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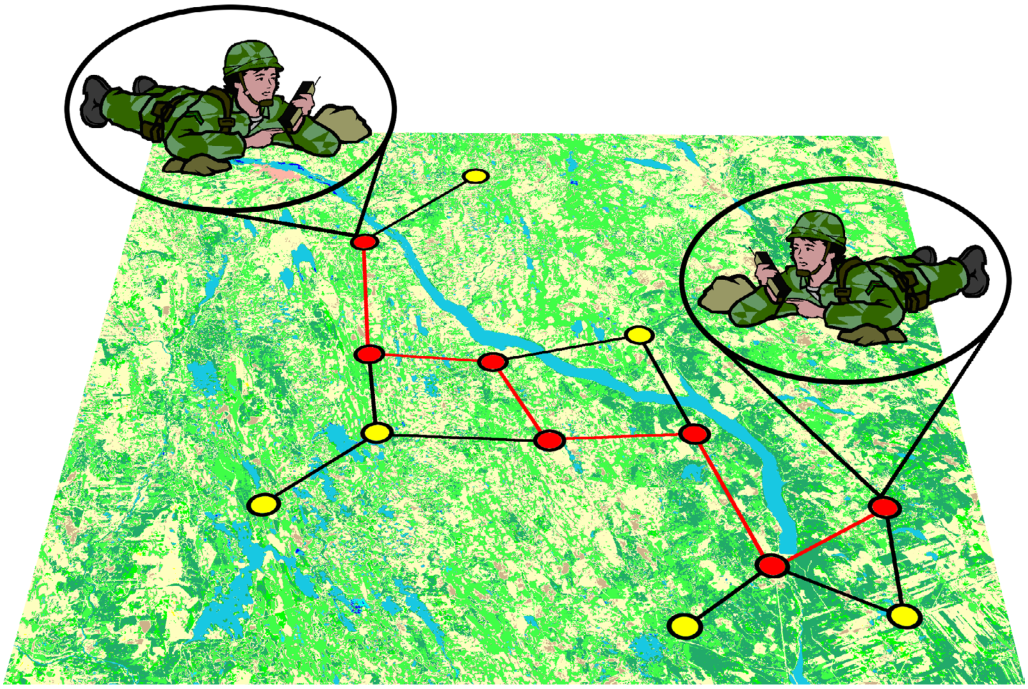

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On QoS and Throughput Tradeoffs for Tactical Ad Hoc Networks

**Abstract**

In wireless ad hoc networking, in particular for tactical networks, the demand for Quality of Service (QoS) provision is increasing. However, to which extent QoS can be provided at reasonable cost is an open issue. To investigate this issue we consider the network capacity, in terms of throughput, that can be supported for best effort and delay sensitive traffic, and a mix of those. That is, our cost measure is related to reduced throughput. In the investigation we consider four different ways of treating the two traffic types. When nothing is done, i.e., all packets are treated equally in the nodes, we have situations where more than half of the capacity is lost. However, by fairly simple methods involving dropping packets and priority queuing much better results are obtained.

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Avvägningar mellan tjänstekvalitè och genomströmning för taktiska ad hoc-nät

**Sammanfattning**

Det finns ett ökat behov av att tillhandahållande tjänstekvalité (QoS) i trådlösa ad hoc-nät, i synnerhet för taktiska nät. En viktig fråga är dock i vilken utsträckning tjänstekvalité kan erbjudas till en rimlig kostnad. För att undersöka detta studerar vi kapaciteten i form av genomströmning för två olika trafiktyper samt en kombination av dessa. Den ena trafiktypen är utan fördröjningskrav och den andra med fördröjningskrav. Kostnaden har vi valt att relatera till minskad genomströmning av trafik i nätet.

I studien jämför vi fyra olika sätt att hantera de två trafiktyperna. Ett sätt är att hantera alla paket lika i noderna. Resultat visar på att omkring hälften av genomströmningen tappas vid en mix av trafiktyperna jämfört med trafiktypen utan krav. De andra sätten att hantera trafiktyperna är genom att slänga gamla paket, använda prioritetsköer i noderna samt en kombination av dessa. Genom dessa enkla metoder uppnås mycket bättre resultat i form av genomströmning.


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# Contents

# Chapter 1

# Introduction

Efficient methods for QoS provision is crucial in military wireless networks. The importance of the QoS issue is also increasing in future wireless commercial systems. Providing cost effective, affordable wireless bandwidth (almost) everywhere is one of the key success factors. However, to combine this with Quality of service (QoS) is one of the key research areas. Furthermore, what the QoS-cost performance tradeoffs look like is of paramount importance.

In the affordable wireless service and infrastructure project [1] these issues are researched. The project considers wireless systems beyond 3G including ad hoc network components. The ad hoc networks are of different types, where one is similar to tactical network, that is, flat, distributed multi-hop networks. Our focus is on tactical networks but our results is also input to the above mentioned project. A key problem we share is to find cost efficient ways to deal with QoS.

An ad hoc network operates without any support of a pre-deployed infrastructure and can be dynamically and quickly created, i.e., the radio stations can be connected and disconnected without involving an administrative station. No centralised station exists in this type of network, i.e., distributed network control is used, which increases robustness. To provide coverage, for example when there is no line-of-sight communication due to terrain, the network supports multihop functionality.

QoS is a very broad term and may cover all the perceived QoS by the user. Topics of importance include context aware services and user assistance down to link layer issues. More or less all the layers are involved and there is a need to both understand cross-layer interactions and what can be done on a single layer.

When controlling the network resources for QoS it is important to assign them where and when they are needed. At the lowest layers adaptive rate [2], power [3] and antennas [4] can be used. Also, the Medium Access Control (MAC) protocol needs to adapt. At the next layer, packets can be re-routed when a link gets jammed. Changing the MAC protocol and routing normally require interactions with other nodes. However, priority queuing, i.e., how packets are treated and given priority in a node, will also have a great importance and is a fairly simple QoS method to implement. No interaction with other nodes are needed, and a limited interaction with other layers.

To get a picture of the QoS and cost tradeoffs we first consider a reference system without any QoS adaption, i.e., all packets are treated equally in the nodes. The reference system is then compared to systems with priority queuing.

The report is organized as follows. In Chapter 2, we describe the traffic models and in Chapter 3 the radio network model. Thereafter, the different methods of priority queuing are specified in Chapter 4. In 5 the networks are described. The simulation results and performance measures are presented in Chapter 6, we make our concluding remarks in Chapter 7 and in Chapter 8 we present some possible topics for future work.

# Chapter 2

# Traffic models

The throughput in a multihop network is very dependent on the actual traffic that the users want to communicate. Here we consider two random traffic models for best effort and delay sensitive traffic. In order to obtain the throughputs, we simulate the actual network traffic that can be supported for the two traffic classes, and a mix of those.

## 2.1 Best Effort (BE) traffic

Best Effort traffic is modeled as point-to-point traffic, i.e. a packet entering the network has only one destination. All packets are of equal size, and are arriving to the network according to a Poisson proces with a mean arrival rate of $\lambda$ packets per time slot. The packet size is equal to 256 bits. By the assumed uniform traffic model, the total average of traffic load entering the network in a given node is then given by $\lambda/N(N-1)$, where $N$ denotes the number of nodes. For BE traffic we will study uniform throughput, that is the highest arrival rate for which the end-to-end delay is bounded. More precisely, the smallest value on $\lambda$ such that one of the link queues are saturated.

## 2.2 Delay sensitive Constant Bit Rate (D-CBR) traffic

The second traffic is called Constant Bit Rate (CBR) and models unicast voice calls between nodes. However, when the delay constraint is imposed in the measurement we call the traffic Delay sensitive Constant Bit Rate (D-CBR). We assume that the voice calls arrive to the network according to a Poisson process and that they have a exponentially distributed duration. During the call the source is assumed to transmit to the destination with a constant bit rate. Notice that the CBR flows are modeling one way connections, from a source to a destination.

The parameters for a voice call are taken from the GSM standard (Enhanced Full Rate) [5]. The vocoder produces bits at a rate of 12.2 kbps, and we assume the packets are of size 256 bits.

The maximal acceptable delay between sender and receiver is selected, as recommended by ITU [6], to 150 ms. We have tested two mean call duration rates: 12 s and 120 s. For D-CBR a call is considered failed when a given percentage of voice packets are delayed more than 150 ms. That is, all packets in the call are then considered lost. Thus, a parameter to chose for the evaluation is the acceptable rate of delayed packets, called packet drop rate. However, dependent on the packet drop rate and traffic load we also get a call failure rate, or equivalently, a call success rate.

To evaluate the D-CBR traffic, different average numbers of CBR flows is tested. A peak D-CBR throughput is obtained. However, it may occur for an unacceptable low call success rate. To illustrate this consider Figure 2.1, where the same 20-node network (Network 2) as in Figure 6.2 is used. We have set the packet drop rate to 5 percent. For about 13.5 CBR flows we get a maximum throughput of about 144 kbps at a mean call duration at 12 s. However, we then have to accept a call success rate of about 90 percent. At call success rates of 95 percent, we get a throughput of about 140 kbps (12.1 flows). Notice that for CBR (without delay constraints) the throughput can be much higher by simply adding more flows. Also illustrated is the impact of the mean call duration. We notice for D-CBR that selecting 12 s or 120 s give similar results. However, selecting 12 s, instead of 120 s, results in a higher maximum throughput for CBR (about 268 kbps, shown in Figure 6.2, instead of 245 kbps). For this sample network the maximal throughput for BE is about 273 kbps. The difference in throughput for CBR traffic and BE traffic is similar to the differences for the other sample networks.

Based on these tests we have decided to chose the mean call duration to 12 s (shorter simulation time). Furthermore, we use a packet drop rate of 5 percent and require a call success rate of 95 percent in the following investigations.

Thus, for D-CBR we calculate the maximal throughput obtainable under the constraint that 95 percent of the calls are successful.

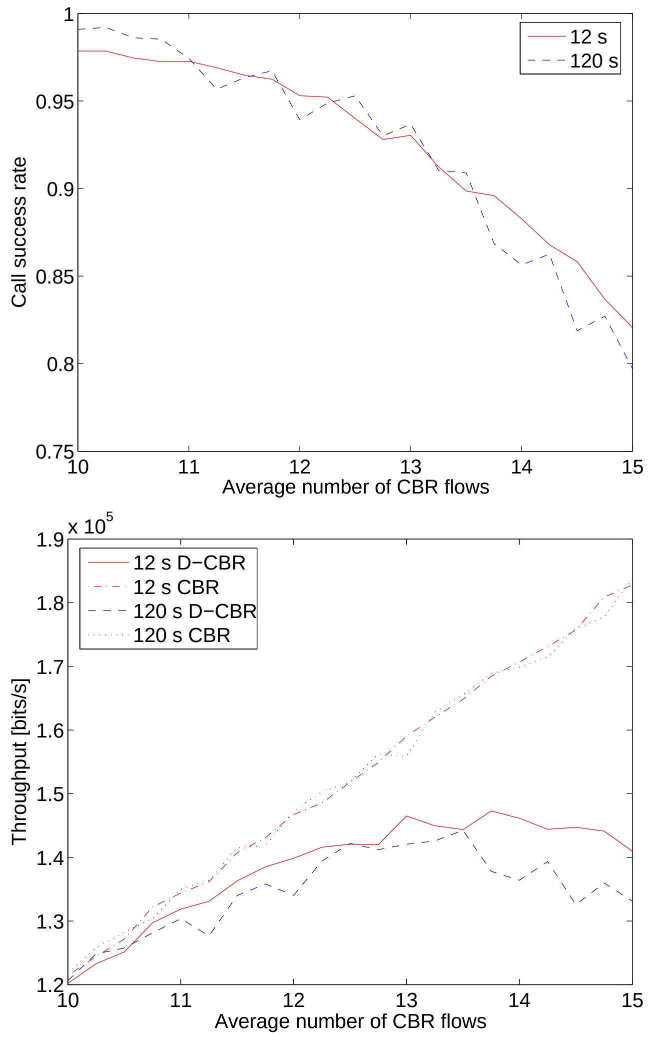


Figure 2.1: Illustration of throughput and call success rate relations, see the text for further explanations.

# Chapter 3

# Radio network model

## 3.1 Link model

An essential part of modeling an on-ground or near-ground radio network is the electro-magnetic propagation characteristics due to the terrain variation. A common approach is to use the basic path-loss, $L_b$, between two nodes (radio stations). We estimate $L_b$ using uniform geometrical theory of diffraction (UTD) model. Furthermore, we use a digital terrain database and all our calculations of $L_b$ are carried out using the ground wave propagation tool, DetVag-90® [7].

We define the signal-to-noise ratio (SNR) as $E_b/N_0$. For any two nodes $(v_i, v_j)$, where $v_i$ is the transmitting node and $v_j \neq v_i$, we can write the SNR in node $v_j$, $\Gamma_{ij}$, as follows

$$\Gamma_{ij} = \frac{P\, G_T(i,j)\, G_R(i,j)}{N_R\, L_b(i,j)\, R_{ij}}. \tag{3.1}$$

Here $P$ denotes the power of the transmitting node $v_i$ (here assumed equal for all nodes), $G_T(i,j)$ the antenna gain of node $v_i$ in the direction of node $v_j$, and $G_R(i,j)$ the antenna gain of $v_j$ in the direction of $v_i$. $N_R$ is the receiver noise power, $R_{ij}$ is the data rate, and $L_b(i,j)$ is the basic transmission path-loss between nodes $v_i$ and $v_j$.

After selecting system parameters we calculate the SNR at the receiver. We assume a bandwidth of 10 MHz and a block size of 256 bits at a packet error probability of $10^{-4}$. In [8] the required SNR to support a given data rate for different block sizes are presented.

In this work we consider spread-spectrum communications and have a fixed data rate on the links of 500 kbps. Let us point out that using variable data rates on the links will improve the network capacity, but will also require a more careful scheduling and routing [2].

## 3.2 Data link layer and above

The traffic, both the BE and the CBR, is routed over the minimum hop paths. After the routing the traffic loads on the links are calculated. The traffic loads are re-calculated every time a CBR flow starts or ends. Thereafter, this information is sent down to the MAC layer, having the task of channel resource assignment.

CSMA is one of the most frequently used MAC protocols in ad hoc networks, but it has inherent problems with providing QoS guarantees. Another MAC protocol that is more suitable from a QoS perspective is TDMA. TDMA is a static, collision-free protocol where the channel sharing is done in the time domain, i.e. time is divided into time slots, with duration $T_s$. Each node is assigned one or several time slots where it is allowed to use the channel. Since each node has a fixed resource allocation, delay-bound guarantees can be made for bounded network loads. In our study the protocol is node-oriented.

Nevertheless, TDMA performs poorly in our situations where the traffic intensity varies on the different links. We have decided to use traffic adaptive TDMA, which performs much better and assigns bottleneck nodes in the network more time slots than other nodes. Further improvements are possible by using Spatial reuse TDMA [9], but the spatial reuse gains will not change our conclusions, from comparing different traffic, in any significant way. The spatial reuse gains are first of all related to network size and connectivity. Here, for simplicity the slot assignment is centralized, however there are ways to distribute the slot assignment, e.g., as in [10].

Before the slot assignment is described in detail let us point out that the traffic adaptive TDMA re-schedules the time slots when a CBR flow start or ends. However, the BE traffic on a link is calculated based on average BE traffic. Furthermore, each node is assigned at least one time slot otherwise that node would be excluded from the network.

To make a traffic adaptive TDMA schedule we need a measure of the capacity requirement, $c_i$, that each node, $i$, has. We define that capacity requirement as

$$c_i = \sum_{\forall j : R_{ij} > 0} \frac{\Lambda_{ij}}{R_{ij}}\,, \tag{3.2}$$

where $\Lambda_{ij}$ is the average amount of traffic over link $(i, j)$, and $R_{ij}$ is the data rate over link $(i, j)$.

We let $M_{ij}^{\mathrm{BE}}$ and $M_{ij}^{\mathrm{CBR}}$ denote the number of BE and CBR flows that passes link $(i, j)$. Furthermore, we let $R_{\mathrm{BE}}$ denote the data rate of the incoming BE traffic and $R_{\mathrm{CBR}}$ denote the data rate of one CBR flow. With this we can write $\Lambda_{ij}$ as

$$\Lambda_{ij} = \frac{R_{\mathrm{BE}}}{N(N-1)} \cdot M_{ij}^{\mathrm{BE}} + R_{\mathrm{CBR}} M_{ij}^{\mathrm{CBR}}, \tag{3.3}$$

where the first term in the sum is traffic from BE flows and the second term is traffic from CBR flows.

Using Eq. (3.3) we can compute the capacity requirement in Eq. (3.2). With that capacity requirement given, our strategy for traffic adaptivity is to assign node $i$ $c_i / \sum_j c_j$ time slots, where the sum is over all nodes in the network. However, since this is a number less than one it cannot be used directly, because each node is assigned at least one time slot. The optimal solution is to find the smallest integer $q$ such that $qc_i$ is an integer for all $c_i$, but this solution may lead to unrealistically long frame lengths. To keep the frame length short, we calculate the slot requirement, $\hat{t}_i$, for node $i$ as

$$\hat{t}_i = \frac{c_i}{\min\limits_j (c_j)}\,. \tag{3.4}$$

This is not an integer, so it cannot either be used directly as a slot allocation. Instead we use it as a measure in the slot allocation algorithm.

If we let $t_i$ be the number of slots currently assigned to node $i$, then the slot allocation algorithm works as follows. First each node is allocated one time slot. Then the node for which the quotient $(\hat{t}_i - t_i)/\hat{t}_i$ is largest is assigned one extra time slot. The last step is repeated until $(\hat{t}_i - t_i) < 0.5, \forall i$ or there are no more time slots to distribute. In this work we have set a constraint on the frame length that guarantees each node at least four time slots per second.

# Chapter 4

# QoS Aware-Queuing

We consider four different methods of serving packets in the nodes. We assume an accurate time synchronization is available, e.g., by GPS. Such a synchronization is required also for the MAC scheduling. Furthermore, we limit the maximum queue length in a node to 500 packets. The four methods are:

**First-come-first-serve (FCFS):** The most common way to deal with packets where the packets are served in the order of arrival.

**Delete packets only:** Requires that a time stamp when the packet is transmitted is stored in the CBR packets. The time is checked in the nodes and a packet is deleted when 150 ms has expired. The advantage is that obsolete packets are removed and do not occupy network resources.

**Priority queue 1:** The CBR packets are stored in an separate queue and served on a separate first-come-first basis. BE packets are served first when the CBR packet queue is empty.

**Priority queue 2:** As above, with the addition that obsolete CBR packets are deleted.

More about queuing methods can be found in [11]. It is clear that more powerful queuing methods can be applied. The key is to try to acquire a more accurate estimate of the time a packet needs to reach the destination. Information from the routing and MAC schedule can be used. Furthermore, nodes can send their queue status information to the neighbours. Also, it can be helpful to use piggy back information about the last packets path in a CBR flow whenever it is available.

Our ambition in this report is not to propose a new and better queuing method, but instead to investigate the possible throughput gains with some simple queuing principles.

**Remark:** With more advanced queuing methods fairness needs to be considered, another dimension in the assessment. Should network resources be used to try to assure that long D-CBR flows have the same success rate as short flows? In fact, we tried a simple method using the number of hops in a D-CBR flow. A D-CBR packet arriving to the priority queue is sorted after: remaining time to live divided with remaining number of hops. By such method the D-CBR flows was treated more fairly, but with the cost of a slight decreased throughput. The average delay was also reduced slightly. An interesting issue for further research is to investigate the cost and fairness tradeoffs.

# Chapter 5

# Networks tested

Connected networks, i.e. all nodes can reach all other nodes through multi-hop, were used, consisting of 20 respectively 40 nodes. The nodes are randomly distributed and scattered over an square area of 1 $km^2$. The terrain is mainly flat although there are some hilly sections. The locations of the nodes in the terrain will determine which pairs of nodes that can establish a link, since the distance and the terrain between the nodes affect the elementary path loss.

The sample networks, consisting of 20 nodes, represent a low, medium and high average number of hops between the source node and the destination node, in this study chosen to 1.3, 1.7, and 2.7 (respectively). These networks are labeled Network 1, 2, and 3 respectively.

The network consisting of 40 nodes represents a network with a medium average number of hops, chosen to 1.8, and are named Network 4.

# Chapter 6

# Results

Consider Figure 6.2 presenting the results for the 20-node network with an average number of hops of 1.7, labeled Network 2. The same network used for the illustration in Section 2. The throughput of BE traffic is shown on the x-axis and the throughput of D-CBR traffic on the y-axis. However, the maximal throughput for CBR (without delay requirement) is also plotted on the y-axis. The total throughput for a mixed traffic is simply obtained by summing up the throughputs on the x- and y-axis.

The straight line (D-CBR vs BE), connects the points for only D-CBR traffic and only BE traffic. When the traffic is mixed the points should follow this line if the traffic mix is scalable. However, as can be seen with normal *FCFS* the total throughput falls below such a scalable mix. This means that the BE traffic affects the D-CBR negatively. On the other hand, for *Delete packets only*, *Priority queue 1*, and *Priority queue 2* we can see an improvement in throughput for the mixed traffic. In particular, we can see a substantial gain with *Priority queue 2*, i.e., when packets older than 150 ms are deleted, and the priority queue gives priority to D-CBR traffic. Notice that the small measure to delete old packets gives an improvement in throughput for the case when we only have D-CBR traffic, the second point ($\times$) on the y-axis.

For Network 1 representing the dense network (1.3 hops), see Figure 6.1, the throughput is increased compared to Network 2, see Figure 6.2. This is due to that the more number of hops a path from source to destination involves, the higher the cost in terms of network capacity will be. This can also be seen in Figure 6.3 which represents the sparse network (2.7 hops), where there is a

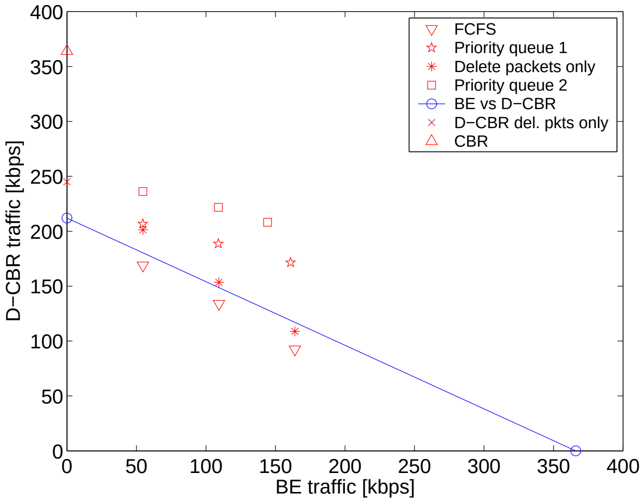


Figure 6.1: Network 1: Dense 20-node network: 1.3 number of hops in average

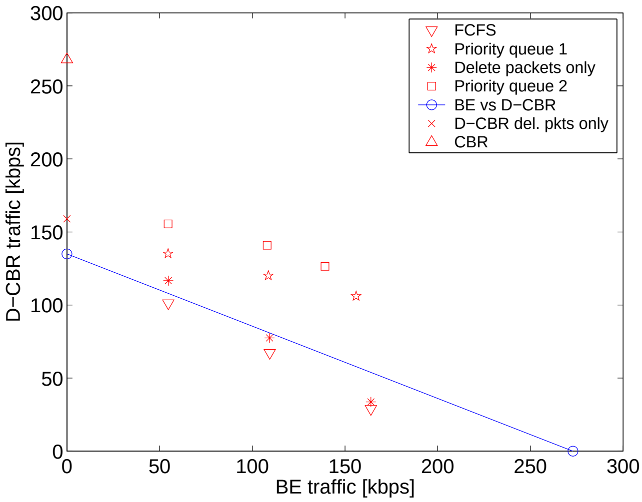


Figure 6.2: Network 2: 20-node network: 1.7 number of hops in average

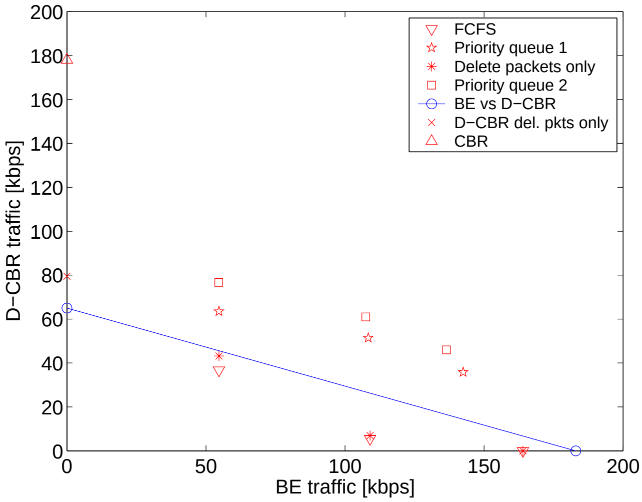


Figure 6.3: Network 3: Sparse 20-node network, 2.7 number of hops in average

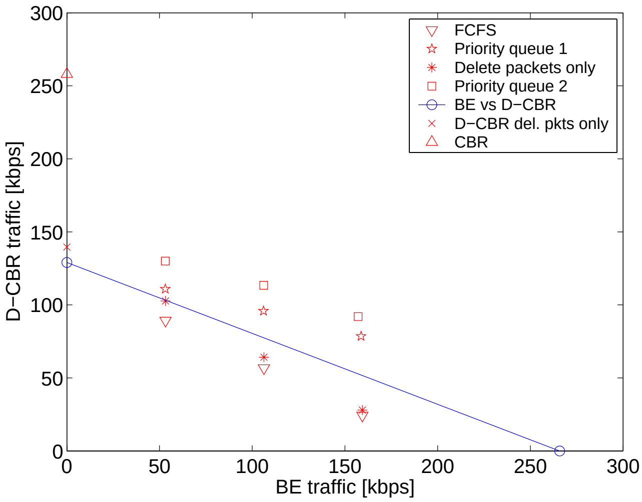


Figure 6.4: Network 4: 40-node network: 1.8 number of hops in average

| **D-CBR only** | 20-node Networks No.1 | No.2 | No.3 | 40-node No.4 |
|---|---|---|---|---|
| D-CBR/BE | 0.58 | 0.49 | 0.36 | 0.48 |
| **Mixed traffic/** Normal (FCFS) | 0.67 | 0.61 | 0.47 | 0.57 |
| Delete packets only | 0.73 | 0.67 | 0.51 | 0.62 |
| Priority queue 1 | 0.91 | 0.86 | 0.68 | 0.74 |
| Priority queue 2 | 0.97 | 0.96 | 0.78 | 0.84 |

Table 6.1: Reduced throughputs: see the text for further explanations.

noticeable decrease in throughput compared to both Network 1 and 2. Further, when BE traffic dominates the traffic mix, the *FCFS* and *Delete packets only* work very poorly. Nevertheless, by using *Priority queue 1* and *Priority queue 2* an improvement in throughput can be achieved compared to *FCFS*. Finally, the relationships between the different methods are about the same for this network compared to the previous mentioned.

Figure 6.4, Network 4, represents the 40-node network with 1.8 hops, close to the number of hops for Network 2. The results show a slight decrease in throughput for the different methods in the larger network. However, the relationship between the different methods are the same as for Network 2.

Should D-CBR be compared to BE or, CBR, when estimating the cost? The traffic distribution is slightly different between BE traffic and CBR traffic, and BE result in a slightly higher throughput. However, since the difference is small, as can be seen in the figures, we only compare to BE. Thus, in the following we present throughputs normalized with BE.

To simplify the reading we summarize some of the results in Table 6.1. The numbers presented are normalized throughputs, or reduced throughputs when compared to BE traffic. In the table we only present results from one possible traffic mix, the one chosen when we have equal amount (50 percent) of D-CBR and BE traffic.

By summing the throughputs of the two traffic classes when they are equal, and dividing with the maximum BE throughput we obtain the numbers in the table. Notice, however, that the numbers are approximations, by linear interpolation, using the results from the obtained traffic mixes from the simulations. To get a pre-defined traffic mix as a result from a simulation an iterative procedure must be used since the BE traffic affects the D-CBR traffic.

In the table (first row) we can for example see, when the normalized throughput is 0.36, that the cost of sending D-CBR traffic is 2.8 times higher than sending BE traffic. Noticeable is also that sending D-CBR traffic is more costly in the sparse Network 3, many hops, compared to Network 1 and 2, fewer hops. Further, for the mixed traffic the improvement in throughput for *Priority queue 2* compared to using *FCFS* is substantial, from 0.47 to 0.78.

## 6.1 Discussion

To estimate the cost of an ad hoc network solution is difficult. It involves many aspects from the cost of building the system to running it. In particular, what is the cost difference of two network solutions. Here, we have chosen the approach to related cost to reduced throughput, or to more required bandwidth, to provide a demanding service. Bandwidth is a cost factor, maybe even more so in the future with software defined radios. Assuming the same hardware radio platform is used, a more powerful system could either be implemented by a more advanced network solution in software, or using more bandwidth.

# Chapter 7

# Conclusions

The report investigates the QoS and bandwidth cost tradeoffs for some sample tactical networks. In particular, we gain insights into the cost issue of transmitting delay sensitive traffic. Of interest is to investigate what happens in the normal case when all packets are treated equally on a *FCFS* basis, to the case when some simple queuing methods are applied. We have not applied advanced queuing systems, for which also the issue of fairness needs to be considered.

The cost is measured in terms of reduced throughput when delay sensitive traffic is served, compared to best effort traffic only. The results show a substantial cost increase for serving delay sensitive traffic, unless something is done about it. The cost increases more than two times in some of the cases. Of particular interest is the much better results that can be obtained by fairly simple methods, such as deleting old packets and using priority queues. Nearly two times improvements can be seen in the examples tested. Using those methods becomes essential when a mix of traffic is served, in particular when a large portion of the traffic is of best effort type.

# Chapter 8

# Future work

The next major step is to consider the influence of mobility on the possibilities to support different types of traffic. Mobility will have a number of implications on the network control as well as on how the assessment should be carried out. In this report we have adapted to traffic changes, however, mobility will have a large influence on the performance. The link quality, connectivity and overall network topology will change. To deal with the changing conditions the network has to be monitored and data rates, MAC schedule and routing have to be adapted. Some of the issues we need to consider in the future work are the following.

**Protocol interaction:** The network has to adapt to the changing topology due to mobility. We need a well balanced adaptation and interaction between the options: adapt data rate, re-schedule the MAC scheme and re-route the traffic. To adapt the data rate can be done rather quickly but re-scheduling a TDMA based protocol takes longer time. Furthermore, when, and how, to re-route the traffic flows need to be addressed.

**Traffic models:** The best effort traffic model can be maintained, however the model for delay sensitive traffic (voice calls) may need to be modified. One reason is that the latter traffic is modeled as a CBR flow of many packets and that the mobility will introduce a large variation in the amount of traffic that can be supported at a given moment. The measurement of that traffic throughput will be very cumbersome with the present assumption that an average success rate of 95 percent is required. An average over a whole scenario is required. Another reason is that it is unclear how well our simple model really reflects the quality of a voice call.

**Mobility model:** Two commonly used random mobility models are the random waypoint and random walk model. Another option is to use a deterministic model, that is, define the movements in a scenario, e.g., as done in the projects TRAN and KOFTA [12, 13]. The issue here is what we want to investigate, algorithm performance or try to assess what can be expected in terms of service availability in a given scenario.

**Assessment:** Our first aim is to investigate the improvements with variable data rates, as compared to a fixed rate, in supporting delay sensitive traffic in a mobile scenario. In doing this we will start by applying a type of ”genius” MAC having full network knowledge and which can adapt without delay as well as ”perfect” routing. Thereafter our aim is to introduce realistic delay in the adaptation process and employ practical algorithms for MAC and routing to further investigate how to do protocol adaptations and interactions in order to deal with mobility.

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