How to Assign Traffic Sources to Nodes in Disaster Area Scenarios

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Abstract

This paper deals with the characteristics of multicast push to talk voice traffic sources for disaster area scenarios. The goal is to design models that can be used to assign voice traffic sources to nodes. The modelling is based on an analysis of real-life measurements during a catastrophe maneuver. The analysis shows that about half of all calls originate from the communication head of a talk group. Based on this characteristic, different models are considered. Synthetic distributions of traffic sources for the different models are generated and evaluated by statistical analysis. Finally, the impact of the different models is evaluated in an exemplary simulative network performance analysis.

1 Introduction

In catastrophe situations, there is a need for reliable communication systems. The work of public safety units and the impact of disaster relief depend on these systems. Due to the fact that any kind of infrastructure may have been destroyed by the catastrophe, there is a demand for communication systems independent from infrastructure in this "disaster area scenario". Especially for Mobile Multi-hop Ad hoc NETworks (MANETs) this scenario follows as a quasi canonical scenario. MANETs meet the requirement of being independent of any kind of infrastructure by their very definition.

For the evaluation of network performance aspects, modelling the data traffic realistically is an important issue. The distribution of the traffic sources may only have a minor impact on the evaluation result as long as the nodes move randomly over the whole simulation area. In MANET simulation studies uniformly or exponentially distributed constant bit rate traffic is widely used (e.g. [4], [10], [6], [13], [14]). Some approaches (e.g. [8]) simulate voice traffic based on studies in telecommunication or cellular systems, others (e.g. [11], [7]) use file transfer protocol (FTP) traffic for performance evaluation. There are several traffic models. However, in all papers mentioned, the traffic sources and sinks are uniformly distributed over all nodes or at least all nodes of a communication (e.g. multicast) group.

However, units in disaster area scenarios do not move randomly over the whole simulation area. Instead, the movement in disaster areas scenarios is area-based (cf. [1], [2]): Each unit is sent to a specific working area. Thus, the distribution of the traffic sources in a disaster area scenario focussing on appropriate distribution models is a relevant research area.

The goal of this paper is to present the results of analysis concerning the distribution of voice traffic sources in a disaster area scenario and to show how this distribution can be integrated into an appropriate traffic model for disaster areas [3]. Our study is based on an analysis of traffic measured during a civil protection maneuver.

The remaining parts of this paper are structured as follows: Section 2 describes the voice communication used in disaster areas and proposes several characteristics concerning the distribution of the traffic sources. In section 3 we describe the measurement architecture, the concrete scenario in which the data was acquired, as well as the generation of time series. Next, we analyze the time series with respect to dependencies (section 4) and validate the characteristics anticipated in section 2. In section 5 we derive appropriate distribution models based on these characteristics. After this, we evaluate statistical properties of the different models (section 6). Based on this, we show that the characteristics of the different models have an impact on simulative network performance analysis (section 7). Finally, we conclude the paper and point out topics for future work (section 8).

2 Voice Communication in Disaster Areas

Today, the main application in disaster areas is voice communication: different users communicate via push to talk voice calls. The users that communicate with each

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other (*talk group*) share one broadcast voice communication channel. Technically this broadcast voice communication channel may be realized e.g. as a separated physical channel or as a multicast group. The term *talk group* does not restrict the technical realization.

In the future, voice communication may evolve: The voice codecs may change, the technical realization of a talk group may change, or the voice traffic may become video traffic (video-phone). However, the communication time will still be dominated by the actual message to be said. The message to be said (holding time), the pause between two messages (idle time), as well as the one who speaks will not change. Considering the distribution of traffic sinks, each member of a talk group is a sink of his traffic group (as long as his device is switched on).

In a disaster area scenario there are typically both local and global talk groups. Local talk groups communicate inside a specific area (e.g. a casualties clearing station). The global talk groups are used for communication across the whole disaster area (e.g. a command channel). Thus, the position of the nodes inside the different areas (cf. [2]) influences the distribution of traffic sources and sinks especially as far as the local talk groups are concerned.

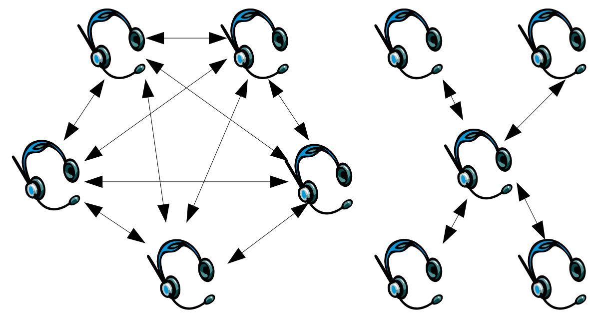
In general, each call of a talk group is done by one sender that starts speaking and stops after a certain amount of time. There is only a half-duplex connection (unlike a telephone call): while one user speaks, the others have to listen. Different calls with semantic connection (e.g. question and answer) may be regarded as one conversation or session, where a conversation consists of an arbitrary number of calls between two callers, and, typically, the callers alternate in calling each other. However, in each session there are only two users that communicate directly (speak). Nevertheless, all the other members of the talk group need to listen to the whole communication for tactical reasons.

Furthermore, there are two different communication patterns used for group communication (cf. fig. 1) that influence the distribution of two callers of a conversation:

- *mesh communication*: Each user is allowed to communicate with any other user.

- *star communication*: There is one user that is the head of this talk group. Each user is only allowed to communicate with this head. The other users are not allowed to communicate directly.

In catastrophe situation as well as in maneuvers star communication patterns are used to manage the flood of information. However, even if there is a head and the users know that they should communicate using star communication, they sometimes communicate directly with each other. Thus, in reality there is a mixed form of star and mesh communication patterns. For modelling the distribution of traffic sources the fraction of each traffic pattern has to be examined. Furthermore, it has to be examined whether the calls of two users really alternate. Another important aspect when modelling the star communication pattern is who starts the conversation, i.e. whether the main fraction of all conversation is started by the head (e.g. polling for some information) or do the other users start conversations (e.g. reporting events) on their own.



(a) mesh communication

(b) star communication

Figure 1. Communication patterns

3 Measurement Architecture

In this section we describe our measurement architecture that allows us to distinguish between head and group calls.

Ideally, the traffic model should be based on load in real catastrophe situations. (Un)Fortunately catastrophes are quite rare, happen all over the whole world, and most importantly can not be planned. Thus, we decided to measure in a catastrophe maneuver. The maneuver we base our analysis on took place in November 2005 during a course for squad leaders. The scenario was that more than 40 people were injured by a catastrophe in an event hall. More than 40 disaster area units joined the maneuver.

In Germany, the public safety units use the analog frequency modulated German national radio system, called BOS-system. Several frequencies are reserved, e.g. 68-87.5 MHz (4m channels) and 146-174 MHz (2m channels) are reserved for the BOS system. In the maneuver measured one channel (168.56 MHz) was used.

On this channel it is not possible to distinguish between different senders technically. To distinguish between head and group calls, there are two possible approaches:

- *Speaker recognition*:

  One possible way is to use methods of speaker recognition [12]. The calls of the head may be separated by the speakers speech patterns. As long as the speaker at

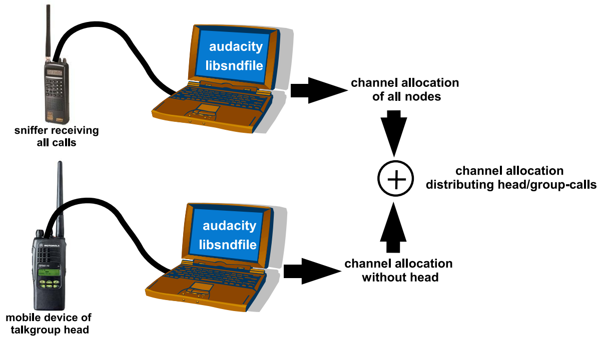


Figure 2. Measurement system

all conversations	
$P_{\text{conv}}(H) = \frac{195}{423} \approx 0.46$	$P_{\text{conv}}(G) = \frac{228}{423} \approx 0.54$
$P_{\text{conv}}(H_i\|H_{i-1}) = \frac{103}{195} \approx 0.53$	$P_{\text{conv}}(G_i\|G_{i-1}) = \frac{135}{228} \approx 0.59$
$P_{\text{conv}}(G_i\|H_{i-1}) = \frac{92}{195} \approx 0.47$	$P_{\text{conv}}(H_i\|G_{i-1}) = \frac{93}{228} \approx 0.41$
only star communication conversations	
$P_{\text{conv}}(H) = \frac{195}{392} \approx 0.497$	$P_{\text{conv}}(G) = \frac{197}{392} \approx 0.503$
$P_{\text{conv}}(H_i\|H_{i-1}) = \frac{107}{195} \approx 0.55$	$P_{\text{conv}}(G_i\|G_{i-1}) = \frac{108}{197} \approx 0.55$
$P_{\text{conv}}(G_i\|H_{i-1}) = \frac{88}{195} \approx 0.45$	$P_{\text{conv}}(H_i\|G_{i-1}) = \frac{88}{197} \approx 0.45$

Table 1. Probabilities for the initiator of a conversation (H = head originated, G = group originated)

the head device does not change (e.g. due to end of his working time) this approach appears promising. But when looking into the details there are a lot of challenges. The signal is sometimes very noisy and the speakers are sometimes quite stressed. Furthermore, the trace we observe contains an unknown set of speakers of unknown size. So it is basically a worst case for speaker recognition, which is even under good conditions still quite error-prone with current algorithms.

- *Technical separation*:

  The BOS-system that is used, is a half-duplex communication system. Thus, the device that is used to perform the call does not receive the call itself. To distinguish between head and group call, one has to scan the audio stream at the device of the head. The advantage for this approach is that the technical challenges like time synchronization of the two scanners are resolvable. However, there are organizational challenges, because the measurement has to be performed on the device of the head of the talk group.

During our measurement we followed the second approach and used the measurement system shown in figure 2. The system consists of two parts: one measuring all calls (upper part in the figure), and the second measuring all calls that are not performed by the head (lower part in the figure). For the first, we used a radio scanner Uniden Bearcat UBC60XLT-2. For the second, we used a Motorola GP360-11b. From both devices the received audio stream was analyzed for channel characteristics like channel holding times and idle times using the same methods as described in [3]. Finally, the derived time series could be compared figuring out whether a call was made by the head or the group.

Furthermore, we grouped the calls to conversations using a time threshold based approach (cf. [3]): If the idle time between two calls was larger than a threshold of 3s, a new conversation is assumed. Grouping calls to conversations and knowing the origin of the calls, the initiator of each conversation (head or group) is also obtained.

4 Analysis of Traffic Distribution

During our measurement we observed 1511 calls. 707 of these came from the head, 804 from the group. The calls group into 423 conversations. 195 of these were started by the head, whereas 228 were started by another group member.

To distinguish between star and mesh conversation pattern we assume that a conversation is a mesh conversation if it is not containing any call from the head. Theoretically (cf. description of communication patterns in section 2) mesh conversation can also contain calls from the head. Nevertheless, during our measurements we observed either star communication between the head or a group member or mesh conversation between two group members.

Thus, to examine the fraction of mesh communication we counted the conversations which contain no call from the head and consist of more than one call. 31 of the 423 (less than 8%) follow the mesh communication pattern. When the conversation is started by a group member, the probability that it is a mesh conversation is:

$P(\text{compl. conv.}|G) = \frac{31}{228} = 0.136$

When considering star communication only, 195 of 392 conversation were started by the head, whereas 197 were started by another group member (see also table 1). The distribution of starters between head and group seem to be equally distributed.

Furthermore, we examined whether the initiators of two conversations are dependent or independent and identically distributed. We examined autocorrelation as well as conditional probabilities for the initiators of two successive conversations. The autocorrelation function showed no significant hints. The calculated conditional probabilities for the initiators of two successive conversations are shown in table 1. The table shows the results for all conversations as well as for the case where we ignored mesh communication con-

$P_{\text{call}}(H) = \frac{479}{1025} \approx 0.47$	$P_{\text{call}}(G) = \frac{546}{1025} \approx 0.53$
$P_{\text{call}}(H_i \mid H_{i-1}) = \frac{68}{479} \approx 0.142$	$P_{\text{call}}(G_i \mid G_{i-1}) = \frac{104}{546} \approx 0.190$
$P_{\text{call}}(G_i \mid H_{i-1}) = \frac{411}{479} \approx 0.858$	$P_{\text{call}}(H_i \mid G_{i-1}) = \frac{442}{546} \approx 0.810$

Table 2. Probabilities for the initiator of a call (H = head originated, G = group originated)

versation. It shows the probability for the originator (head or group) of the next conversation knowing the originator of the current conversation. There are only small dependencies concerning the initiators of successive conversations. These small differences may also be caused by the small amount of samples.

Next, we focus on the calls inside the star communication conversations. One characteristic of the calls inside a conversation was that the callers alternate (cf. section 2). To verify this property we counted the number of conversations that contain two calls from one caller in a row. 98 of our 392 conversation (25%) do not fulfill the alternation criterion. The reason for this are supplementary calls: One caller says something, stops for a short while, sometimes also releases the push to talk button, before adding additional information. For the traffic model it makes sense to model this as two different calls with an idle time between, because the medium is not occupied perpetually. Nevertheless, for the distribution of traffic sources the two calls do not alternate.

Concerning these 25% of non-alternating conversations, we decided to calculate conditional probabilities for the callers of two successive calls. The results (cf. table 2) show that the alternation property holds but sometimes (about 20%) two calls of the same caller follow each other.

5 Distribution Models

In general, to model the mesh communication pattern we assume an equal distribution of traffic sources. According to the mesh communication pattern there is no reason why one station should send more frequently. Thus, the two callers of a conversation may be chosen uniformly from all members of the talk group.

When modelling star communication, a larger number of calls is originated by the head of the talk group. Thus, an equal distribution of traffic sources can not be assumed. According to the star communication pattern every second call can be assumed to be a call from the head. The peers that communicate with the head may be assumed to be uniformly distributed. According to the star communication pattern there is no reason why any other group member should originate a larger number of calls.

Next, we describe different models with increasing level of detail. With increasing level of detail, we include more properties examined in the previous section.

- *Randomly distributed*:

  This is the basic distribution model most commonly used in literature (cf. section 1). The traffic sources are uniformly distributed over all nodes of a talk group. This approach does not model any characteristics examined in the previous section. Therefore, we use this approach mainly for reference purposes.

- *Alternate - conversation always started by head*:

  This approach models the star communication pattern in a basic way. The calls of a conversation alternate between the groups head and a uniformly chosen member of the group. All conversations are started by the group head.

- *Alternate - initiator of conversation according to distribution*:

  Extends the previous approach by choosing randomly (according to the distributions of the previous section) whether the conversation is started by the head or a randomly chosen group member (uniform distribution).

  This approach does not model mesh communication patterns. Thus, for this approach the probability of a conversation started by the head without considering mesh traffic should be used. Otherwise, the fraction of conversations started by the head compared to the ones started by the group would be too small.

- *Choose next caller according to distribution - initiator of conversation according to distribution*:

  Extends the previous approach by not alternating between head and group member strictly. The next caller (group or head) inside a conversation is chosen randomly depending on the previous one using the conditional probabilities calculated in the previous section.

  This approach still does not model mesh communication patterns. The probability of a conversation started by the head without considering mesh traffic is used. Furthermore, the conditional probabilities choosing the caller without considering mesh traffic are used (see table 2). However, when using these probabilities independent from the count of calls per conversation it may seldom happen that a conversation without a head call results.

- *Choose next caller according to distribution - initiator of conversation according to distribution - model mesh communication explicitly*:

  Extends the previous approach by modelling mesh communication. If the first caller is not the head, it

Model	Frac. of Calls by Head	Frac. of Conv. started by Head	Frac. of Conv. non-alternate	Frac. of Conv. mesh com.
random	0.0732	0.0747	—	0.5021
1st head	0.5922	1.0000	0.0000	0.0000
distri.	0.4989	0.4948	0.0000	0.0000
distri. dep.	0.4883	0.4991	0.2236	0.0174
mesh	0.4506	0.4594	0.2196	0.0562
trace	0.4679	0.4610	0.3050	0.0733
trace (star only)	0.4673	0.4974	0.25	0

Table 3. Statistics for different models

is chosen randomly with probability 0.136 (cf. section 4) whether the communication is to another member of the group (not the head). If it is not the head, the conversation is a mesh communication.

The distributions used for a conversation started by the head is now used considering mesh communication pattern. The calls between two members are distributed according to the conditional probabilities in table 2 assigning the probabilities of the head to the second group member.

None of the models proposed considers dependencies over more than one call. Furthermore, it is assumed that there are no dependencies between the call length and probability of a conversation not containing a call of the head (mesh communication). To model these details a larger amount of samples would be needed and would result in more detailed and complicated models. Thus, we leave it for future work and show the impact of the models presented as impact on statistics as well as on simulation results.

6 Statistical Evaluation

In this section we evaluate the statistical characteristics of traffic distributions according to the models proposed in the last section. To use the distribution models, appropriate traffic is needed. We generated multicast push to talk traffic according to the three-state semi-markov model with lognormal state holding time distributions [3]. We used this traffic instead of the ones measured because it bases on a larger amount of samples and an appropriate fitting.

We generated one trace of 18,000s seconds length. According to our traffic model, this results in 2511 calls grouped in 956 conversations. For each of the models proposed in the previous section we generated 20 traffic streams.

As metrics for the statistical analysis we used the *fraction of calls by the head*, the *fraction of conversations started by the head*, the *fraction of conversations that do not alternate*, the *fraction of conversations following a mesh communication pattern*. The results for the different models are shown in table 3.

In general, the results show that the more detailed the model the better the statistical fitting is to the distributions of the trace. Alternating with all conversations started by the head seems too simple. As a result the fraction of calls started by the head is too high: All conversations that consist only of one call are made by the head. Thus, the fraction of calls by head for this model is too large (cf. 3).

If we distribute the initiator of a conversation according to the distribution observed, the model is much closer to the trace. However, there are no dependencies modelled. This also explains why the fraction of calls by head still shows small differences.

Considering also dependencies between calls achieves better results. However, without modelling mesh communication patterns explicitly, this model fits to the trace where only star communication was analyzed.

Additionally, modelling mesh communication patterns explicitly results in the best fits concerning the original trace. However, there are still some shortcomings concerning the fraction of conversations with mesh communication patterns. A reason for this may be the dependencies between the call length and probability of a mesh communication that are not considered in these models.

7 Evaluation by Simulation

In this section we perform an exemplary simulation study. The goal is to show that the different models have an impact on simulative network performance analysis.

7.1 Setup

The scenario (cf. tactical map in figure 3) is based on a large disaster area maneuver. The whole disaster area is approximately 350m x 200m. The scenario was that more than 250 people were injured by a catastrophe in an event hall (incident location). There is one *patients waiting for treatment area*, directly in front of the hall. The patients are taken to four *casualties clearing stations*. Furthermore, there is a *technical operational command* and an *ambulance parking point*.

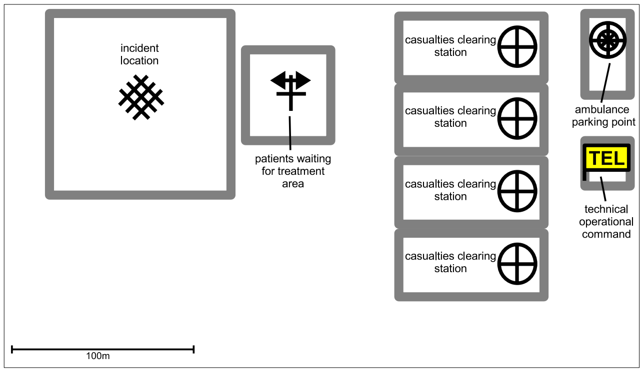


Figure 3. Scenario - tactical map

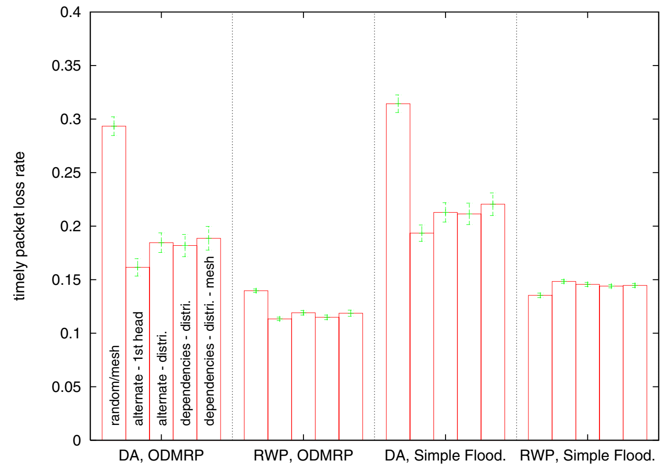


Figure 4. Timely packet loss rate for global group

More than 955 disaster area units (firefighters, paramedics, etc.) and 279 vehicles of four administrative districts joined the maneuver. Only a subset of the 955 units was equipped with his own communication device resulting in 150 mobile devices.

We performed simulations using ns-2.29. A fixed communication range (TwoRayGround propagation model) of 100m was used. As MAC protocol, IEEE 802.11b Wireless LAN with 11 MBit/s was chosen. As routing protocols we used simple flooding as well as the On-Demand Multicast Routing Protocol (ODMRP) [15].

The simulation area was 350m x 200m. As movement pattern we used random waypoint (RWP) moving over the whole simulation area as well as the disaster area (DA) mobility model where the node movement is restricted to tactical areas with parameters taken from [2].

The traffic was modelled (as also described in the previous section) as disaster area multicast voice traffic using a three state semi-markov model as proposed in [3]. In analogy to the maneuver there are six multicast talk groups: five local ones and one global command channel. We modelled the voice traffic according to the MELPe [5] codec (developed for tactical comunication) with 1.2kbps and Forward Error Correction (FEC) of 1:2. This results in 21 byte IP payload every 67.5ms.

As simulation time we chose 3,000s and performed ten replications for each distribution model. We chose such a large simulation time due to long movement cycles and varying traffic characteristics of the models used.

7.2 Metrics

To examine whether the traffic distribution has an impact on the performance of the network, we measured the *timely packet loss rate* - a combination of packet loss rate and packet delay:

$$PLR_n^\delta = \frac{|P_n - R_n^\delta|}{|P_n|}$$

where n is a node, R_n^δ is the number of application data packets the node n received with a delay smaller than δ, and P_n is the number of application data packets the node n should have received. The traffic simulated is voice traffic. Thus, the packet delay δ has a decisive impact on the packet loss rate. A packet that is too late will not be of any use for the voice data communication. Thus, we assumed a packet with a transmission delay larger than a threshold as lost. According to [9] a time of 150ms can be assumed as a threshold.

The timely packet loss rate can also be calculated for a group:

$$PLR_G^\delta = \frac{\sum_{\forall n \in G} |P_n - R_n^\delta|}{\sum_{\forall n \in G} |P_n|}$$

where G is a talk group, R_n^δ is the number of application data packets all nodes n of the group G received with a delay smaller than δ, and P_n is the number of application data packets all nodes n of the group G should have received.

Furthermore, we examine the number of hops for a data packet until it reaches its destination.

7.3 Results

The strongest impact may be seen at the results of the global command talk group. Thus, we focus on presenting the results of this group.

Figure 4 shows the average PLR_G^{150ms} of the global group and 0.95 confidence intervals for 20 replications for

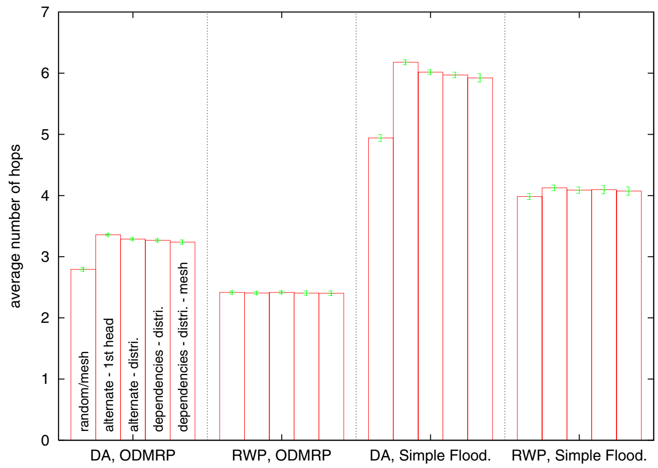


Figure 5. Average number of hops for one node of the global group

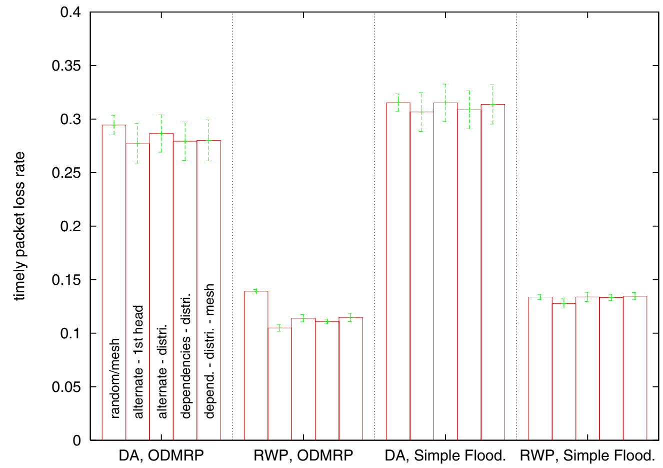


Figure 6. Timely packet loss rate for one ambulance node

each model, routing protocol, and mobility model. Considering the random waypoint mobility model (RWP) the different distribution models for the traffic sources show only small differences. The disaster area mobility model (DA) shows worse PLR_G^{150ms} overall. This difference between the two models comes from the separation to areas and higher node density in this model (cf. [2]).

The traffic distributions show significant differences for the PLR_G^{150ms} of the whole group. Distributing traffic sources randomly yields a significant larger packet loss rate compared to the other distributions. A reason for this is the higher probability in this model that a node in an area with high node density becomes the traffic source. In these areas the probability of packet loss is higher. In the other distribution models only about half of the traffic sources may be distributed in such areas. The other half is originated by the head of the group, which position is near to the boarder (*technical operational command* in figure 3). Thus, the probability that a packet reaches a destination is higher for packets originated by the head. This also leads to the lowest PLR_G^{150ms} when all the conversations are initiated by the head. This leads as shown in section 6 to a larger fraction of calls originated by the head. The other three models do not show significant differences.

Next, we examine the average number of hops for one stationary node. This node has a position at the *ambulance parking point* near the border. The average number of hops (see figure 5) for the global talk group shows that the paths are shorter for the random distribution. The reason for this is that the probability of choosing a traffic source in the middle is higher for the random distribution. The group head (*technical operational command* in figure 3) for the other distribution is placed near to the boarder. Thus, more hops are needed to reach the members of the global talk group.

In general, the PLR_G^{150ms} and average number of hops seem to show that the new distribution models lead to more optimistic results. However, there is a certain impact when considering some specific nodes. There are some nodes that model ambulances which pick up a patient at a *patients waiting for treatment area* and take him away to a hospital. These nodes are less stationary and travel over the complete simulation area.

Figure 6 shows the PLR_n^{150ms} for one of the ambulance nodes. The results show that there are no significant differences between the different distribution models. Compared to the global view using PLR_G^{150ms} the results of the new distribution models are as worse as for the random distribution. The reason for this lies in frequent route breaks due to higher mobility of the ambulance node. Due to its specific behavior the node has not perpetually a (good) connection to the talk group head. But, when thinking of the tactical scenario it is important that all nodes can communicate with their leader. Thus, even if the new more realistic distribution models show more optimistic results concerning the PLR_G^{150ms}, for some specific nodes the results are worse.

Similar to the results of the statistical analysis in the previous section a random distribution seems not to accurately model the traffic source distribution observed in the measured scenario. Thus, we propose to use a model in which the next caller inside a conversation as well as the initiator of a conversation is chosen based on the results of the measurements.

8 Conclusion and Future Work

In this paper, we analyzed the distribution of traffic sources to nodes based on a measurement during a catastrophe maneuver. The analysis shows that about half of the alternating push to talk calls are originated by the group head. The next caller inside a conversation as well as the initiator of a conversation should be chosen according to the analyzed probabilities.

The analysis of the measured traffic distribution led to new more detailed models of the distribution of the traffic sources. Different levels of detail were examined concerning the observed characteristics. In combination with an appropriate traffic model, we generated traffic streams including distribution to nodes for the different models.

The statistical analysis shows that the more detailed the models yield better statistical fittings to the distributions of the trace. Finally, the impact of the different models on network simulation was examined. The models that consider non-random distribution of traffic sources show significant differences when examining timely packet loss rate and average hop count for the global group as well as for a single node of this group. The level of detail modelling non-random distribution of traffic sources has an impact especially considering nodes with outer positions. In general, distributing traffic sources according to our new model that considers dependencies based on our measurements discloses new information and has a significant impact on performance analysis.

In the future, we plan to perform further measurements to sustain our results. Furthermore, we plan to examine the impact of the different distribution models in other scenarios.

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