Analysis of network coded retransmission techniques for wireless multicast links

Pedro R. S. Lopes, Yuri C. B. Silva and Francisco R. P. Cavalcanti

***Resumo*— Este trabalho avalia o desempenho de três algoritmos para transmissão *multicast* em redes de comunicações sem fio considerando uma única célula. Uma técnica eficiente de retransmissão para serviços *multicast* baseada em codificação de rede é proposta e comparada a outros esquemas comuns de retransmissão, como o caso sem retransmissão e um esquema de retransmissão simples. É demonstrado que o algoritmo proposto possui um desempenho superior às outras técnicas e que a qualidade da transmissão *multicast* é melhorada.**

***Palavras-Chave*— Codificação de rede, transmissão *multicast*.**

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

***Keywords*— Network coding, multicast transmission.**

I. INTRODUCTION

In recent years the area of telecommunications has been the focus of many research and technological advances and, in this field, the area of wireless networks was the one that presented the most impressive advances. And by allowing connection to telecommunications systems, these networks have become a promising field with prospects of even greater future use.

Among the wireless technologies, the mobile telephone was the one that evolved the most and for which a great deal of applications and features have been developed. The mobile telephone systems offer the most advanced types of services and also support several types of traffic.

Advanced wireless systems such as the UTRAN Advanced Long-Term Evolution (LTE-Advanced) in 3GPP [1], [2], [3], [4], [5], have performance requirements in the physical layer that are difficult to reach with transceivers of conventional access technologies. In particular, in terms of spectral efficiency, a notable increase is expected when compared to 3G systems. In order to provide this increased spectral efficiency, efficient radio resource management techniques must be implemented. In the case of multicast services, network coding can be employed to improve the overall throughput.

Wireless multicast [6], [7], [8] is an important service with applications to file distribution, delay tolerant networks, home entertainment, and video conferencing. As more wireless networks are deployed on city-wide scales, and as mobile wireless devices continue to replace their fixed wired counterparts, this importance will increase.

Through wireless multicast transmission 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 timeslot, 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 [9], which defines resource allocation and transmission procedures specific for multicast.

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.

In order to improve the multicast performance, we propose to employ network coding [10], [11], [12], [13], which is a new area of networking, in which data is manipulated inside the network to increase the received data throughput, reduce delay, and improve robustness. This field has recently found commercial applications in content distribution, peer-to-peer design, and enabling high-throughput wireless networks.

More generally, network coding combines packets before transmitting them and has two important benefits relevant to multicast routing. First, it decreases the number of transmissions necessary to route packets to multiple receivers for both multi-hop and single-hop routing. Second, it reduces the need for coordination among network nodes in multi-hop routing.

This service 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 two different packets in a single resource, which are coded into a single packet.

The main contribution of this paper is the proposal of an

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efficient multicast retransmission algorithm based on network coding, whose performance is analyzed and compared to that of other retransmission schemes. This paper is divided as follows: section II describes the system model, section III presents a detailed description and analysis of the considered algorithms, section IV shows and analyzes the numerical results and the last section presents the conclusions and perspectives for future works.

II. SYSTEM SIMULATION MODEL

In complex wireless communication systems, the computer simulation appears as an effective analysis tool for evaluating system performance and proposing improvements in its major functions and architectural aspects. However, this approach requires the selection of models capable of providing detailed information about the system, as well as the proper selection of software tools enabling the effective use of computer resources.

TABLE I

List of considered simulation parameters.

Parameter	Value
Cell radius	1 km
Number of users (U_{T})	10
Transmission power	50 dBm
Transmission gain	4 dB
Reception gain	2 dB
Noise power	-110 dBm
Simulated frames	10
Slots per frame	6
Retransmission slots (R_s)	2
Iterations	10,000

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, where the transmission and reception gains refer to the gains of transmitting and receiving antennas, respectively.

The simulated system assumes a uniform spatial distribution of users within a single cell. All users have a different signal-to-noise-ratio (SNR) that depends on the additive white Gaussian noise (AWGN) and the received signal power perceived by the users. The reception power depends on the transmission power, transmission gain, path loss and reception gain. The path loss is mainly a function of distance between the user and the base station. It was modelled according to [14] as follows:

$$\dot{P}L = 120.9 + 37.6 \log(d), \tag{1}$$

where *PL* is the path loss and *d* the distance between user and base station.

In order to estimate the throughput achieved by the users, the SNR needs to be mapped to a certain packet error probability. We assume a simple mapping procedure consisting of two SNR thresholds, which determine the range of the packet error probability. The values associated to this mapping are modelled according to the following equation:

$$PEP = \begin{cases} 5\%, & \gamma > 70 \text{ dBm}, \\ 10\%, & 50 \text{ dBm} < \gamma \leq 70 \text{ dBm}, \\ 15\%, & \gamma \leq 50 \text{ dBm}, \end{cases} \tag{2}$$

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

Note that these values were obtained experimentaly and characterize a typical scenario. 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.

III. RETRANSMISSION ALGORITHMS

This section describes the three different algorithms considered by the simulation analysis, which are namely: no retransmission, simple retransmission, and network coded retransmission. 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 certain number of slots 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. The algorithms described below differ in how they choose or combine the packets for retransmission.

All graphics in the results session have three curves that quantify the amount of succesfully received packets of the three algorithms in terms of a specific parameter. This metric indicates the multicast quality-of-service in terms of a transmission efficiency η, which is given by:

$$\eta = \frac{L_{\text{success}}}{U_{\mathrm{T}} L_{\text{tx}}}, \tag{3}$$

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 L_{tx} is the total number of transmitted packets (not counting retransmissions).

A. Transmission with no retransmission

The first algorithm is the simplest of all and has 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 η 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 (R_s) is reserved. Based on the feedback by 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 transmitted within the reserved retransmission slots.

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. Retransmission for network coding

The third and last algorithm has retransmission, but in encoded form. This is the proposed algorithm that reduces the total number of slots used for retransmission. 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. Within each frame a maximum of $2R_s$ 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: $L_j = min\{L_q, 2R_s\}$. 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, i.e., pairs are formed with packet indices $\{1, L_j\}$, $\{2, L_j - 1\}$, $\{3, L_j - 2\}$, and so on.

When L_j is odd, the intermediary packet will be retransmitted in a simple way. The actual packet encoding can be done by a simple logical operator such as XOR. 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 one 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 one of the packets would be quite low. The proposed combination scheme is therefore a low-complexity algorithm that aims at increasing this probability.

IV. Numerical results and analysis

In order to quantify the performance of the three considered algorithms, the following key parameters have been chosen for analyzing their behavior:

1) Number of users within the cell,
2) Total number of simulated frames,
3) Transmission power,
4) Number of reserved retransmission slots per frame (with fixed number of slots per frame),

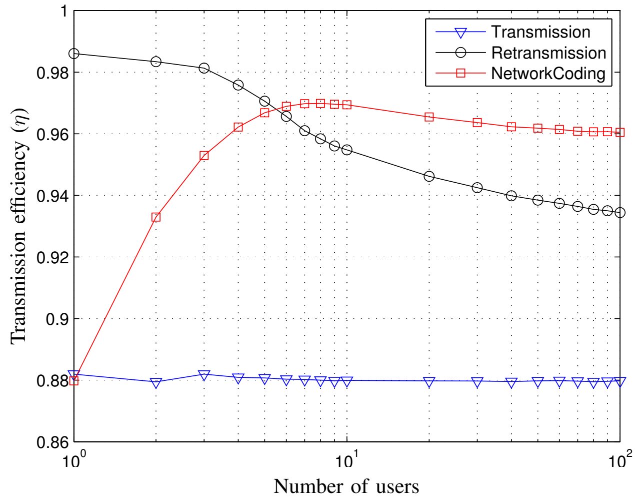

Fig. 1. Impact of the number of users.

5) Number of slots per frame (with fixed number of reserved retransmission slots per frame).
6) Number of reserved retransmission slots per frame (maintaining the same proportion with regard to the number of slots per frame),

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

Note that the "Transmission" algorithm, which has no retransmissions, is shown as a worst-case for comparison. This algorithm presents the same efficiency regardless of the variation of all key parameters, since the efficiency metric converges to the average packet success probability, which depends on the mapping given by (2). The only exception is the variation of the transmission power parameter, which has a direct impact on the perceived SINR.

Figure 1 represents the graphic of the transmission efficiency as a function of the number of users. For the "Retransmission" algorithm, the higher the system load the lower the transmission 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 "Network Coding" algorithm, its transmission efficiency is the highest only for more than 6 users. This is because, in the network coding scheme, the decoding of packets requires that the users should already know at least one of the packets in order to decode the other. Moreover, the probability that the same user does not have any of the coded packets is very high for less than 5 users in the system. As soon as the number of users is enough to make up for this coding loss, which happens at roughly 10 users, it follows the same behavior of the "Retransmission" algorithm, i.e., efficiency decreases with more users.

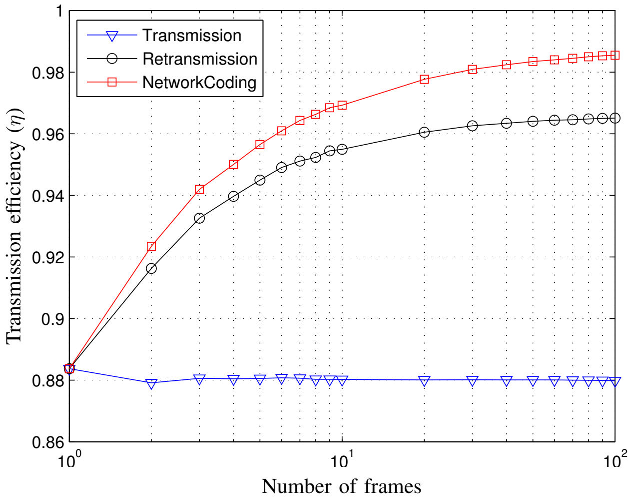

Fig. 2. Impact of the total number of simulated frames.

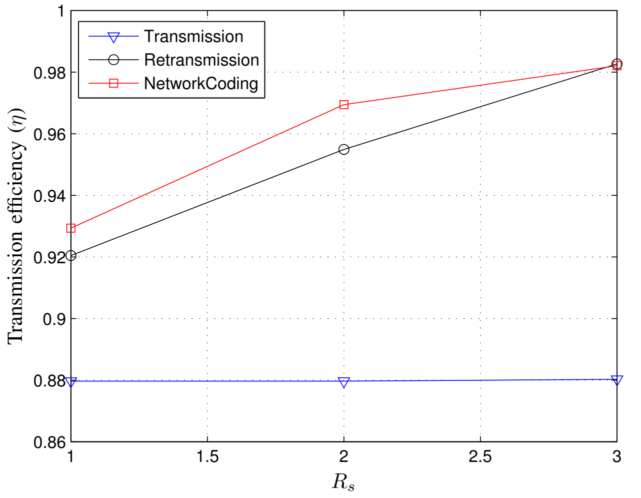

Fig. 4. Impact of the number of reserved retransmission slots per frame.

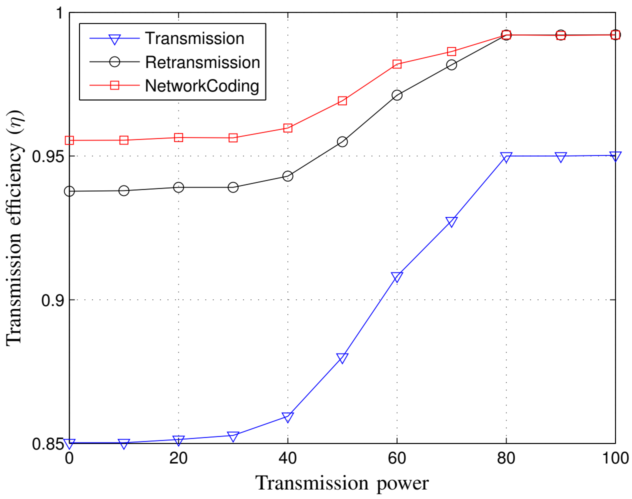

Fig. 3. Impact of the transmission power.

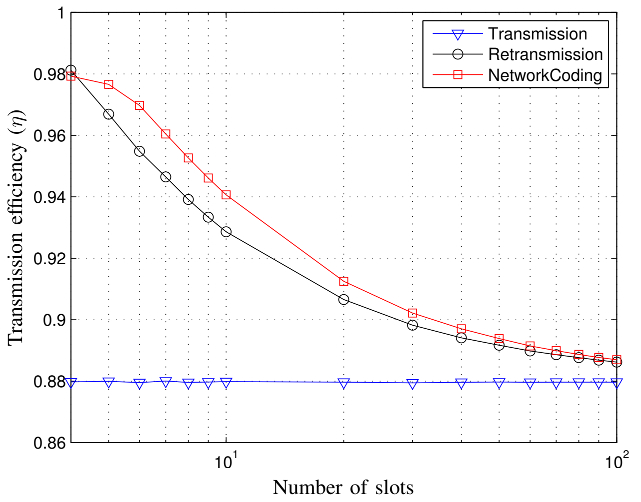

Fig. 5. Impact of the number of slots per frame.

The impact of the number of frames on the efficiency of the algorithms is shown in Fig. 2. When there is only one frame, all algorithms have the same efficiency, since no retransmissions take place. The efficiency increases with the number of frames, since the impact of the first frame without retransmission tends to be minimized, converging to a certain value for each algorithm. The relative performance among the algorithms is the same as that of Fig. 1 for the case with 10 users, due to the previously explained reasons. Note that as convergence is achieved, the advantage of network coding over the other schemes becomes more evident.

Fig. 3 shows the impact of the transmission power. Since there is no interference, increasing the transmit power is always beneficial for the users. The achieved results are directly related to (2), which explains the saturation for low and high transmit powers. It can be seen that from the upper saturation level of 80 dBm network coding no longer presents gains with regard to the simple retransmission scheme.

Figure 4 illustrates the efficiency as a function of the maximum number of reserved retransmission slots (R_s) per frame. Note that R_s has only been simulated up to the half of the number slots per frame, since it is not reasonable to have more slots reserved for retransmission than for the actual transmission, as the throughput would be drastically reduced. It can be seen that the efficiency of the "Retransmission" algorithm increases linearly with R_s. As for the "Network Coding" algorithm, it increases up to $R_s = 2$, but for $R_s = 3$ it converges to the "Retransmission" algorithm. This means that $R_s = 3$ is already enough to accomodate 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. 5. The x-axis starts from 4 slots, in order to have at least half the number of slots dedicated for transmission. For a large number of 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 for retransmission.

Finally, the last results are shown in Fig. 6, representing the efficiency as a function of R_s, but with the same proportion between R_s and the number of slots per frame. As in the standard simulation, R_s is 2 and the number of slots per frame is 6. This proportion corresponds to the third part of the slots. It can be seen that there is a slight transient for small frame

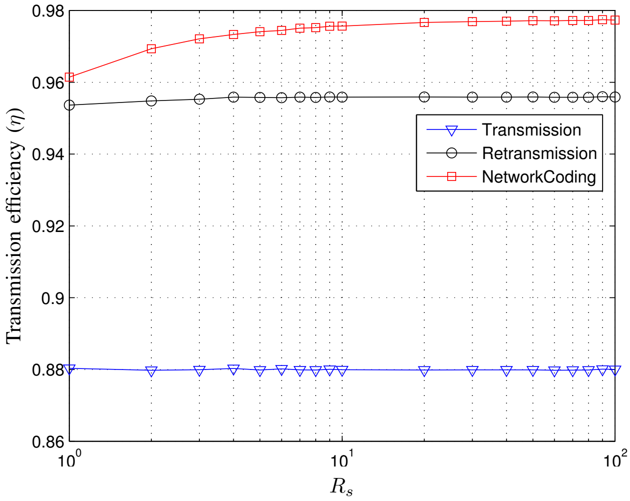

Fig. 6. Impact of the number of reserved retransmission slots per frame, but maintaining the same proportion of the number of reserved retransmission slots per frame.

sizes, but the efficiency of the algorithms rapidly converges to different values.

V. Conclusions

In this article, three types of multicast transmission schemes are analyzed and an algorithm based on network coding 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 improving the transmission 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.

An interesting extension of the current work is to extend the analysis to multi-cell scenarios, in which interference might have an impact on the design of the algorithms. Relay stations with network coding and multicast services is also another related subject that might profit from efficient retransmission algorithms.

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