Reliable video-streaming delivery in wireless access networks using Network Coding

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Abstract: Providing real-time streaming services such as Internet Protocol television (IPTV) over last mile wireless networks is challenging due to high bandwidth demands, scarce radio resources and lossy characteristics of wireless networks. This paper presents wireless system for reliable video-streaming over wireless networks based on advanced retransmission scheme. The concept embraces characteristics of wireless media and is based on Network Coding (NC) in order to improve bandwidth utilization in terms of bandwidth reduction. Bandwidth reduction is for Internet Service Providers (ISPs) of paramount importance as it means additional clients over the same infrastructure or, alternatively, additional services that can be offered. Results from practical testbed show that the number of retransmissions can be significantly reduced i.e. total bandwidth can be reduced for up to 15 % assuring the same service.

Key-Words: - Retransmission, wireless network, broadcast, network coding, streaming, IPTV

1 Introduction

The provision of high quality streaming services over wireless systems with increased demands for scarce radio resources, introduces never-ending challenges of efficient delivery of streaming video content, in particular when wireless technology (e.g. Wi-Fi) is used in the last mile. Wireless IEEE 802.11 networks, where one antenna covers several different clients / groups of clients, are often used due to its ease and fast deployment and relatively low deployment costs, especially in underdeveloped countries and rural areas.

However, spectrum limitations in wireless medium shift the bottleneck in content delivery to the last mile. Increasing the last mile access performance in such cases is for Internet Service Providers (ISPs) of paramount importance as bandwidth reduction means additional clients over the same infrastructure or, alternatively, additional services that can be offered.

In this article we are interested in the efficient and quality delivery of video-streaming services such as Internet Protocol Television (IPTV) in the last mile over the IEEE 802.11 wireless network.

In general, video content can be delivered over IEEE 802.11 wireless network using broadcast / multicast or unicast mechanism. The unicast mechanism delivers the content to users individually and supports retransmissions (and back-off) that assure reliable content delivery and is thus preferred solution in currently deployed systems. Instead of unicasting multiple streams of video content, a goal of broadcasting and multicasting is to reduce bandwidth. Broadcast reduces bandwidth when multiple users watch the same video stream, since the server has to send only one packet, instead of multiple packets to different clients. However, broadcast mechanism does not assure reliable content delivery. To support reliability and reduce bandwidth different retransmission schemes have been proposed, where some also consider Network Coding (NC).

Since the pioneering work of NC [1] numerous papers (i.e. [2] – [14]) appeared on this subject and significant progress has been made in applying NC to different networks. For example NC is increasing throughput in satellite [2] and P2P networks [3], improving delivery reliability over the lossy links either in wireless networks over TCP [4] or in Delay Tolerant Networks such as deep space links [5]. Depending on the application of NC the implementation affects different OSI layers. In multicast scenarios NC is typically implemented in the application layer while two stage NC for increased spectrum efficiency is deployed in the physical layer [6].

NC concept can be used in the retransmission scheme to reduce bandwidth consumption. Let us illustrate NC using elementary examples from Fig.1 and Fig.2. Consider two clients and a broadcasting server that had transmitted packets A and B. Assume that Client 1 has not successfully received packet A and Client 2 has not successfully received packet B. Instead of retransmitting packets A and B separately as in Fig.1, the server codes packets A and B together using linear operation (e.g. XOR) into a single packet and retransmits the coded packet as it is depicted in Fig.2. Clients can now obtain their lost packets by performing decoding operation using packets that they already have stored in their packet pools.

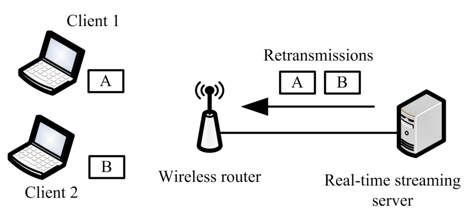

Fig.1. Retransmissions without NC

(E.g. Client 1 can decode coded packet using packet B from its pool and thus obtain packet A. In the same way Client 2 can obtain packet B).

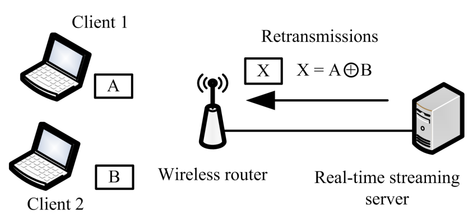

Fig.2. Retransmission with NC

By applying the NC mechanism and code packets together bandwidth reduction can be achieved.

NC has already been successfully used to increase throughput as show in deployment in [7]. They increased throughput over a broadcast medium, by mixing packets from different flows into a single packet. Delivery of multiple video streams over multihop networks using opportunistic NC scheme is proposed in [8]. In [9] delivering single video stream to multiple users in hyper dense Wi-Fi networks using NC is presented. However, they consider problem of satisfying different receivers requests regarding the same flow. Also [10] considers video streaming of a single flow to multiple receivers. Moreover in [10] retransmissions are not included but random linear codes that incorporate redundancy already when transmitting the packets for the first time. Retransmission schemes based on NC are discussed in [11], [12] and [13] but do not consider video-streaming applications requirements.

We are interested in the retransmission scheme that is integrated in the application layer and is adapted for video-streaming applications such as IPTV. In this paper we propose approach for reliable video- streaming delivery. We also evaluate its performance using practical Wi-Fi testbed, which comprise streaming server and multiple Wi-Fi clients.

The paper is structured as follows. Section 2 proposes concept of reliable video-streaming retransmission scheme. Section 3 gives short overview about implementation details. Section 4 describes performance metrics and results. And finally, Section 5 concludes the paper.

2 Concept of video-streaming retransmission scheme using NC

Consider the system setup illustrated in Fig.3. The system is intended for transferring multiple video streams to multiple clients. In depicted example streaming server is streaming two channels (streams A and B) to two clients (Client 1 and Client 2). Wireless router broadcasts the content over the wireless media. All clients listen to all transmissions and store all the received packets (even the ones not intended for them).

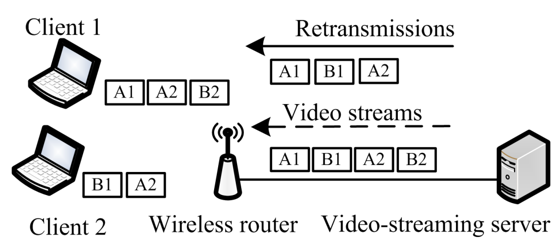

Fig.3. Traditional retransmission approach

System from Fig.4 has the same setup. Dissimilarity of the two systems is reflected in the way of implementation of retransmissions. System from Fig.3 is using traditional retransmission approach where every lost packet is retransmitted separately (three packets are retransmitted) while system from Fig.4 is using retransmission scheme with NC (only two packets are retransmitted).

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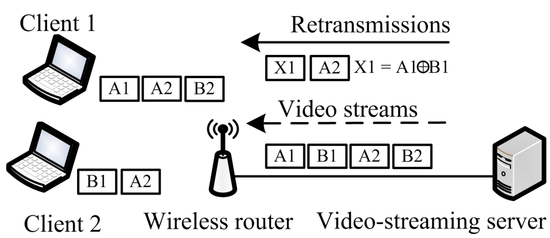

Fig.4. Retransmission approach using NC

Given the described system from Fig.4 the design parameters for development for the retransmission scheme using NC were:

1. Only packets that require retransmission will be retransmitted
2. The algorithm retransmits packets as late as possible, but still assuring the Quality of Service (QoS), i.e. after the Retransmission Timeout (RTO) or opportunistically.
3. All coded packets must be decodable by all clients
4. Packets that have not been received by any client will be sent out as they are, i.e. not coded.

The streaming server records the status of the received and not received packet for each of the clients. Information about the not received packets is provided through the Negative Acknowledgement (NACK) packets. Current status of the received packets at individual clients is represented in a transition table depicted in Table 1.a. Transition table has M rows that correspond to packets and N columns that correspond to the clients.

Table 1. Transition and coding tables

a) Transition table

	C_1	C_2
A_1	0	1
B_1	1	0
A_2	0	2
B_2	2	1

b) Coding table

	C_1	C_2
A_1	0	1
B_1	1	0
A_2	0	0

Packets with the lowest index have been represented in the table for the longest period. Three different states are noted in the transition table:

1. Packet received successfully is represented with state 1.
2. Not received packet intended for the corresponding client is represented with state 0 and means that packet has to be retransmitted.
3. Not received packet not intended for the corresponding client is represented with state 2 and means that packet does not have to be retransmitted for this client.

For the retransmission and coding process transition table is transcript into the coding table (i.e. Table 1.b) where only packets that require retransmissions are represented. Here, two states are used to describe the status of the received packets on the clients. State 0 indicates that a packet has not been received and state 1 indicates that packet has been received on the client. Moreover, packets that do not need to be retransmitted are not presented in the coding table such as the case for packet B_2 (packet has not been received by the C_1 but it does not require stream B so the packet has not have to be retransmitted).

Algorithm for making decisions on which packets to code together is presented in Alg.1. The algorithm is called, if there are packets for retransmissions (i.e. there are packets that exceed the RTO).

Algorithm 1. *NC Algorithm*

```
while (packets for retransmission)
    number_of_coded_packets = 1;
    if (P1 exceed RTO && P1 is codable)
        k = 1;
        coded_packets [k = &P1;
        for (m = 2; m <= M; m++)
            packets_codable = 1;
            for (n = 1; n <= N ; n++)
                codable = coding table [1, n] +
                        + coding_table [m, n;
                if (codable < number_of_coded_packets)
                    packets_codable = 0;
                    break;
                end if
            end for
            if (packets_codable)
                number_of_coded_packets++;
                k ++;
                coded_packets [k] = &Pm
            end if
            if (number_of_coded_packets == N)
                break;
            end if
        end for
    end if
    if (number_of_coded_packets > 1)
        encode packets from coded_packets
        sent encoded_packet;
    else
        sent P1 uncoded;
    end if
    update coding_table;
end while
```

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The algorithm takes the status of the first packet from the coding table (that is first row) and checks if it is codable. Packet is considered codable, if it has been received at least by one of the clients.

In the case that the packet is not codable the packet is retransmitted as is. If the packet is codable the algorithm searches for coding opportunities with the rest of the packets intended for retransmission (even the ones that have not reached the RTO).

Algorithm looks to code with the first packet as many packets as possible, but prioritizes packets that have been waiting in the retransmission queue for a longer period. If it finds two packets codable, it will try to find the third one, than the fourth one etc. After algorithm is done, coded packet will be broadcasted to all clients.

Algorithm considers two packets codable if coded packet can be decoded by all the clients. In practice this means that if all sums over the corresponding packets columns from the coding table are greater than or equal to 1. To generalize N packets are codable if all the sums over the corresponding packet columns is larger or equal than N – 1. As shown in [7] client can decode coded packet if it has previously received at least N – 1 native packets coded in the encoded packet.

Let us also use example from the Table 1.b to explain the algorithm in practice. Algorithm checks the status of the first packet A_1 using the coding table. Assume that RTO has expired for this packet. Packet is codable, since it was received by one of the clients. Then, algorithm checks status of the second packet B_1 with the help of the coding table and looks if the two are codable. Referencing to the coding rule these packets are codable. Since the two packets is the maximum packets we can code when only two clients are present the algorithm stops the coding procedure. If there were more clients, the algorithm would look for other coding opportunities trying to code more packets together. Packets A_1 and B_1 are coded together and sent out in one retransmission.

Furthermore, when RTO is reached for packet A_2 it will be sent out as is, as it had not been received by any of the clients. In a given example two retransmissions are required with the use of NC in contrast to the traditional approach where server would need three retransmissions. Even higher gains can be obtained in cases with more clients.

3 Implementation details

In order to make proposed scheme deployable, several supporting mechanisms have been implemented on the client and on the server.

Signalization mechanism in form of NACK packets is deployed at the client side. With NACK packets clients inform server about packets they did not receive. Also two more packet types are used in proposed scheme: native packet and coded packet.

Besides sending NACK packets, clients listen to all the transmissions even the ones not intended for them and stores all packets in the packet pool, for decoding purposes. With every received packet the client checks if it is native or coded packet. In the case when a coded packet is received, the process checks the packet pool where all the received packets are stored for decoding purposes. If the client has enough information, it decodes the coded packet using previously stored packets with the XOR operations, thus gaining a native packet that has not been received before [14].

At the other side supporting mechanisms at the server side are handling NACK packets and broadcasting native / coded packets. For every NACK that server receives updates in transition and coding table are made. Moreover, after broadcasting native packet server searches for coding opportunities in the coding table. After coding opportunity is found codable packets are coded together and broadcasted as a one packet to all clients.

In addition, NC brings an additional overhead into network in terms of additional packets and headers added to the regular packets.

4 Performance evaluation

4.1 Experimental setup

Practical wireless testbed deployed for proposed scheme consist of streaming server, wireless router and 7 clients (i.e. laptops). Clients were arranged arbitrarily around wireless router (within the same room) and their positions were fixed throughout the entire experiment. Wireless router used was Linksys WRT54GL with DD-WRT firmware and fixed wireless parameters as in Table 2.

Table 2. Wireless router configuration

Parameter	Value	Comment
Wireless Mode	AP	Access Point
Wireless Network Mode	G-Only	802.11 g
Wireless Channel	5-2.432 Ghz	Free channel

Server was connected to the wireless router via Ethernet interface while clients (laptops of different brands) were connected via WLAN interfaces. Server streamed one UDP stream in total 608 MB of data at 1.525 Mbps rate. Packets size was constant (i.e. 1210 bytes). To all outgoing packets retransmission scheme related headers were added. Inter-arrival time between packets was constant (i.e. 0.6 s).

Results were gathered through a one hour experiment and are presented for the time interval between 1000 and 1500th second to observe steady state condition.

Parameters used for the proposed testbed at server and client sides are presented in the Table 3.

Table 3. Parameters as used in testbed at server and client side

	Parametar	Value	Parametar Full Name
Server side	RTO	300 ms	Retransmission timeout
	T_i	6 ms	Packet inter-arrival time
	T_{THR}	300 ms	Threshold time
Client side	T_i	6 ms	Packet inter-arrival time
	T_{THR}	150 ms	Threshold time

4.2 Performance metrics

Several performance metrics were used to evaluate the performance of the proposed scheme:

1. Number of retransmitted packets N_T is the number of native packets that are retransmitted with no coding while number of retransmitted packets N_R is the number of packets that are retransmitted with coding.

2. N_N is the number of native packets sent.

3. Bandwidth reduction B_R is calculated as the proportion of difference of the N_T and N_R packets to the sum of N_N and N_T packets. We use B_R to show how much bandwidth in percentage we reduced by using the proposed scheme.

$$B_R = \frac{N_T - N_R}{N_N + N_T} * 100\% \quad (1)$$

4. Delivery probability of i-th client is defined as the proportion of successfully delivered packets N_{Di} to the total packets that were transmitted N_{TOi} in percentages.

$$DP_i = \frac{N_{D_i}}{N_{TO_i}} * 100\% \quad (2)$$

Delivery probability DP is average of delivery probabilities over single clients (N_C).

$$DP = \frac{1}{N_C} \sum_{i=1}^{N_C} DP_i \quad (3)$$

5. In the evaluated scenarios we primarily observed gain G_i which is on i-*th* client defined as the proportion of transmitted packets N_{Ti} to the retransmitted packets N_{Ri}.

$$G_i = 1 - \frac{N_{T_i}}{N_{R_i}} * 100\% \quad (4)$$

Gain G is average of gains of single clients (N_C).

$$G = \frac{1}{N_C} \sum_{i=1}^{N_C} G_i \quad (5)$$

6. With coding table size *CTS* we refer to the number of packet statuses on the clients. It is sampled periodically.

All the presented results were observed over the sample period T_S (i.e. *5 s*).

4.3 Results

Bandwidth reduction B_R of proposed scheme is shown in Fig.8. We can see that during the given time interval B_R ranges from 7 % to 14.75 %.

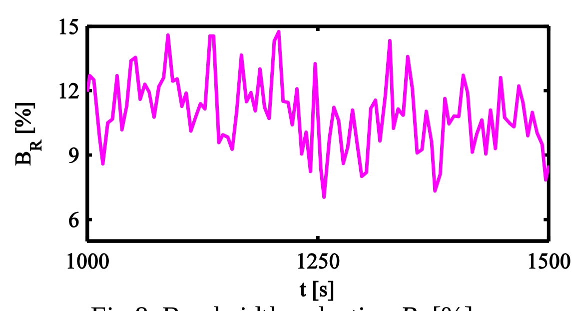

Fig.8. Bandwidth reduction B_R [%]

Using Fig.9 we explain in more detail dependencies between *DP*, *CTS* and *G*.

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As the *DP* decreases coding table size *CTS* increases i.e. number of packets in the coding table increases. This is because more NACKs are sent by clients. Otherwise, as the *DP* decreases, so does the coding gain.

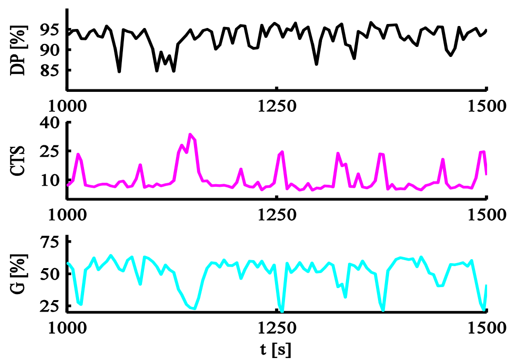

Fig.9. Dependencies of delivery probability *DP* [%], coding table size *CTS* and Gain *G* [%]

When *DP* is low there are more occasions where the same packet is not received by multiple clients. This affects the number of coding opportunities i.e. there are fewer, and gain is lower.

In the following, we also show relations between average number of native packets coded together and number of retransmissions. Fig.10 shows number of average native packets that are coded together while Fig.11 shows number of retransmissions.

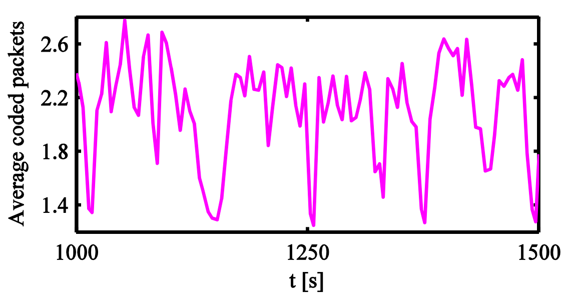

Fig.10. Average number of coded packets

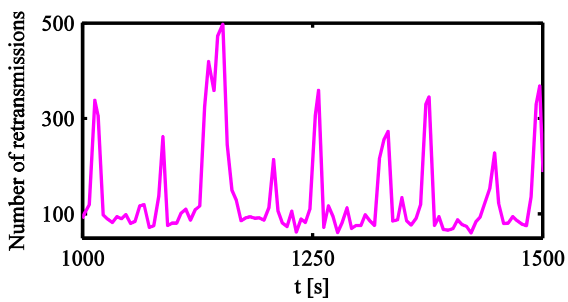

Fig.11. Number of retransmissions

If we compare the two figures, paying special attention to the time interval from 1050 to 1150th second, we can conclude that if more native packets are coded together, fewer packets need to be retransmitted.

I.e. for t = 1052 s number of average coded packets is 2.77 while number of retransmissions is 80. Contrarily for t = 1088 s number of average coded packets is 1.71 while number of retransmissions is 262.

Furthermore, the coding and decoding require additional delay. Still, we expect that the additional delay introduced by our scheme will not affect the Quality of Experience (QoE) on client side as all the operations will be carried out within the buffer time of the stream which is in our case 0.3 s (RTO).

5 Conclusion

Presented approach can be implemented in the wireless broadcast network when wireless technology is used for the last mile access. Our scheme introduces bandwidth reduction for video-streaming applications such as IPTV.

Using the NC, instead of retransmitting every lost packet separately we combine different packets and send them in fewer retransmissions. In this way bandwidth reduction can be achieved. We have shown from wireless testbed deployment that our solution compared to no NC retransmission approach reduces up to 15 % of bandwidth. This bandwidth reduction in practice is important for ISPs as they can avoid installing new equipment to get more clients or alternatively they can introduce new services.

With the proposed scheme high gains can be obtained for different delivery probabilities. Even higher gains can be obtained with more clients, which is the case in the real environment, as we have more different streams. With more clients, more different packets can be coded together which results in more coding opportunities and fewer retransmissions.

References:

[1] R. Ahlswede, N. Cai, S.-Y. R. Li, and R. W. Yeung, "Network information flow " *Information Theory, IEEE Transactions on* vol. 46, pp. 1204 - 1216, 2000.

[2] T. Matsuda, T. Noguchi, and T. Takine, "Survey of Network Coding and Its

Applications," *IEICE TRANSACTIONS on Communications,* vol. E94-B, pp. 698-717, 2011.
[3] D. Niu and B. Li, "Asymptotic optimality of randomized peer-to-peer broadcast with network coding," presented at the INFOCOM, 2011 Proceedings IEEE 2011.
[4] C. Jing, S. Xue-peng, and L. Fang, "Effective Retransmission in Network Coding for TCP," *International Journal of Computers Communications & Control*, vol. 6, pp. 53-62, 2011.
[5] S. Haoliang, L. Lixiang, and H. Xiaohui, "A Network Coding based DTN Convergence Layer Reliable Transport Mechanism over InterPlaNetary Networks," *International Journal of Computers, Communications & Control,* vol. 6, pp. 236-245, 2011.
[6] K. Yasami, A. Razi, and A. Abedi, "Analysis of Channel Estimation Error in Physical Layer Network Coding," *Communications Letters, IEEE,* vol. 15, pp. 1029 - 1031, 2011.
[7] S. Katti, H. Rahul, W. Hu, D. Katabi, M. Médard, and J. Crowcroft, "XORs in the Air: Practical Wireless Network Coding," *IEEE/ACM Transactions on networking,* vol. 16, pp. 497 - 510, 2008.
[8] H. Seferoglu and A. Markopoulou, "Video-Aware Opportunistic Network Coding over Wireless Networks," *IEEE Journal On Selected Areas In Communications,* vol. 27, pp. 713 - 728, June 2009.
[9] D. Ferreira, R. A. Costa, and J. Barros, "Real-Time Network Coding for Live Streaming in Hyper-Dense WiFi Spaces," *Selected Areas in Communications, IEEE Journal on,* vol. 32, pp. 773 - 781, 2014.
[10] P. Vingelmann, F. H. P. Fitzek, M. V. Pedersen, J. Heide, and H. Charaf, "Synchronized multimedia streaming on the iphone platform with network coding," *Communications Magazine, IEEE,* vol. 49, pp. 126 - 132, 2011.
[11] W. Fang, F. Liu, Z. Liu, L. Shu, and S. Nishio, "Reliable Broadcast Transmission in Wireless Networks Based on Network Coding," presented at the Computer Communications Workshops (INFOCOM WKSHPS), 2011 IEEE Conference on Shangai, 2011.
[12] D. Nguyen, T. Tran, T. Nguyen, and B. Bose, "Wireless Broadcast Using Network Coding," *Vehicular Technology, IEEE Transactions on,* vol. 58, pp. 914 - 925 2009.
[13] X. Xiao, Y. Lu-Ming, W. Wei-Ping, and Z. Shuai, "A Wireless Broadcasting Retransmission Approach Based on Network Coding," presented at the Circuits and Systems for Communications, 2008. ICCSC 2008. 4th IEEE International Conference on Shangai, 2008.
[14] K. Alič, E. Pertovt, and A. Švigelj, "Bearing-opportunistic network coding," *International Journal of Computers, Communications & Control,* vol. 10, pp. 154-164, 2015.