global Throughput represents the total number of bits (in bits/sec) forwarded from wireless LAN layers to higher layers in all WLAN nodes of the network. End-to-End Delay represents the total mean delay of all the packets received by the wireless LAN MACs of all WLAN nodes in the network and forwarded to the higher layer. This delay includes medium access delay at the source MAC [10].

For every increase in the number of stations, the maximum theoretical throughput is calculated and then compared with the one resulting from the simulation. In the resulting graph we see the percentage of the maximum theoretical throughput achieved by the simulated network for each number of WSTAs. In broadcasting, the maximum theoretical throughput is given by the equation:

$$Throughput_{max(theoretical)} = (n-1)\sum_{i=1}^{n} Ai \quad bits/sec \tag{2}$$

where n the Number of WSTA and Ai the Data load produced by the individual WSTA. In the case that all stations create the same data payload (A), the maximum theoretical throughput is given by:

$$Throughput_{max\,(theoretical)} = n[(n-1)A]bits/sec \tag{3}$$

2.3 Simulation Results

The resulting graphs for *Throughput* and *End-to-End Delay* are shown in Figs. 4 and 5 respectively.

2.4 Performance Analysis

As shown in the graphs from Figs. 4 and 5, the standard performs well in networks where the number of WSTAs remains small. For the number of WSTAs equal to or less than 30, the measured throughput appears to be the 100% of the theoretical maximum, which means zero losses. At the same time the delay remains significantly low. As expected, by increasing the number of WSTAs the performance becomes poor, and in the case of 70 WSTAs the total loss reaches the level of 25% of the total transmitted data. Respectively, the End-to-End delay also increases. It is important to note that although the offered network bandwidth is much higher than that required by the produced data, it cannot override the inherent problem of the standard caused by the random backoff process. As it is analyzed in Section 1.3, the inability of the backoff process to alter the CW size in broadcasting combined with the lack of a NAV distribution mechanism reduces dramatically the protocol performance in an audio data multiple broadcasting environment.

In the next section a modification of the 802.11 MAC is proposed in order to address the above problems.

3 THE MODIFIED 802.11 MAC

As mentioned earlier in this paper, in order to override the inability of the 802.11 protocol in handling multiple broadcasting audio data, a modified MAC mechanism is proposed. The amendments focus on two main areas: the NAV distribution and the random backoff algorithm.

3.1 NAV Distribution Using CTS-to-Self

NAV distribution is normally used in broadcasting only in cases where legacy technologies coexist with an ERP (802.11g) or HT (802.11n) physical (mixed-mode networks). It is achieved by sending a CTS-to-Self control frame in appropriate (usually lower) data rate and modulation that all WSTAs can understand. CTS-to-Self frame contains in its duration field the time that all non-transmitting WSTAs must defer before trying to access the medium.

In our modified MAC we proposed the use of CTS-to-Self control prior to every data transmission. In order to decrease delay the MAC process is reprogrammed to transmit this control frame using the operational data rate used for data transmission.

3.2 Exclusive Backoff Number Allocation Algorithm (EBNA)

In classic 802.11 broadcasting, the CW size remains constant, getting its minimum possible size. In order to prevent WSTAs from choosing similar backoff numbers, which leads to a collision regardless of the use of NAV distribution mechanisms, a simple EBNA algorithm is implemented. This algorithm linearly increases the size of CW according to the number of WSTAs in the network. It is also designed to maintain fairness while allocating exclusive backoff values for each transmission attempt. In order to do this, the algorithm needs two external variables, the

```
No_of_STAs=10  %total no of stations
STID=2          % current station
Group=rand (1,2)
if (group=1)
   backoff_slots=STID
else
   backoff_slots=No_of_STAs*2-STID+1
end if
```

Fig. 6. Exclusive Backoff Number Allocation algorithm pseudocode

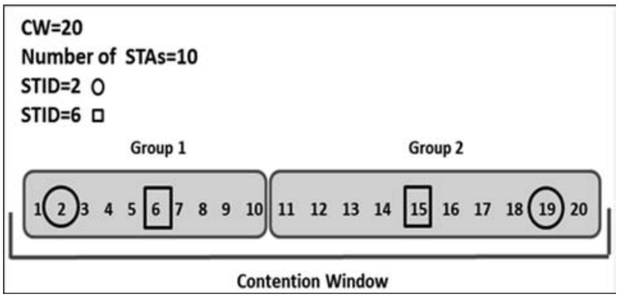

Fig. 7. EBNA example

total number of WSTAs in the BSS *(No_of_STAs)* and the Station ID *(STID)* that every STA obtains upon joining the network. The CW is always given by:

$$CW = No\_of\_STAs * 2 \tag{4}$$

The algorithm divides the CW in two equal groups. Values in the groups are allocated as follow:

$$\begin{aligned} group1 &\leq No\_of\_STAs/2 \\ group2 &> No\_of\_STAs/2 \end{aligned} \tag{5}$$

For each transmission attempt a random value between 1 and 2 is generated in order to select one of the two groups. If group 1 is selected the algorithm allocates to the STA a backoff value equal to its STID; in other cases the value given by the algorithm is a projection of the STID value to group 2 and it is given by the formula:

$$Backoff\_slots = [(No\_of\_STAs * 2) - STID] + 1 \tag{6}$$

For a network with 10 STAs the station with STID = 2 will take randomly one of the backoff values 2 or 19, while a station with STID = 6 will take the backoff value 6 or 15 (Fig. 7). The pseudo-code describing the above process is illustrated in Fig. 6.

4 IMPLEMENTATION OF THE MODIFIED MAC MECHANISM

In this section the implementation of the modified MAC mechanism is presented. We give details about the modification in an OPNET wireless model in order to function according to the new rules we set in the MAC mechanism. We also describe the simulation procedure and illustrate and analyze the results.

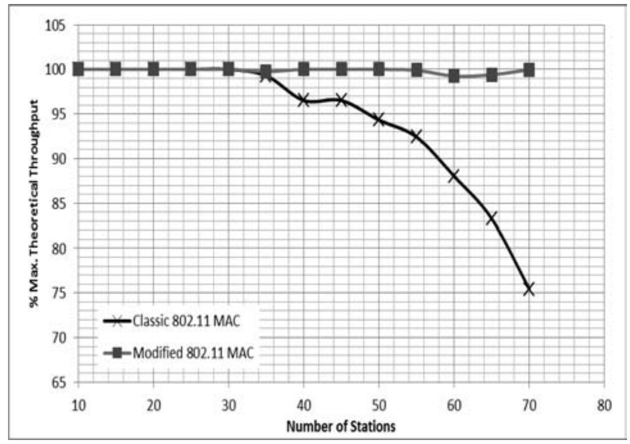

Fig. 8. Throughput, Modified vs. Classic 802.11 MAC

4.1 OPNET Wireless Model Modification

OPNET modeller is a powerful simulation tool that allows users to have full access to the executed code and gives the ability to create and modify complex communication protocols. It has its own C++ library and it uses state machines to design and implement processes. For our implementation the *OPNET wireless station node model* is used. First, a custom *Station ID* attribute is created in order to allow users to give each station a unique ID number. Then, the "*wlan_dispatch*" process and the "*wlan_mac*" child process are modified. In the wlan_mac process (BKOFF_NEED state), the backoff_slots allocation algorithm has been changed according to our proposed EBNA algorithm. In "*Function Block*," where all functions of this process are defined (including CTS-to-Send control frame creation and transmission), the necessary modification has been done in the code in order for the CTS-to-Self frame to be transmitted in all cases prior to every data frame transmission. Finally a new local statistic was created in order to be used in tracing our custom variables *(STID, group and backoff_slots)* during the simulation.

4.2 Simulation of the Modified Model

The topologies of the network, the wireless technology, and the transmission rates as well as the protocol of this simulation are similar to those presented in Section 2.2. The duration of the simulation is again 2 minutes and the population of the WSTAs reaches also the number of 70. Multiple simulation seeds have also been set in order to ensure the reliability of the results.

In the following graphs we present the results in Global Throughput and End-to-End delay obtained from the modified IEEE 802.11 MAC simulation. For comparison we also include the results from the classic 802.11 MAC simulation as we presented them in Section 2.3.

Fig. 8 shows the percent of the maximum theoretical throughput while Fig. 9 shows the average End-to-End delay of the entire network.

In Fig. 10, the fairness of the EBNA algorithm is illustrated.

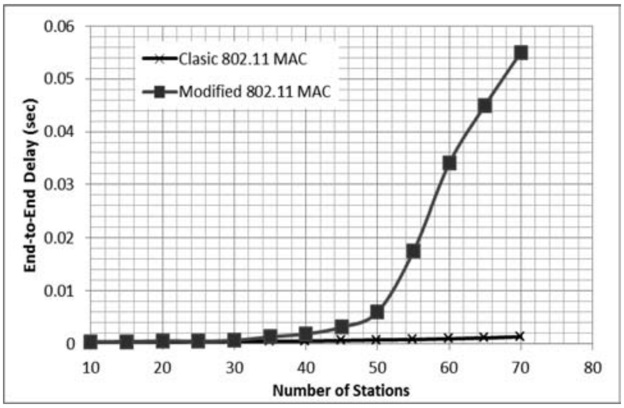

Fig. 9. End-to-End Delay for Modified and Classic 802.11 MAC

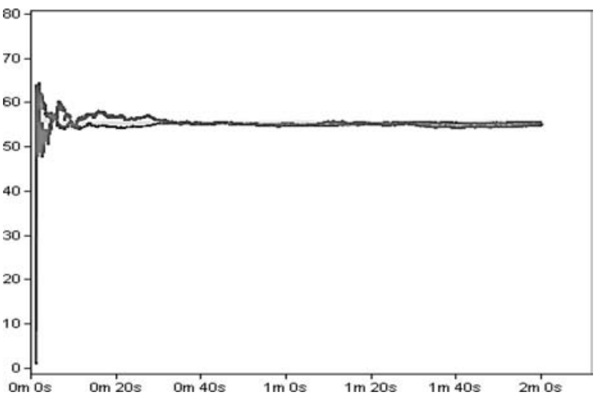

Fig. 10. Mean average of Backoff Values

4.3 Modified 802.11 MAC Performance Analysis

The graph in Fig. 8 shows the percent of the Maximum Theoretical Throughput achieved from each simulation. It actually illustrates the amount of data that is successfully delivered during the simulation. This measurement refers only to data delivery and does not contain control frames. As we can see from the results, using the modified 802.11 MAC model the system manages to deliver almost 100% of the produced data with near zero losses. It is important to understand that using the EBNA modifications, although the overall load of the network is increased, the performance is also increased. This is not unexpected, as the drawback of 802.11 MAC in handling multiple broadcasting data is not related to the bandwidth.

The graph in Fig. 9 shows the overall End-to-End delay for each simulation and also refers only to the data frame delivery and not to the control frames. As it is expected the End-to-End delay is increased. This happens for two reasons. The first reason is the additional control traffic due to CTS-to-Self transmission. The second reason is the increase of the CW. As we described in Section 3.2 the EBNA algorithm is implementing a linear increase of the contention window. This increase is mainly responsible for the increase of the delay. However, this technique has significant advantages compared to the classic exponential increase of the CW. Exponential increase of CW improves performance but the delay reaches unacceptable levels for real time audio delivery. In this study the proposed EBNA algorithm manages to keep the CW relatively small and the overall delay at acceptable levels.

The graph in Fig. 10 is extracted from the simulation with 55 STAs in the network. According to the EBNA algorithm the CW size for this case is 110. The graph shows for three randomly chosen STAs the mean average of the values that the *Backoff_Slots* variable is taking for each STA. As we can see, all STAs have the same mean average for their backoff slots and also this is located in the middle of the CW. That means that all STAs have equal overall backoff delay during the simulation.

5 CONCLUSIONS

In this paper we examined the ability of the IEEE 802.11 standard to be used as the networking technology in a wireless live audio environment. We first analyzed the standard and highlighted its drawbacks. Thereafter, we evaluated the standard by simulating a realistic wireless audio network setup. The results of this test show that, although there is a wide bandwidth available, the 802.11 MAC by its nature is not able to handle this type of traffic. The problems are mainly in the way that 802.11 MAC handles broadcasting and, more specific, in random backoff algorithm and also in the lack of a NAV distribution mechanism. To address these problems a modified MAC was proposed. In this amendment, the NAV distribution is achieved by using CTS-to-Self control frames that are transmitted with the operational bit rate of the network, prior to each data transmission. In addition an alternative to random backoff Exclusive Backoff Number Allocation (EBNA) algorithm is proposed. This algorithm implements a linear increase of CW according to the number of STAs, while fairly allocating exclusive backoff values to each STA. This modified 802.11 MAC is simulated and the results show that it drastically improves its performance in a multiple broadcasting environment. The system manages to deliver almost 100% of the produced data with near zero losses. As expected the End-to-End delay is increased. This is caused by the additional control traffic due to CTS-to-Self transmission and also due to the increase of the CW. The classic 802.11MAC keeps the CW size at its minimum level, which for ERP and HT technologies is extremely small. That gives an impressively low delay but this is only measured for the traffic that manages to be delivered. When loss becomes high this low delay it is not representative of the actual quality of the network. Using the modified 802.11 MAC proposed in this paper the delay remains at acceptable levels. In addition, the EBNA algorithm maintains fairness during the backoff values allocation process, while keeping the size of the CW at a relatively low level. Future studies should focus on incorporating the EBNA philosophy proposed here, into a traffic adaptive backoff algorithm in order to further improve delay performance.

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