Multicast voice performance within a public safety cell

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ABSTRACT

In public safety communications the first responders are getting directions about the tactical action plan with multicast voice whereas they can report back to the dispatcher with unicast voice. In this paper, the aim is to find the maximum number of voice calls for situation reporting in the presence of multicast voice for tactical coordination. In order to increase the reliability of our analysis we verify our simulator against a test bed prototype consisting of three 802.11 terminals. The simulation study is applied within a mobile cell. The proposed mobility model applies for initial deployment in emergency scenarios. We investigate the statistical properties of the model by simulations.

Keywords

Distributed coordination function, mobility model, multicast voice over internet protocol, simulator verification.

INTRODUCTION

The dearth of communication infrastructure to support public safety finds lucrative prospects in WLAN networks. The public safety activities require rapid deployment, affordable communication infrastructure cost and support for broadband applications. WLAN based networks embody these attributes and they have been already proposed for public safety communications. For instance, standalone mobile ad hoc and mesh networks, integration of WLAN with TETRA and WLAN cell extensions to WiMAX mesh networks are investigated in [1][2][3] respectively.

In order to avoid loss in time and casualities, the emergency response requires reliable communication channels between the command center and the on-field personnel. In this context, group-oriented services have proved to be useful. The group services are used for team coordination while the one-to-one services are utilized for reporting to the command centre. The described use case underlies the need for multicast voice transmitted by the command centre to the on-site users and unicast voice transmitted from the on-site users to the command centre.

The support of real-time multimedia applications over 802.11 networks has been extensively investigated in the literature. The number of full duplex unicast voice connections that a single 802.11 DCF access point (AP) can support is investigated in [4]. Therein the experimental analysis is performed with respect to the voice codec, the length of the audio payload and the channel bit rate. The voice quality is measured w.r.t. the packet loss, the round trip delay and its variation. A similar analysis is executed in [5] where the voice quality is evaluated against the E-model standardized by ITU. In both papers it is concluded that the amount of simultaneous voice calls with acceptable service quality is limited mainly due to the significant overhead introduced at the MAC layer.

The limitations of 802.11 DCF are investigated by executing point to point maximum throughput measurements in [6] [7]. The inefficiency of the MAC layer for small UDP payloads is pointed out. In order to increase the capacity under a single AP, a multiplex-multicast scheme is proposed in [8]. The system architecture requires a voice multiplexer that is responsible for multiplexing the unicast streams into a single voice stream for multicast over the 802.11 users. In this way the MAC layer overhead is significantly reduced. The simulations show that the number of simultaneous voice communications can be almost doubled.

In this paper, we evaluate the performance of downlink multicast voice for team coordination and uplink unicast voice for reporting. The simulation study is repeated for different bit rates of IEEE 802.11b. The performance degradation due to a single uplink video feed for better situation awareness at the command centre is also examined.

Our simulation could be considered reliable enough since the simulator is first verified against an 802.11 prototype of three nodes. Qualnet 4.0 provides the reliable and comprehensive modeling for simulating the large scale wireless networks [9]. The simulation analysis models the initial deployment of the users towards the hotspot. A typical disaster area can be divided into incident, transportation and treatment sub areas. Some mobility models consider the user movement over the complete disaster area [10]. we model the mobility of the rescue team only in the incident sub area. We throw light on the statistical properties of the mobility model by using simulations.

PERFORMANCE METRICS FOR VOICE AND VIDEO TRAFFIC

Recall that the real-time services must be QoS sensitive in order to support the public safety demands. Throughout our analysis the critical performance factor is transmission reliability in terms of packet delivery ratio and delay:

Packet Delivery Ratio (PDR) is defined as the number of data packets received to the application layer of the receiving node divided by the number of packets supposed to be received.

Delay is the time difference between the packet generation time at the source node and the packet reception time at the receiving node. We are interested in unicast flows from the wireless nodes to the AP and multicast flow from the AP to the nodes. Therefore only the one-way delay has been considered.

The ITU G.114 recommends that the one-way delay should be less than 150ms for acceptable conversation quality for most of the user applications [11]. Hence, the local delay within the 802.11 cell should be as small as possible in order to relax the delay constraint across the backbone network.

SIMULATOR VERIFICATION

Before following the common practice of using the simulator we address the reliability concern of simulator performance. For that purpose we verify the simulator against an experimental setup of three terminals. The purpose of the experimental study is two-folded: Firstly, it is to identify the PHY and MAC layer parameters employed by the test bed and use the same parameter values in the simulator. Secondly, it is to investigate the effect of packet collision in the test bed and ensure a similar effect in the simulator. For this purpose we create a single test bed setup and measure the maximum aggregate unicast throughput of two nodes competing for medium access.

Test bed setup

The test bed consists of two PCs and one laptop that emulates the AP. The PCs and the laptop are equipped with LINKSYS Wireless-G 801.11 b/g PCI adaptors and Proxim ORiNOCO Gold 801.11 b/g PCMCIA Atheros chipset respectively. The Atheros chipset offers the flexibility for configuring most of the 802.11 DCF MAC layer parameters. The drivers worked best in Linux environment are RT2500 [12] and MadWifi v0.9.3.3 [13] for LINKSYS and Proxim adaptors respectively.

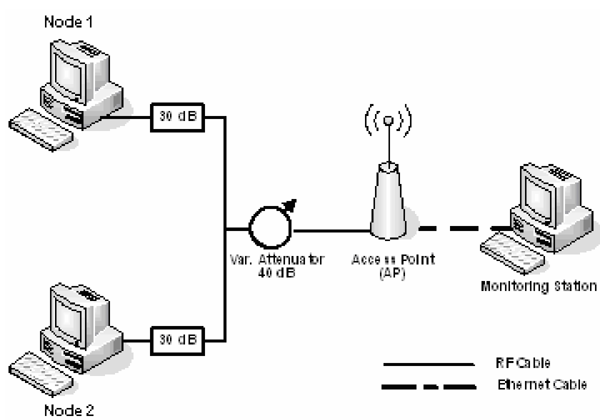

Figure 1 : Test bed setup

Figure 1 shows the physical setup of the test bed. The AP is connected to the monitoring station by 100Mbps 802.3 LAN. The connection quality between the PCs and the AP is excellent. In order to eliminate the hidden terminal problem we arrange the two PCs to be within the interference range of each other. This means that all the packet drops happen when the two PCs select the same backoff value.

In order to build the ideal propagation environment described above we use RF cables and attenuators. The constant propagation environment is necessary both for the result consistency in repeated experimentation and for the replication of same pathloss values in the simulator. The attenuation between the PCs is 60 dB whereas the attenuation between the PC and the AP is 70 dB. The accurate attenuation values accounting for RF cables, connectors, splitters, fixed and variable attenuators are measured with network analyzer after calibration. The monitoring station depicted in Figure 1 is used for the termination of the two UDP flows.

In order to characterize the PHY and MAC layer parameters we use UDP traffic because of its connectionless nature. For various sizes of UDP payload we generate flooding at both PCs by using the MGEN v.4.0 toolset [14]. Then the TRPR v.2.0b1 is used to extract the necessary statistics including aggregate throughput of the two flows and average delay [14]. The flooding adjusts the packet generation periodicity so that there is always a packet in the transmission queue. Therefore the two PCs will be always competing for channel access.

Throughput measurements and simulations

The same scenario is now created in the simulator for the verification of its performance. In particular, the channel model is defined to be a pathloss model where the attenuation values are set equal to the corresponding values used in the test bed. The sensitivity and the transmission power employed by the WLAN cards are imported into the simulator. Note that the manufacturers do not provide enough information regarding the radio properties of the cards. Instead we measure the transmission power and the sensitivity by utilizing spectrum analyzer and according to the procedure described in [15]. As the next step the ACK payload, its transmission bit rate, the beacon interval, the slot size and the PHY preamble type are retrieved in the test bed by using a packet sniffer [16].

So far the consistency of the parameter settings between the simulator and the test bed has been kept. However there are no explicit means to identify the DIFS/SIFS intervals and the average channel idle time in the test bed. A viable solution is to import their standard values in the simulator and check the validity of our assumption by comparing the maximum throughput values between the simulator and the test bed.

Figure 2 presents the comparison of maximum aggregate unicast throughput between the test bed and the simulator for 2Mbps and 11Mbps. The experiments and the simulations were repeated many times for each payload size and it was observed that the two nodes shared the throughput equally. One can observe that the performance results are satisfactorily close to each other. It can be deduced that the consistency of the parameter values between the test bed and the simulator has been maintained. A summary of these values can be found in Table 1.

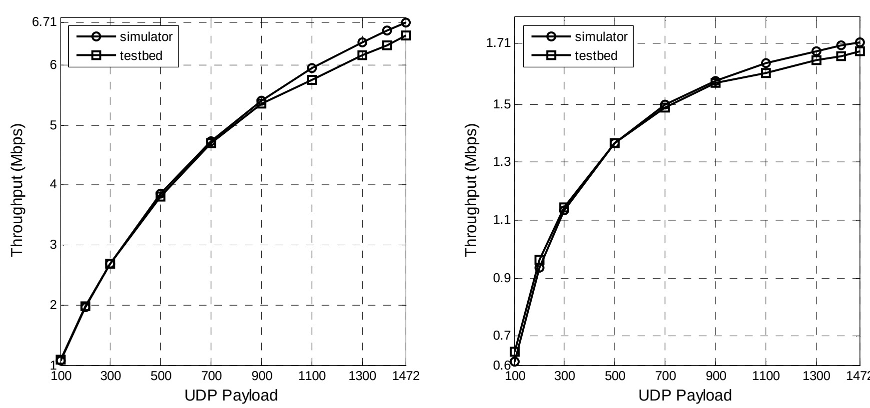

Figure 2 : Unicast aggregate throughput of two nodes versus UDP payload size for IEEE 802.11b

Figure 2 indicates a small discrepancy between the simulator and the test bed performance for payload size smaller than 300 bytes and larger than 900 bytes. Such difference is explained in [17] by means of non uniform distribution of the channel idle time in the test bed. The experiment in [17] accounts for a single unicast flow. In our test bed the possible non uniform distribution of the channel idle time affects the collision probability too; thereby the maximum aggregate throughput.

Besides the small difference in the performance of the test bed and the simulator we have ensured the consistency of their parameter settings. Additionally, we have shown that packet collision in the test bed happens with almost equally probability as in the simulator. Knowing that the performance of the test bed and the simulator matches quite well we can now execute large scale simulations. Our simulations incorporate a new mobility model for public safety that is presented in the next section.

CWmin	31
Preamble type	Long preamble
Slot time	20µsec
SIFS	10 µsec
DIFS	50 µsec
Beacon interval	100 ms
ACK payload	14 Bytes
ACK bit rate	Channel bit rate
Transmission power	15dBm
Sensitivity	-91dBm@2Mbps and -83dBm@11Mbps

Table 1: 802.11 DCF parameters in the test bed and the simulator

GROUP MOBILITY MODEL WITHIN A PUBLIC SAFETY CELL

We assume that a catastrophic event takes place in a low density building area, hereafter the incident area or the cell. The rescue personnel involved in the event are uniformly distributed over the area taking care of individual tasks. At some point a rescue agent brings a car mounted AP that can be connected to the dispatcher via a backbone network. After establishing the connectivity to the dispatcher, the rescue personnel are ordered to be deployed inside a smaller area that belongs to the cell. Hereafter we refer to this area as the hotspot. The personnel start moving towards the hotspot with different speeds. We assume that the personnel located far from the hotspot move with higher speeds compared to the personnel located nearby. The speed of every rescue agent is constant throughout the moving time. The personnel select their destinations uniformly within the hotpot.

We evaluate the public safety applications during the movement of the personnel, nodes, for the mobility model described above. Below we investigate the statistical properties of this model. The analytical method appears rather cumbersome and thus we resort to simulations. We assume a unit circle R=1 in x-y plane and a hotpot area with radius R_0=0.1 units. The hotspot resides completely in the unit circle and its centre is selected randomly. The velocity w.r.t the distance between the initial location and the destination of a node is plotted in Figure 3a. It is assumed that V_{min} = 0.005 units/s and V_{max} = 0.015 units/s.

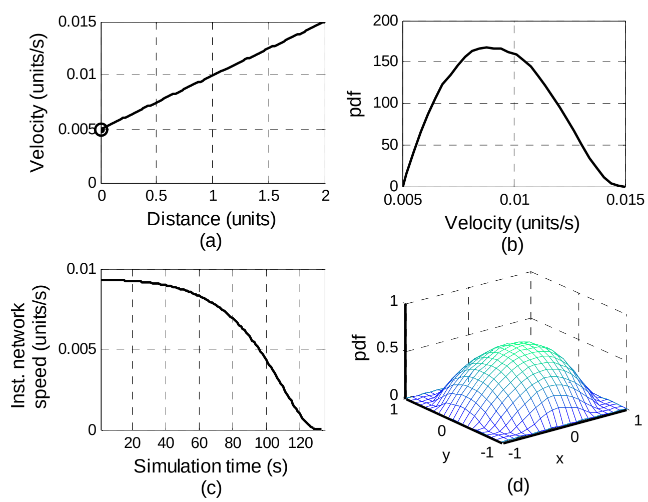

Figure 3 : Properties of the mobility model

Figure 3b shows the probability distribution of the initial speed. For our parameter settings the average speed at the beginning of the simulation equals E[V]=0.0093 units/s. As the simulation evolves the nodes reach their destinations and then become static. Finally, the network would reach a zero speed state. Note that the simulation ends when the last node reaches its destination. Figure 3c presents the instantaneous network velocity as a function of the simulation time. Finally, in Figure 3d we sketch the spatial distribution of nodes. One can deduce that the nodes spend more time close to the center of the cell while moving towards the hotspot.

PERFORMANCE OF MULTICAST VOICE IN THE PRESENSE OF UNICAST VOICE AND VIDEO TRAFFIC

In public safety the first responders get directions about the tactical action plan with multicast voice whereas they report back with unicast flow to the dispatcher. Considering this type of communication, we evaluate the multicast voice performance in the presence of increasing number of unicast flows. Particularly, we simulate the performance of multicast voice within an 802.11 cell for two cases: when the users associated to the AP send unicast voice and when at most one of them transmits unicast video. Note that when a node transmits video, it does not join the multicast group. One should consider the video feed transmitted from a standalone camera that is used to improve the situation awareness at the dispatcher.

Simulation set up

The cell size to be used for the simulation analysis requires few considerations. In the absence of interference, we require that a node located at the cell border should receive the multicast packets without errors. In order to determine the target SNR we use simulations. The frame error rate (FER) is calculated based on the signal power. For independent bit errors within a packet, the FER is a function of the bit error rate (BER) and the packet size. Then, the BER for particular modulation coding (MC) scheme can be derived based on the SNR.

The target SNR for 11Mbps and 2Mbps equals 10dB and 6dB respectively. The corresponding cell radiuses R_c are approximately equal to 240m and 300m for the two-ray channel model. Since the packet loss in unicast flows is protected with positive ACK the selected radiuses are also suitable for the unicast traffic.

For the PHY and the MAC layer parameters we import in the simulator the values assessed in the test bed, Table 1. The bit rate remains fixed throughout the simulation. The number of unsuccessful unicast packet retransmissions before dropping a packet has been taken equal to 4.

For voice communication, we use the G.711 A-law codec that encodes every 10ms of audio into a packet. The constant bit rate (CBR) audio frames consist of 40-byte IP/UDP/RTP headers followed by a relatively small payload. Every millisecond of voice is encoded into 8 bits resulting in 80 bytes per packet. Usually, the VoIP applications employ RTP that adds an additional overhead of 12 bytes. Therefore the UDP payload of each voice packet is 92 bytes. The video communication is modeled as CBR application too. We assume an inter-packet generation gap of 30ms. Including the RTP header the data rate of the simulated application is 360kbps.

Regarding the mobility model we consider minimum user speed V_{min}=1m/s and maximum user speed V_{max}=4m/s as realistic values. The mobility model parameter values result in simulation time $t_{max}=2R_c/V_{max}$=120s and 150s for 11Mbps and 2Mbps respectively. The radius of the hotspot area has been taken equal to R_0=50m.

Simulation results

The performance metrics for voice communication are the PDR and the average delay as defined in section II. The scope is to investigate the performance w.r.t. the number of unicast flows towards the AP in the presence of a multicast flow from the AP. It is expected that the performance would degrade as the number of unicast sessions increases. Due to the lack of link layer ACKs, the degradation of multicast session in terms of PDR is expected to be more severe.

Figure 4a and Figure 4b present the multicast voice PDR for 2Mbps and 11Mbps respectively. The PDR when an additional video feed is present is also depicted. One can see that adding a video feed does not introduce significant degradation. Instead, the multicast performance is affected mainly due to the number of unicast voice connections. This happens because of the significant overhead that the small voice packets carry. It is also interesting to observe that the multicast PDR for 2Mbps is lower but close to the PDR for 11Mbps. We have shown by simulations that the MC scheme employed for 2Mbps experiences less FER for the same signal power degradation compared to the MC scheme for 11Mbps. Therefore simultaneous transmission might not lead always to packet drop at 2Mbps.

Figure 4c and Figure 4d show the average delay of multicast and unicast voice respectively. As it was expected the delay is increasing for increasing number of nodes competing for channel access. The phenomenon is more intense for lower channel bit rates. It is interesting to observe that the average delay is higher for the unicast than for the multicast voice. The reason is that there are no link layer ACK and packet retransmission for multicast.

Note that all the simulations have been repeated 5000 times to ensure the reliability of the results.

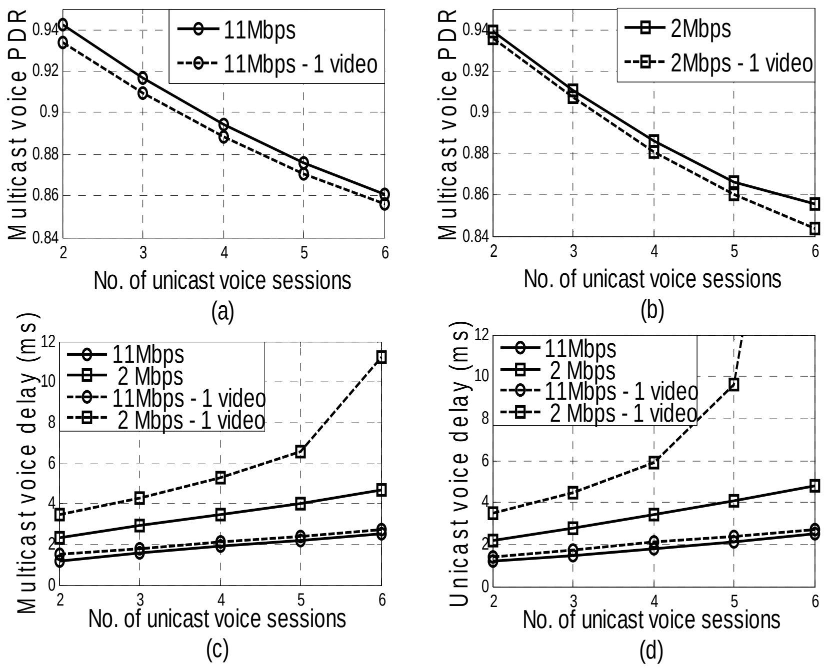

Figure 4 : Simulation results

Figure 4 indicates that for 2Mbps both unicast and multicast voice sessions break when 6 unicast connections and 1 video feed are simultaneously carried by the AP. For unicast traffic the reason is the high average delay that exceeds 150ms. On the other hand, for multicast traffic the reason is the excessive packet loss implying that the voice quality is not acceptable when the PDR falls below 85%.

For 11Mbps the multicast session breaks due to the low PDR when there are more than 6 unicast voice connections. On the other hand the unicast voice and the video experiences 99% PDR even for 6 unicast voice connections. At the same time their average delay remains low.

CONCLUSIONS

In this paper we considered a typical emergency response scenario where the rescue personnel utilize 802.11 communication technologies. The dispatcher coordinates the rescue personnel by multicast voice. At the same time unicast voice for reporting and unicast video for situation awareness are transmitted towards the dispatcher. The dispatcher is connected via the backbone network to the AP but we investigate the performance only within the 802.11 cell. We evaluate the performance of this scenario by simulations. The reliability of our analysis is increased by verifying the simulator against a test bed of 3 nodes. By ensuring same parameter settings and similar collision probability between the test bed and the simulator we extend our simulation analysis for more nodes.

In order to evaluate the quality of the voice sessions we require average delay lower than 150ms. We have also assumed that 85% PDR is a reasonable choice for the target PDR value. However, the results of our analysis are general and not dependent exclusively to the specific values for the PDR and average delay requirements. In particular, the paper concludes that for low channel bit rates the unicast voice sessions and the multicast session would both break as the number of unicast connection increases. However the reason behind the performance degradation is different. For multicast voice it is the low PDR due to the increasing collision probability. For unicast voice it is the increasing average delay due to the extensive retransmissions at the MAC layer. On the other hand for higher bit rates the multicast session breaks first. The PDR for unicast voice remains high due to the link layer ACKs and the average delay would remain low due to the high transmission rate.

In our approach we also recognize few drawbacks. The simulation study has been executed for two ray channel model in the absence of shadowing. It is expected that the fading phenomena would trigger significant degradation that can be mitigated by careful coverage planning. As the next step we consider to evaluate the performance under a similar scenario but with an AP implementing PCF instead of pure DCF functionality.

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