Efficient wireless multicast retransmission techniques based on multiple coded packets

Pedro R. S. Lopes, Yuri C. B. Silva, and Francisco R. P. Cavalcanti
Wireless Telecom Research Group - GTEL, Federal University of Ceará - UFC, Email: {pedro,yuri,rodrigo}@gtel.ufc.br

***Abstract*—This paper evaluates the performance of four algorithms for wireless network multicast transmission in a single cell. An efficient retransmission technique for multicast services based on network coding with a buffer is proposed and compared to other traditional schemes, such as the case with no retransmission, a simple retransmission and retransmission with multiple coded packets scheme. It is shown that the proposed algorithm outperforms other techniques and quality of the multicast transmission is improved.**

I. Introduction

Through wireless multicast transmission [1] various users can receive the same information which is transmitted at the same radio resource. A resource can be understood in various ways such as a time-slot, a particular frequency or a spatial location. This form of transmission is very useful, for example, in the digital TV application, where several users of a particular cell generate a demand for data on the same channel. In the case of UTRAN and its evolution, the multicast services are specified by the MBMS standard [2], which defines resource allocation and transmission procedures specific for multicasting.

One particular bottleneck of wireless multicast services is the retransmission of erroneously received packets. Since a single resource is used, the retransmission of a packet occupies the resource and those users who had received it correctly have to wait until a new packet is transmitted. Some efficient retransmission algorithms for error-tolerant multicast services have been proposed in [3].

Network coding [4]–[7] combines packets before transmitting them and allows for a natural and efficient means of loss recovery in the face of low-quality wireless links and provides for economical path diversity, which is particularly important for multicast traffic in the unstable and lossy environments characteristic of wireless networks. In this work, we will consider only a single-hop scenario.

Network coding has some flexibility in terms of how to select which packets to combine, allowing to properly exploit the diversity of the multiple radio links. By mixing packets, network coding is able to reduce the number of transmissions necessary to convey packets to multiple receivers, which can lead to a large increase in performance for multicast traffic. Thus, network coding has the advantage of sending different packets in a single resource, which are coded into a single packet.

This work was supported by the Research and Development Centre, Ericsson Telecomunicações S.A., Brazil, under EDB/UFC.25 Technical Cooperation Contract.

In [8]–[10], network coding techniques are proposed to reduce the number of broadcast/multicast transmissions from one sender to multiple receivers, thus increasing the bandwidth efficiency of reliable multicast in a wireless network. The main idea of employing network coding to multicast retransmissions is to allow the sender to combine and retransmit different lost packets from different receivers in a way that multiple receivers are able to recover their own lost packets with one transmission by the sender.

The main contribution of this work is the proposal of two efficient multicast retransmission algorithms based on network coding, one with multiple coded packets and another that additionaly assumes a buffer that stores network coded packets at the receivers, whose performance is analyzed and compared to that of other retransmission schemes.

This work is divided as follows: section II describes the scenario and the multicast quality-of-service metric, section III presents a detailed description and analysis of the considered algorithms, section IV describes the simulation model, section V shows and analyzes the numerical results and the last section presents the conclusions.

II. Considered scenario

This section describes the scenario and the multicast quality-of-service metrics. The system assumes a uniform spatial distribution of users within a single cell. This group of users are part of a multicast service where the transmission occurs in the downlink through the same radio resource and all users see a same average interference.

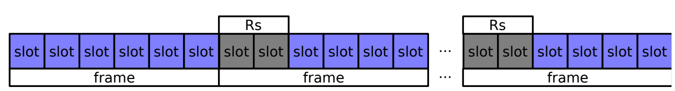


Fig. 1. The temporal structure of the system model.

Fig. 1 represents the temporal structure of the system model. The time was divided in frames and each frame in slots. It is assumed that a packet is transmitted at each slot. Within each frame a maximum number of retransmission slots per frame (R_s) is reserved for the retransmission of packets, with retransmissions starting to occur after the first frame. It is also assumed that at the beginning of each frame the base station has gathered the feedback from the users regarding which packets were not received correctly in the previous frame.

Another parameter is the retransmission criterion. We assume that a retransmission of a packet is allowed to occur only when a certain percentage of the users have not received it correctly, as it is done for example in [3]. This parameter is intended to reduce the delay of the system, since it is not worth utilizing a retransmission slot for only a few users. Let U_{T} denote the total number of users in the system and U_{R} the number of users who have not received a specific packet.

The retransmission criterion for a given packet is satisfied when $\alpha > \alpha_{\mathrm{th}}$, where $\alpha = \frac{U_{\mathrm{R}}}{U_{\mathrm{T}}}$ and α_{th} is the threshold required for retransmission. Note that this threshold allows to emulate the error tolerance of the service. For $\alpha_{\mathrm{th}} = 0$, it means that the service does not tolerate any errors, otherwise, if $\alpha_{\mathrm{th}} > 0$, the service tolerates the loss of some packets.

We consider a metric that indicates the multicast quality-of-service in terms of an efficiency η, which is given by

$$\eta = \frac{L_{\mathrm{success}}}{U_{\mathrm{T}} S_{\mathrm{tx}}}, \tag{1}$$

where L_{success} corresponds to the total number of correctly received packets of all users, U_{T} is the total number of users in the system, and S_{tx} is the effective total number of transmitted slots used.

III. Retransmission algorithms

This section describes the four different algorithms considered by the simulation analysis, which are namely: no retransmission (NRTX), simple retransmission (SRTX), network coded retransmission with the combination of multiple packets (NCRTXp, where p represents the number of combined packets) and network coded retransmission with 2 packets and buffering at the receivers (NCRTX2B). The algorithms described below differ in how they choose or combine the packets for retransmission.

A. No retransmission

The first algorithm is the simplest of all and has transmission with no retransmission. Thus, different packets are transmitted at each slot, resulting in a total number of packets equivalent to the number of frames times the number of slots per frame. Since there are no retransmissions, the calculation of d always has the value null because it simply takes into account all slots, i.e., the number of transmitted packets is the same as the total number of simulated slots.

B. Simple retransmission

The second algorithm corresponds to a simple retransmission. Thus, beginning from the second frame, a maximum number of retransmission slots per frame is reserved. Based on the feedback from the users, the base station computes for each transmitted packet how many users received it correctly.

Let $U_{i,j}$ denote the number of users that correctly received packet i in frame j. The packets are ordered according to the increasing value of $U_{i,j}$, i.e., priority is given to those packets that are received by the smallest amount of users. In frame $j+1$ the packets with the highest priority are retransmitted within the reserved retransmission slots, provided that a minimum tolerated percentage of users who have not received this packet is reached.

There is an exception when there are less packets to retransmit than the reserved amount of retransmission slots per frame. In this case, in order to avoid idle slots, these free retransmission slots are used for transmitting new packets.

C. Network coded retransmission with multiple coded packets

The third algorithm applies network coding to retransmit multiple packages in a single coded packet. Similarly to the previous algorithm, the same steps are done, but in frame $j+1$ coded packets are sent within the slots reserved for retransmission. The packets to be retransmitted are combined in pairs, trios, quartets and quintets. Let n denote the number of packets encoded. Within each frame a maximum of nR_s packets can therefore be retransmitted.

Let L_q denote the total number of packets queued for retransmission, where the packets are ordered according to the same priority scheme of the previous algorithm, and L_j the number of packets to be retransmitted within frame j. For this algorithm the following relationship holds: $L_j = \min\{L_q, nR_s\}$. The L_j packets with the highest priority are selected and they are combined by successively picking the head and tail of this queue of selected packets.

When L_j is not a multiple of n, the intermediary packets will be encoded and retransmitted, i.e., a coded packet will contain less than n packets. The actual packet encoding can be done by a simple logical operator such as XOR [6]. Note that the same exception of the simple retransmission algorithm with regard to free retransmission slots also holds.

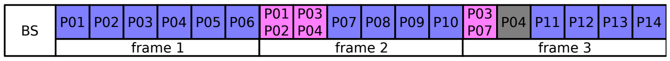
The packet combination is made as proposed because, in order to achieve gains with network coding, the user receiving the coded packets needs to already have correctly received at least $n-1$ of the packets in a previous transmission, so that it can decode the other one. According to the priority scheme, the packets at the head of the queue have been received by the least amount of users, so that if they were to be combined in order, the probability of a user having previously received at least $n-1$ of the packets would be quite low.

The proposed combination scheme is therefore a low-complexity algorithm that aims at increasing this probability, which has been verified to achieve better results than a random packet combination.

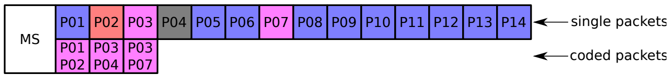
D. Network coded retransmission with buffering at the receivers

The last algorithm also considers the retransmission of network coded packets, but now using a buffer to store encoded packets in the mobile stations. This proposed algorithm aims at reducing the total number of slots used for retransmission. Similarly to the NCRTX2 algorithm, the same steps are done, but in frame $j+1$ the mobile stations store coded packets sent within the slots reserved for retransmission.

Upon receiving the retransmission of a single packet, the mobile stations will check the buffer of coded packets to detect whether some additional packet can be decoded. The mobile stations only need to store coded packets containing at least one packet that has not yet been decoded.



(a) Time structure of BS transmission.



(b) MS buffer.

Fig. 2. Example of NCRTX2B algorithm.

Figure 2 represents an example of the NCRTX2B algorithm. In this example, each frame was divided in six slots where two are reserved for retransmission. Initially the base station (BS) transmits the first six packets in the first frame, but the mobile station (MS) receives only the packets P01, P05 and P06.

In the second frame, two coded packets are retransmitted: the first containing the packets P01 and P02, and the second containing P03 and P04. Assuming MS receives both, MS stores these packets and the algorithm, through the first coded packet, can decode P02 in the same way as the NCRTX2 algorithm. The BS transmits the following four packets next, but MS receives only packets P08, P09 and P10.

Finally, in the third frame, packets P03 and P07 are retransmitted in encoded form, and packet P04 is retransmitted unencoded, since there are no further packets to be retransmitted at the moment. Assuming the MS receives both, it stores the coded packet and the algorithm, through the second coded packet and P04, can decode P03. Moreover, through the third coded packet and P03, the algorithm can decode P07.

IV. Simulation model

In the simulation model, all users have a different signal-to-interference-plus-noise-ratio (SINR) that depends on the additive white Gaussian noise (AWGN), the average interference power and the received signal power perceived by the users. The reception power depends on the transmission power, path loss, shadowing and gains of transmitting and receiving antennas.

The shadowing or slow fading can be caused when a large obstruction such as a hill or large building obscures the main signal path between the base station and the user. The amplitude change caused by shadowing was modeled using a log-normal distribution. The path loss is mainly a function of distance between the user and the base station. It was modeled according to [11] as follows:

$$PL = 120.9 + 37.6 \log(dist)[dB], \qquad (2)$$

where PL is the path loss and $dist$ the distance between user and base station.

In order to estimate the throughput achieved by the users, the SINR needs to be mapped to a certain packet error probability. We assume a simple mapping procedure consisting

TABLE I

List of considered system simulation parameters.

Parameter	Value	Unit
Cell radius	1	km
Number of users (U_{T})	20	-
Transmission power (P_{tx})	40	dBm
Transmission gain	4	dBi
Reception gain	2	dBi
Interference power	25	dBm
Shadowing standard deviation	6	-
Noise power	-110	dBm
Simulated frames	20	-
Slots per frame	10	-
Retransmission slots (R_s)	2	-
Retransmission threshold (α_{th})	5	%
Iterations	10,000	-

of two SINR thresholds, which determine the range of the packet error probability. The values associated to this mapping are modeled according to the following equation:

$$PEP = \begin{cases} 1\%, & \gamma > 30 \text{ dB}, \\ 10\%, & 0 \text{ dB} < \gamma \leq 30 \text{ dB}, \\ 99\%, & \gamma \leq 0 \text{ dB}, \end{cases} \qquad (3)$$

where PEP is the packet error probability and γ is the SINR value to be mapped.

Note that these values roughly characterize a typical scenario. They have been inspired on block error rate link-level results of systems such as HSDPA and LTE [12]. Even though the absolute results may change significantly depending on this mapping, it is expected that the relative behavior among the algorithms remains approximately the same.

V. Numerical results and analysis

A simulation tool was built to evaluate several different scenarios and algorithms concerning the provision of multicast services. The standard simulation parameters can be found in Table I. In order to quantify the performance of the considered algorithms, the following key parameters have been chosen for analyzing their behavior maintaining the values of the other parameters according to Table I:

1) Number of users within the cell;
2) Power transmitted by the base station;
3) Number of reserved retransmission slots per frame;
4) Number of slots per frame;
5) Retransmission threshold (α_{th}).

The first parameter was chosen in order to analyze the impact of an increased load on the multicast system. The variation of the second parameter refers to the convergence of the simulation results. The third parameter has an impact on the radio link quality. Finally, the variation of the last parameters provide some insights on how to properly adjust these aspects of the transmission protocol.

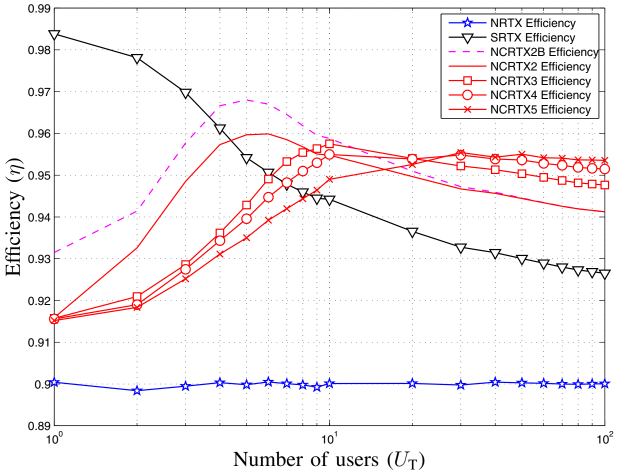


Fig. 3. Impact of the number of users (U_T) on the efficiency (η).

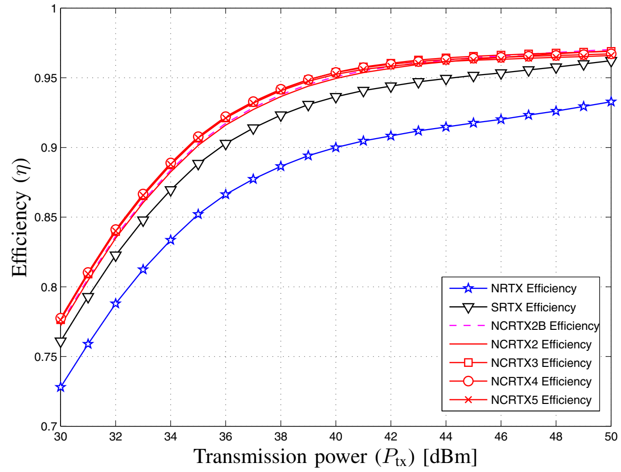


Fig. 4. Impact of the transmission power (P_{tx}) on the efficiency (η).

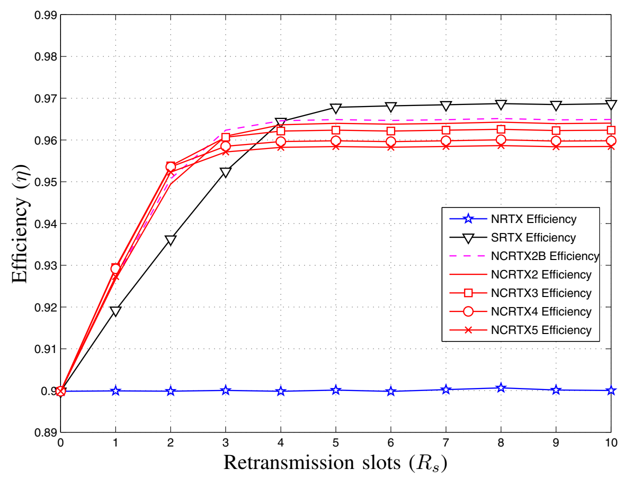


Fig. 5. Impact of the number of reserved retransmission slots (R_s) on the efficiency (η).

Note that the NRTX algorithm, which has no retransmissions and is shown for comparison purposes, corresponds to a worst-case in terms of the efficiency (η). It converges to the average packet success probability, which depends on the mapping given by (3). The only exception is the variation of the transmission power parameter, which has a direct impact on the perceived SINR.

The NCRTX2B algorithm, which has retransmission with a buffer and is the proposed algorithm, always has a better performance, or at least the same, compared to NCRTX2. This happens because the same steps are done, except the storage of coded packets in NCRTX2B. Thus, performance can be improved with the decoding of these packets.

Fig. 3 represents the graphic of the efficiency as a function of the number of users. For the SRTX algorithm, the higher the system load the lower the efficiency. Since the number of retransmission slots is fixed, the increase in the number of users ends up overloading the system, i.e., there are not enough resources to retransmit all packets.

With regard to the NCRTX2B algorithm, its efficiency is greater than the SRTX algorithm for more than 3 users. In return, it is surpassed by the algorithms with multiple packets encoded for more than 10 users. This is because, in the scheme of network coding with a buffer, the decoding of packets requires that the users should already know at least one of the packets in order to decode the other. The NCRTX2B algorithm highlights NCRTX2 for few users, in this way, these algorithms converge for many users in the system.

Also, the algorithms with multiple encoded packets can retransmit more different packets according to the need of the multicast group. Moreover, the probability that the same user does not have any of the coded packets is very high for less than 4 users in the system. As soon as the number of users is enough to make up for this coding loss it follows the same behavior of SRTX, i.e., efficiency decreases with more users.

Fig. 4 shows the impact of the transmission power. Since the interference power is constant at 25 dBm, increasing the transmit power is always beneficial for the users. The achieved results of the efficiency are directly related to (3).

Fig. 5 illustrates the efficiency as a function of the maximum number of reserved retransmission slots per frame. Note that R_s has been simulated up to the the number of slots per frame. So, it is possible to have more slots used for retransmission than for the actual transmission, i.e., in this case the delay is dramatically increased.

It can be seen that the efficiency of the SRTX algorithm increases linearly up to four R_s while other algorithms up to two R_s. As for the efficiency of the algorithms, it converges with the number of Rs. This means that this number of Rs is already enough to accommodate all retransmissions, thus not requiring the combining of packets in order to make them fit into the available number of retransmission slots.

The impact of the number of slots per frame, with R_s fixed, is shown in Fig. 6. The x-axis starts from 1 slot, but only in this case, the maximum number of retransmission slots per frame is limited by the size of the frame. For a large number of

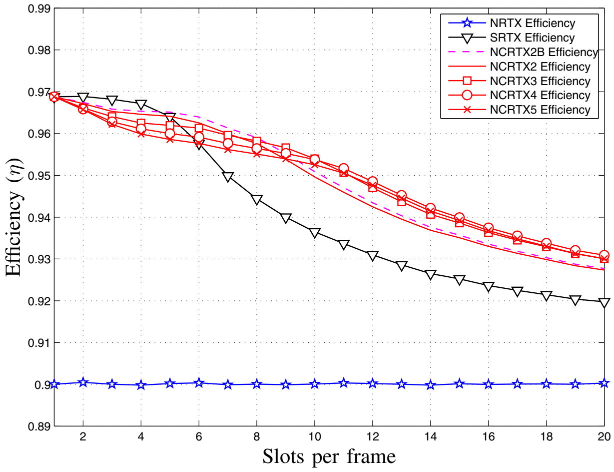


Fig. 6. Impact of the number of slots per frame on the efficiency (η).

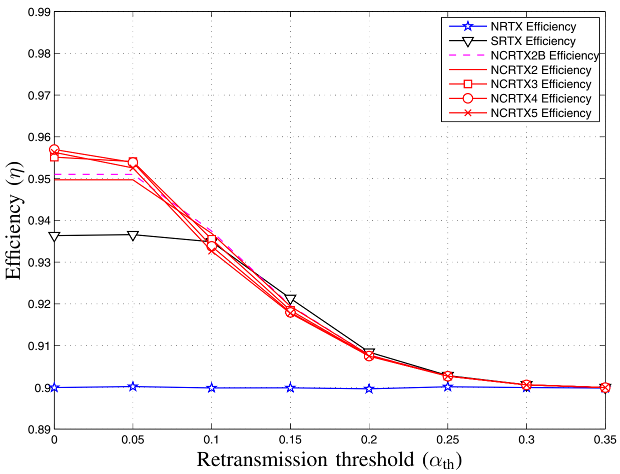


Fig. 7. Impact of the error tolerance on the efficiency (η).

slots all algorithms approach the case without retransmission, since R_s is fixed. The achieved results are straightforward and confirm the importance of properly reserving slots dedicated for retransmission.

Finally, the last results are shown in Fig. 7, representing the efficiency as a function of the tolerance. Since the efficiency metric depends on the number of successfuly received packets, increasing the error tolerance ends up decreasing the efficiency. It can be seen that the network coding strategies are more adequate to error-intolerant services, since for thresholds higher than 0.1 their performance converges to that of SRTX.

VI. Conclusions

In this paper, four types of multicast transmission schemes are analyzed and an algorithm based on network coding with a buffer is proposed. The proposed algorithm aims at using more efficiently the radio resources available for retransmission and it is shown to provide the best performance and significantly improve the efficiency in almost all cases, with the exception of rather small multicast group sizes. The impact of several system parameters have also been analyzed, in order to provide a detailed comparison among the algorithms.

Gains of roughly 1.5% with regard to the case with simple retransmission and 5% with regard to the no retransmission are achieved by the two proposed algorithms, taking into account the standard simulation parameters in Table I. Note that higher gains are expected for larger multicast group sizes as well as for scenarios with less robust transmission schemes, i.e., with more pessimistic SINR to PEP mapping.

References

[1] M. Bakhuizen and U. Horn, "Mobile broadcast/multicast in mobile networks," *Ericsson Review*, issue no. 01/2005, May 2005.

[2] 3GPP, "Multimedia Broadcast/Multicast service (MBMS); architecture and functional description," Technical Specification Group Services and System Aspects, Tech. Rep. TS 23.246 v6.10.0, Jun. 2006.

[3] M. Rossi, M. Zorzi, and F. H. P. Fitzek, "Investigation of link layer algorithms and play-out buffer requirements for efficient multicast services in 3G cellular systems," in *Proc. IEEE PIMRC*, vol. 3, Sep. 2004, pp. 2256–2261.

[4] W. Chen, K. B. Letaief, and Z. Cao, "Opportunistic network coding for wireless networks," in *Proc. IEEE ICC*, Jun. 2007, pp. 4634–4639.

[5] H. Yomo and P. Popovski, "Opportunistic scheduling for wireless network coding," in *Proc. IEEE ICC*, Jun. 2007, pp. 5610–5615.

[6] S. Katti, H. Rahul, W. Hu, D. Katabi, M. Médard, and J. Crowcroft, "XORs in the air: Pratical wireless network coding," *IEEE/ACM Trans. Networking*, vol. 16, no. 3, pp. 497–510, Jun. 2008.

[7] C. Fragouli, D. Katabi, A. Markopoulou, M. Médard, and H. Rahul, "Wireless network coding: Opportunities & challenges," in *Proc. IEEE MILCOM*, Oct. 2007, pp. 1–8.

[8] D. Nguyen, T. Tran, T. Nguyen, and B. Bose, "Wireless broadcast using network coding," *IEEE Transactions on Vehicular Technology*, vol. 58, no. 2, pp. 914–925, Feb. 2009.

[9] 3GPP, "Applications of network coding in LTE-A," Technical Specification Group Radio Access Networks, Tech. Rep. R1 084130, Nov. 2008.

[10] ——, "Optional retransmission schemes for LTE-A MBMS," Technical Specification Group Radio Access Networks, Tech. Rep. R1 084132, Nov. 2008.

[11] ——, "Physical layer aspects for evolved universal terrestrial radio access (UTRA)," Third Generation Partnership Project, Tech. Rep. TR 25.814 V7.1.0, Sep. 2006.

[12] ——, "LTE physical layer - general description," Third Generation Partnership Project, Tech. Rep. TR 36.201 V8.3.0, Mar. 2009.