Performance Evaluation of VoIP Codecs over Network Coding in Wireless Mesh Networks

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Abstract: Voice over Internet protocol (VoIP) is used for transmitting voice signals in a packet-switched Internet protocol (IP) networks in real time. For transmitting voice over a wireless mesh networks (WMNs), the analog voice signal has to be digitalized, encoded, and packetized. Codecs based on different Quality of Service (QoS) requirements are used. One of the main QoS requirements is that packets are transmitted through the network in real time; one-way transmission time or End-to-End (ETE) packet delay, and packet delay variation or jitter have to be lower than thresholds. ETE delay depends on various parameters; among them is also network delay. Various mechanisms are used to lower the network delay in WMNs. A promising mechanism, for improving the performance of streaming services such as the case also in VoIP, is network coding. In this paper, we evaluate the benefits of using wireless network coding for VoIP in WMNs. Network coding procedure in combination with various VoIP codecs is used to observe the impact on network delay and jitter of the VoIP application. The simulation results show that network coding can decrease network delay and jitter. Moreover, results show that network coding benefits are codec dependent.

Keywords: Voice over Internet protocol; network coding; wireless mesh networks; performance evaluation

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1. Introduction

Voice over Internet protocol (VoIP) is a paradigm dealing with delivery of voice application and multimedia sessions over the packet-switched broadband Internet protocol (IP) networks in real time [1]. Voice signal is divided into VoIP packets based on various used codecs, which consider different Quality of Service (QoS) requirements [2]. VoIP packets are transmitted with other IP packets over the network.

Typical representatives of wireless packet-switched IP networks are wireless mesh networks (WMNs), where nodes are connected to each other through multi-hop wireless links forming a wireless access/backbone network [3, 4]. In order to improve the WMNs performance, various mechanisms are used. Among the promising mechanisms, which experience an increasing attention, is also network coding [5]. Instead of using "classical" receive and forward mechanism for packets, network coding combines multiple received packets either from the same or from different traffic flows into one encoded packet and then forwards it in order to increase the network capacity. In wireless networks, network coding exploits the broadcast nature of the wireless medium, where nodes can overhear packets, which are not destined to them, resulting in new coding opportunities, which enable combining even more packets together [6]. A practical network coding procedure, COPE, is proposed in [7] that encodes two or more packets in a single transmission based on the nodes knowledge on what information (which packets) neighboring nodes have. The procedure was tested in a real WMN deployment, which is of a particular importance [8].

VoIP application is highly exposed to QoS impairment in wireless IP networks, such as WMNs [9], even when using QoS enforcement [10]. VoIP QoS performance can be improved with various mechanisms. As the VoIP is a real-time application and requires specific QoS, the benefits of using the novel mechanisms for VoIP have to be tested.

In this paper, we investigate the benefits of using wireless network coding for VoIP application in WMNs. The VoIP application is presented with the emphasis on the one-way transmission time or End-to-End (ETE) delay and packet delay variation or jitter QoS requirements. We also show various codecs delay characteristics. Moreover, we present network coding for WMNs used to decrease the network delay. In addition, we perform extensive simulations to evaluate the performance of various VoIP codecs when using network coding in WMNs in the sense of a network delay and jitter.

2. Voice Over Internet Protocol Application

As the internet is a packet-switched network, the voice of VoIP telephone call has to be packetized before being sent through the network. The packetization is the process of dividing the data of a stream into structured blocks, called packets [11]. The packetization of the voice has to consider the fact that real time delivery of packets has to be performed [12]. For this, different types of codecs (coder/decoder) exist. Codec is a coding/decoding device which samples a voice signal and transforms it into a digitalized form with a predefined bit rate [13]. It also compresses the data of the signal to reduce the bandwidth requirements of established call. Codec selection is a balance between the bandwidth efficiency and the quality (compression level) of transmitted VoIP calls [14]. Some of the most frequently used (standard) codecs for VoIP packet transmissions are G.711, G.722, G.723, G.726, G.728 and G.729 [15].

VoIP application is a real-time application and IP network is not perfectly designed for such applications. The IP network is not as robust as public switched telephone network (PSTN) network in terms of the network reliability. There is no guarantee that packets are successfully delivered in sequential order to the destination, therefore, QoS is not guaranteed. Instead of that, best-effort transmission takes place in IP networks. If the network conditions are bad, a receiver will have difficulties understanding the speaker's speech. In the worst case scenario, receiver will not be able to understand or hear the speaker at all. In these cases, the conversation through VoIP call is not possible. There are several QoS specifications in the sense of various parameters limitations to be followed. These limitations have to be taken into account in the case of using VoIP.

Parameters with major impact on the VoIP call QoS are: one-way transmission time or end-to-end (ETE) packet delay, packet delay variation or jitter, packet loss rate, bandwidth, out-of-order packet delivery and hardware capacity [16]. The stated parameters have to be under the required threshold values to prevent call degradation that can result in the high delay, the understanding difficulties, etc.

In the following, one-way transmission time and jitter influences on the VoIP QoS will be presented and the threshold values of these parameters, beyond which network should not go if supporting a certain QoS of VoIP application, will be given. The one-way transmission time and jitter parameters are then investigated with network coding in Section 4.

2.1 One-way Transmission Time

Group TIPHON [17] classifies VoIP application into different network QoS performance classes regarding the voice packets one-way delay [18]. In the case of speech transmission it is a "mouth-to-ear" delay; the delay between the time a packet is sent from the "speaker" and the time a packet is received at the "listener". The classes are provided in Table I.

TABLE I. VOIP QOS CLASSES REGARDING THE ONE-WAY PACKET DELAY.

	3 (Wide-band)	**2 (NARROWBAND)**			**1 (Best effort)**
		2H (High)	**2M (Medium)**	**2A (Acce-ptable)**	
Relative Speech Quality (one way, non-interactive speech quality)	Better than G.711	Equivalent or better than ITU-T Recommendation G.726 at 32 kbps	Equivalent or better than GSM-FR	Not defined	Not defined
Delay	< 100 ms	< 100 ms	< 150 ms	< 400 ms	< 400 ms
NOTE: The delay for best effort class is a target value.					

In ITU-T G.114 [19] recommendations is stated that one-way delay should never exceed 400 ms for general network planning. As long as the ETE delay is kept below 150 ms, only a few VoIP sessions may get affected. From the user point of view, delays up to 290 ms are satisfactory. Delays between 290 ms and 400 ms cause the dissatisfaction to some users. Delays above 400 ms can only be used if we suppose that the user is familiar with higher delay as, e.g., in a satellite communication.

VoIP application delay has different causes [20]: coding/ encoding, packetization, jitter buffer and network delay (or network latency). The one-way delay caused by the first three stated causes is described as a codec delay. It can be calculates as:

$$CodecDelay = CSI + CPP + CPP + JBS \quad (1)$$

CPP is so-called pooling period of Central Processing Unit (CPU) or CPU pooling period and is ½ of the *CSI*, which stands for codec sample interval and is the time interval of a speech a codec takes and handles at once. *JBS* represents jitter buffer size. For example, *CSI* of codec G.711/10 is 10 ms. Thus, *CPP* is 5 ms and the recommended JBS for G.711/10 is 20 ms. *CodecDelay* results then in 40 ms.

Network delay occurs as the packet is sent through the network. However, it cannot be defined, as delays, presented above. In WMN, packet is sent through several wireless routers to be delivered to its destination. Different packets can be routed through the network with different speeds resulting in the variable delays of packets on the receiver.

2.2 Jitter

Jitter describes a non-constant packet delay at the receiver as the packet latency can vary when packets are sent across the IP network [14]. Jitter can occur when packets of the same stream are sent via different routes through the network. Beside this, it can occur as the traffic intensity of a network can vary through the time thus delaying packets differently. Expected jitter influences the size of a jitter buffer. Higher the jitter, greater the size of a jitter buffer needed to compensate the difference in the delay of packets of the same stream at the receiver. This buffer enables a continuous speech. The jitter buffer size is the same or a multiple (i.e., 1, 2, 3) value of CSI interval. In Table II, delays are represented for various codecs when taking also into account a jitter buffer delay besides the delays in (1), assuming the jitter buffer size of two CSI (i.e., 2 * CSI). Network delay is not considered here.

TABLE II. ONE-WAY CODEC DELAYS FOR VARIOUS CODECS.

	One-way codec delay [ms]
G.711/10	40
G.711/20	80
G.711/30	120
G.723/30	120
G.723/60	240
G.729/20	80
G.729/40	160

Jitter is measured as difference in ETE delays between the two consecutive packets of the same VoIP call stream. The jitter values greater than 100 ms are causing delays which are above ITU-T organization recommendations. Jitter values from

100 ms to 200 ms can be still handled by some jitter buffers introducing some conversational problems. If the packet arrives at the VoIP device too late (i.e., out of the jitter buffer value), it is lost. In the context of a network, packet jitter is measured as the average of all jitter packets values.

3. Network Coding for Wireless Mesh Networks

Network coding is the mechanism to improve the network performance. It experiences an increasing attention in the past few years in both, wired and wireless networks, mainly due to promising results from the initial research and testbed deployments [7, 8].

Network coding enables encoding multiple packets either from the same or from different traffic flows into one encoded packet for saving bandwidth and thus increasing the network capacity while maintaining the desired Quality of Service parameters. It can be also used to decrease the network delay, as will be demonstrated in Section 4. In wireless networks, network coding exploits the broadcast nature of the wireless medium, where nodes can overhear packets which are not destined to them, resulting in new coding opportunities [6]. These packets are later on needed for decoding process.

The network coding principle is presented in Fig. 1, where it is assumed that we have wireless nodes (e.g., wireless routers). Nodes S1 and S2 has to deliver packets m1 and m2 to nodes D1 and D2. Without network coding, packets are first sent to a relay node R and then forwarded to its corresponding destinations. Therefore, four transmissions are required to deliver packets. While with network coding, three transmissions are only required to deliver packets, as both packets are encoded into one packet (linear operation over the two packets) on node R, which is then forwarded to both destinations. Therefore, only one transmission is required by node R. The coding is possible as D1 knows m2 as it hears node S2 and can decode m1 from encoded packet sent from node R. Similar, D2 knows m1 as it hears node S1 and can decode m2 from encoded packet sent from node R. With this, one transmission has been saved.

One of the well-known network coding procedures for increasing the throughput of a WMN is COPE procedure, which is described in the following.

A. *COPE Network Coding Procedure*

COPE [7] is an intra-session network coding algorithm, which exploits the broadcast nature of the wireless medium. It codes packets for one hop, where packet decoding is done. The coding process depends on the nodes knowledge on what information (which packets) neighboring nodes have. In case the node knows which information neighbors have (through listening to neighbor's broadcasts (packets and ACKs) or receiving their updates) the coding process is straightforward and the decoding process will have a high success rate. Information arriving through particular massages and through listening to all the broadcast, is not sufficient and provides only few coding opportunities. In the case that the information on the packet presence at specific neighbor's node is not available the coding needs to guess on the situation. The node estimates probability that the node A has packet P, by looking at the delivery probability between packet's previous hop and node A. With all the needed information the node can code together as many packets as possible, as long as none of the packets have been created on this node, all the packets have different next hops and we know that there is a strong possibility that each next hop (all the neighboring nodes that we are encoding packets in for) will be able to decode the packet. The next hop can decode the packet if it has already received all except one of the packets coded together.

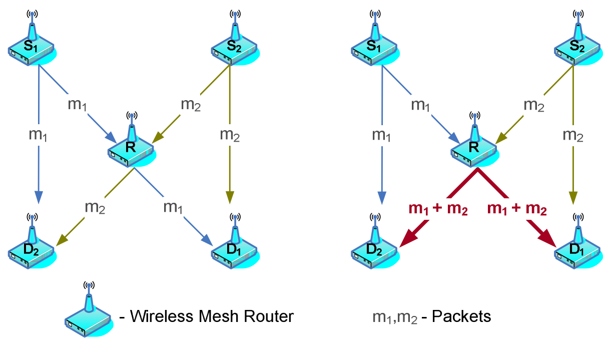

Fig. 1 Presentation of wireless network coding principle.

4. Performance Evaluation of Voip Using Network

We performed the evaluation of VoIP with network coding using network coding simulation model, which we presented in [21, 22]. The simulation model has been built using OPNET Modeler [23] simulation tool. In this chapter, we present and analyze the results obtained by simulation runs in the simulation model. We compare simulation results when using VoIP without network coding to the simulation results when using VoIP with COPE network coding procedure.

The performance of VoIP with network coding was tested in different network topologies and simulation results were collected for each of them. After analyzing the results, one network topology was chosen for the representation as an example, although the similar results were obtained by different topologies. The results from the presented network topology were chosen to present the VoIP performance using various codecs in a typical WMN with or without network coding.

4.1 Simulation Parameters

The parameters used in our case are numerous. In this chapter, we describe the main parameters that are used in different simulation runs.

In our analysis, we assume that all wireless network nodes are of the same type and have identical configuration, representing homogeneous network. Networks with different number of nodes and topologies are investigated, where each node is given a random location within a given area. A typical network topology for WMNs with 10 wireless nodes and 3 neighbors per each node, depicted in Fig. 2, has been selected and is further analyzed in this paper. The wireless nodes have been randomly positioned within the square size of 2000

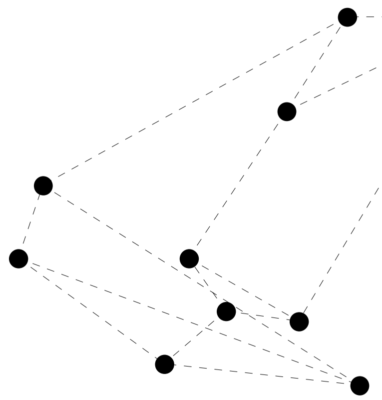

Fig. 2 Network topology with 10 nodes and 3 neighbors per node.

meters per 2000 meters (i.e., 2000 m * 2000 m), which is the size of the simulated wireless environment.

Each node has 1 Mbit/s of channel bandwidth. Wireless connections established between neighbors, which are represented in the network topology as wireless links, are graphically presented in Fig. 2 with dashed lines between nodes. For the simulation purposes all the links are symmetrical are lossless, meaning that no packets get lost during transmissions. Lossless links means that network conditions have to be perfect or there are some good wireless connections in the network, which can be selected as wireless links. Moreover, packets on wireless links are delayed due to propagation through wireless medium.

In the simulation, VoIP application is simulated establishing VoIP calls between node pairs. VoIP call is simulated with two packet streams being sent between the two wireless nodes, which are representing the two speakers of a VoIP call. VoIP calls are established between each node pair in the network. For a network topology with 10 wireless nodes in Fig. 2, it results in 45 individual calls. Only one VoIP call is established at the same time in the network. Each simulated VoIP call lasts for 30 seconds. In Table III, the parameters for various used codecs are presented. The total size of VoIP packet and the number of VoIP packets sent each second (i.e., packet per second, PPS), are calculated for various codecs used in simulation, considering the fact that VoIP application is implemented in 802.11b WMN. The VoIP packet total size represents the size of voice payload data (i.e., the codec sample size, CSS) and 802.11b overhead in one VoIP packet. The CSS depends on the codec bit rate (CBR), which determines the number of bits per second that has to be sent to deliver a voice call, and, the codec sample interval (CSI), which is the time interval of a speech a codec takes and handles at once. PPS represents the number of packets that has to be transmitted every second in order to deliver the codec bit rate. It depends on the CSI. In addition, traffic load per second, produced by VoIP call every second, is calculated by multiplying PPS value and VoIP packet total size for each codec in Table III. The results of various codecs are then compared between them.

TABLE III. CODECS PARAMETERS.

Codec	CBR [kbit/s]	CSI [ms]	CSS [bytes]	PPS [pps]	VoIP Packet Total Size [bytes]	Traffic load per second [bytes]
G.711/10	64	10	80	100	178	17800
G.711/20	64	20	160	50	258	12900
G.711/30	64	30	240	33	338	11154
G.723/30	6.4	30	24	33	122	4026
G.723/60	6.4	60	48	16	146	2336
G.729/20	8	20	20	50	118	5900
G.729/40	8	40	40	25	138	3450

Background traffic is simulated all the time during performing VoIP calls. It is simulated to evaluate its impact on the performance of VoIP calls. Background traffic load is generated as packet streams between all nodes with the same intensity using exponential distribution of inter-arrival times and constant packet lengths (i.e., 10 kbit). The background traffic load is increased through simulation runs until the VoIP packet delay in the network is not being increased due to this traffic and VoIP traffic can not be handled any more by the network to have a feasible speech communication. All network nodes are source nodes for generating background traffic with the same probabilities and select destination nodes using uniform probability distribution among all network nodes. Results are presented for six different intensities of traffic background loads (i.e., for six different total amounts of background traffic sent into the network), denoted by L1, L2, L3, L4, L5, and L6. L1 represents the lowest network traffic intensity used in the presented results, while L6 represents the highest intensity (i.e., the intensity, when network is already congested). It means that the used intensities increase as follows: L1 < L2 < L3 < L4 < L5 < L6. The network diameter is 3. The network average hop count is 2.

COPE network coding procedure [7] has been used for encoding packets for increasing the network throughput. The simulation cases without network coding are compared with the cases when COPE is used in the network to evaluate the impact of network coding on the performance of VoIP application in WMN.

Important modification has been made to the COPE procedure to increase packet delivery reliability at the network coding layer. Instead of using cumulative ACKs as described in the original paper each coded packet is immediately confirmed with the individual ACK packet. This allows us to shorten the round time and schedule possible retransmissions sooner. This is an important modification as it lowers the jitter and decreases the possibility of receiving packets with delay higher than expected by QoS parameters. The individual ACKs increase the overhead in the network and thus lower the network goodput.

Routing of packets through the network was done using static tables, which were calculated ahead of simulation runs. Routing tables are calculated using Dijkstra's algorithm taking into account hop count distances between nodes.

The timeline of simulation was as follows. Every simulation run took 1400 seconds. The background traffic was generated during the whole simulation run. The time of 5 seconds (warm up time) is required at the beginning of the simulation to have steady state conditions. Only one VoIP call between two wireless nodes was established at the same time in the network using one of the codecs in Table III. VoIP calls were generated consecutively with the 1s delay between them. In one simulation run, 45 individual calls were simulated and each VoIP call lasted 30 seconds resulting in 1350 seconds of VoIP calls simulation. At the end, 5 seconds are used for simulation control purposes as, e.g., the intensity of network congestion in the case of high background traffic, which is detected by receiving VoIP packets at the receivers also after the 1355th second of simulation, up to the 1400th second. In each simulation run, the background traffic was increased.

4.2 Simulation Results

We have averaged the network delay of all calls established in one simulation scenario. In every scenario, a particular codec has been used for transmitting VoIP calls. To simulate different traffic densities in the network, we have created different amounts of background traffic. Then, we evaluated how background traffic affects the VoIP application performance with various codecs. The results are presented in Fig. 3.

From Fig. 3, we can see that delays are increasing with the increased background traffic in the network for all codecs, as expected. Codec G.711/10 has the highest average network delay, while G.723/30, G.723/60, G.729/20 and G.729/40 have lower delay. This is because of the specific traffic load per second a particular codec has, which is presented in Table III. Please note that we do not present the scenarios, where the network gets congested (i.e. delays goes towards infinity). Therefore, there is no mark for these scenarios on the graph in Fig. 3 (see curves going into “infinity”, out from the figure). Similar is also done in the figures, which are presented in the following. Background traffic loads (e.g., L4, L5, L6), when delays are very high, cause (in some cases) network congestion, when using a particular VoIP codec. It means that we are presenting the results, when network is highly loaded or is already congested, with the exception for L1.

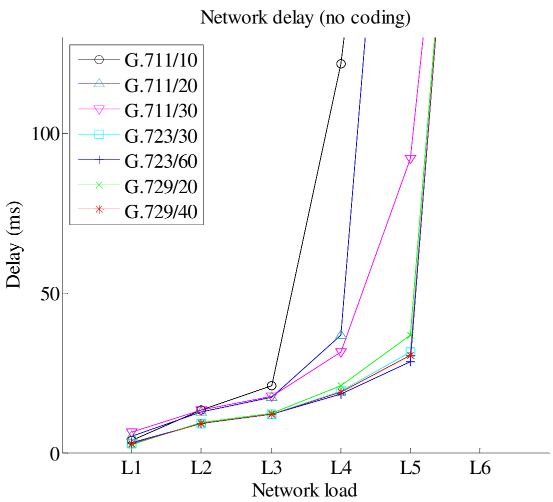

Fig. 3 Network delay when network coding is not used.

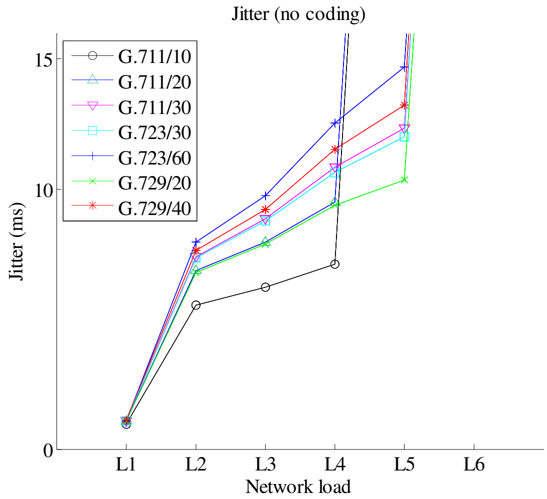

Fig. 4 Jitter when network coding is not used.

We have done the same for jitter measurements. In Fig. 4, jitter is presented for various used codecs in dependency of background traffic load. We can see that average jitter is increasing with background traffic, but not so rapidly as network delay in Fig. 3.

After analyzing the VoIP performance without network coding, we have also performed simulations, when COPE network coding procedure has been used in the WMN network to increase the throughput of the network. The scope of that was to investigate the impact of network coding on the VoIP performance. In Fig. 5 and Fig. 6, network delays and jitters are presented for the cases, when network coding (i.e., COPE procedure) is used on wireless nodes in the network. The results are presented for the same scenarios as in Fig. 3 and Fig. 4.

When using COPE, average network delays are lower, when background traffic load is high, compared to the cases when network coding is not used in the network. This difference between the COPE and no-COPE is increasing with

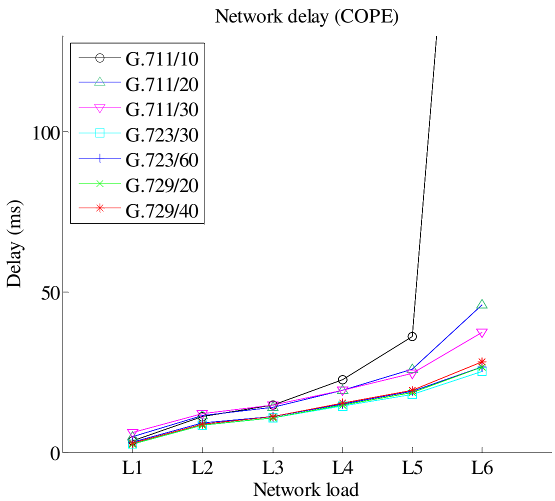

Fig. 5 Network delay when COPE is used.

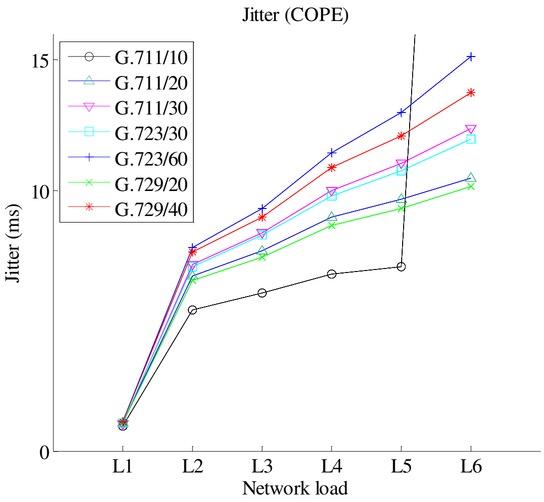

Fig. 6 Jitter when COPE is used.

the increased background traffic, as expected. More packets are in the network, more coding opportunities arise and more packets can be encoded, thus saving more bandwidth at the transmission. Moreover, it can be seen from Fig. 5 that VoIP application using COPE, in most cases (not true for codec G.711/10 with L6), still performs well having L4, L5 and L6 background traffic in the network, while without network coding the VoIP application is degraded due to high delays of VoIP packets (represented with no marks on the graph). Here, we can conclude that network coding improves the performance of VoIP application in WMN, especially when the network is high loaded or overloaded to a certain point. For jitter values, the difference between COPE and no-COPE case is very small, so the improvement is, in most cases, negligible. It is worth noting that using COPE does not increase the value of jitter.

In addition, we have investigated the difference in the impact of using network coding with different VoIP codecs. We have compared the scenario of using COPE and the scenario when network coding is not used for various codecs in Fig. 7. The comparison of network delays and jitters, in dependency of different background traffic loads, using COPE procedure and without using network coding, is presented for various VoIP codecs, separately.

It can be seen that codecs, which require higher traffic load per second, benefit from network coding more than codecs with lower traffic load per second. Once more, this is due to the fact that more VoIP packets are encoded with other packets (because of the increased overall traffic load in the network), thus increasing more the capacity of the network with network coding.

We can conclude that VoIP application benefits from the use of network coding in WMN, as the network delay is decreased and the VoIP performance is improved when the network traffic is high or the network is already congested to a certain point. Moreover, network coding does not degrade jitter.

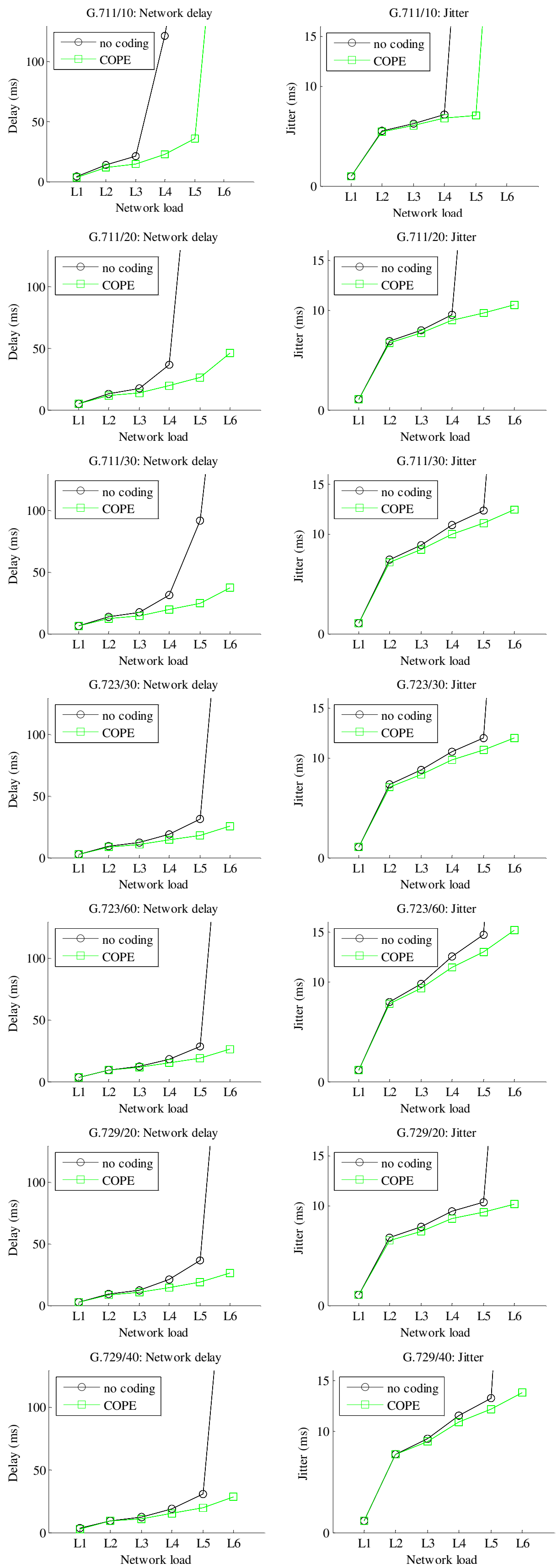

Fig. 7 Network delay and jitter for various codecs with and without COPE.

5. Conclusion and Further Work

This paper evaluates the use of network coding for VoIP application performance benefits in WMNs. We present the VoIP application using various codecs to transmit voice signal in a packet-switched IP networks in real time. Furthermore, we describe VoIP QoS requirements in the sense of one-way transmission time and jitter. Then, we describe the use of network coding in wireless mesh networks and present the well-known COPE procedure for network coding. After that, we compare the use of VoIP with and without using network coding in WMNs. The simulation results show that network coding can improve the VoIP performance in WMNs especially when the network is highly loaded or congested. Network coding decreases the network delay while the influence on jitter is small. The benefit of network coding depends on the used VoIP codec. Codecs, which require higher traffic loads per second, benefit from network coding more than codecs with lower traffic loads per second.

In further work, the use of VoIP with network coding in WMN should be also evaluated in more details. For example, the VoIP scenarios presented in the paper should be also analyzed at the level of individual calls; network delay, jitter and packet loss variation should be investigated throughout each call all the time.

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