Performance of a Novel Adaptive Traffic Aggregation Scheme for Wireless Mesh Networks

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***Abstract*— Wireless Mesh Networks are the most promising application of the IEEE 802.11 to the multi-hop wireless domain. The lack of scalability of WMNs installations represents anyhow a major obstacle to their success: in order to overcome such limitation and increase the number of potential users per installation, suitable techniques should be introduced. In this paper we describe a cross-layer scheme that improves the scalability of WMNs using aggregation of MAC layer frames. The scheme couples the routing metrics to the channel state. A closed formula for online computing the optimal burst length based on measurable routing metrics and the number of stations in range is proposed. We tested the scheme on a testbed in the specific case of VoIP flows, and showed a very large gain in the voice capacity attained, even in case of background traffic.**

I. Introduction

Wireless Mesh Networks (WMNs) are emerging as novel networking paradigm capable of supporting application scenarios where quick installation and flexibility are a major requirement. The major advantage of WMNs, in fact, is the ease of deployment and reconfiguration of the network infrastructure, thus allowing the deployment of networks with a minimum planning phase, while maintaining compatibility with existing WLAN installations (i.e. WMN as WiFi hotspots wirelessy interconnected). From the economic perspective, WMNs are expected to lower the entrance barrier to Wireless Internet Service Provider (WISP). This in turn by allowing them to deploy a wireless back-haul in an incremental fashion. At the same time WMNs can provide dramatic benefits by enabling wireless connectivity where there is not an in-advance knowledge of where the network should be deployed (e.g., tactical and emergency/disaster situations).

WMNs [1] exploit multi-hopping in order to wirelessy interconnect multiple communication nodes, possibly using different technologies/interfaces. WMNs share several features with the traditional ad hoc paradigm (namely the self-organizing, self-healing capabilities), even though research on WMNs shifted the focus from the support of mobility to network scalability issues. In fact, since WMNs are decentralized, at present they fit badly the requirements of nowadays multimedia applications with respect to the required packet loss and transmission delays [2], [3]. In turn, this translates into a strong limitation in the maximum number of hops that WMNs can support [4]. This is usually the concern about the applicability of WMNs to mission-critical scenarios.

In this paper we propose a novel packet aggregation technique which increases the scalability of IEEE 802.11-based WMNs. Such an aggregation is performed on top of the MAC layer, allowing us to reduce the overhead due to both protocol headers and the contention mechanism regulating the IEEE 802.11 standard. The novelty of the proposed approach lies on the adaptive aggregation scheme and leverages the channel probing functionalities of mesh routers: such information are exploited in order to compute the optimal saturation burst length. A closed formula allowing run-time computation of the optimal burst length based on measurable routing metrics and the number of stations in radio range is also proposed. A linear scaling is then applied in order to remodulate the burst length to the unsaturated operation point. We tested the scheme on a testbed in the specific case of VoIP flows, and showed a very large gain in the voice capacity attained, even in case of background traffic.

The paper is organized as follows. Section II is dedicated to related works. In Sec. III, we asses the impact of both encapsulation and contention overhead on the network performances. Section IV analyzes the performance of our traffic aggregation scheme in both saturated and un-saturated conditions. In Sec. V we illustrate the detailed architecture of our packet aggregation scheme. Section VI describes the experimental setup of our testbed. Section VII reports on the outcomes of the measurements exploiting VoIP traffic patterns. Finally, Sec. VIII concludes the paper pointing out future directions.

II. Related Work

There exist a vast literature on the modeling and analysis of the IEEE 802.11 CSMA/CA protocol. The fundamental model, i.e. the Bianchi model, is presented in [5]; there, a two-dimensional Markov chain is exploited in order to model the exponential back-off algorithm of the IEEE 802.11 Distributed Coordination Function (DCF) under saturation. Among several works elaborating on the initial model, such analysis is extended to error-prone channels in [6]. There, the authors conclude that, for any given bit error rate, an optimal packet size exists that maximize the goodput.

Related works on VoIP over WLANs [7], [8], [9] and UWB networks [10] proposed the introduction of a packet aggregation scheme. The technique trades off service time for packet length: the increase of CSMA/CA service time is mitigated by assembling multiple upper layer packets into a single MAC burst. Performance measures proved that the proposed MAC can significantly improve both throughput and delay performance in CSMA/CA based networks. In [3] the authors propose several performance optimization aimed at improving the VoIP support in WMNs. Voice packet aggregation and header compression are then exploited to improve the network capacity in terms of number of voice calls supported. Extensive simulation and experiments run over a real testbed are used in order to validate the proposed approach.

In [11], [12] an analytical model is developed in order to study the impact of packet aggregation on delay, and confirm that packet aggregation can significantly improve the performance of the CSMA/CA protocol. Such result are exploited by the authors in order to provide a novel packet aggregation policy capable of optimizing both the network throughput an delay.

Basic Rate		Full Rate				
PLCP Preamble	PLCP Header	MAC Header	LLC Header	SNAP Header	Payload	FCS
18 bytes	8 bytes	30 bytes	3 bytes	5 bytes	0 - 2304 bytes	4 bytes

Fig. 1. IEEE 802.11 Encapsulation.

In this paper we propose an adaptive packet aggregation scheme which leverages the mesh-routers' channel probing capabilities in order to modulate the burst length according to different link conditions. Our work extends both [5] and [10], [11] by introducing a novel model capable of matching the parameters available in a real-world WMN, namely link status and channel utilization ratio, with the requirements of an adaptive packet aggregation scheme. Finally, the performances of our scheme are evaluated by means of a testbed deployed at UFL premises.

III. Transmission overhead in WiFi networks

Data transmitted over a packet switched network requires recursive encapsulation starting from the highest layer (Application) down to the lowest layer (Physical). The overhead introduced by each layer is particularly relevant in the case of small sized packets, such those used in VoIP. Figure 1 sketches the frame structure for a IEEE 802.11 frame, as we can see the most relevant overhead is introduced by the Link Layer and by the Physical Layer headers.

The IEEE 802.11 standard [13] currently defines six modulation techniques, with data rate ranging from 1 to 54 Mbps, while the upcoming IEEE 802.11n amendment introduces an advanced physical layer based on MIMO techniques that is supposed to deliver data rates up to 540 Mbps. In order to allow the IEEE 802.11 MAC to operate with minimum dependence on the physical sublayer, the Physical Layer Convergence Procedure (PLCP) is introduced. The PLCP maps MAC sublayer Protocol Data Units (MPDU) into a framing format suitable for the Physical Medium Dependent (PMD) layer. The PLCP protocol data unit (PPDU) is unique to the specific PHY layer. The PPDU frame consists of a PLCP preamble, PLCP header, and MAC protocol data unit (MPDU). The PLCP preamble and PLCP header are always transmitted at 1 Mbps, while the MPDU can be sent at higher rates.

Additional overhead is introduced by the MAC and the LLC/SNAP headers. The Subnetwork Access Protocol (SNAP) is an extension to the IEEE 802.2 LLC and provides a mechanism for multiplexing different upper layer protocols. Further overhead is introduced by the Distributed Coordination Function (DCF) of IEEE

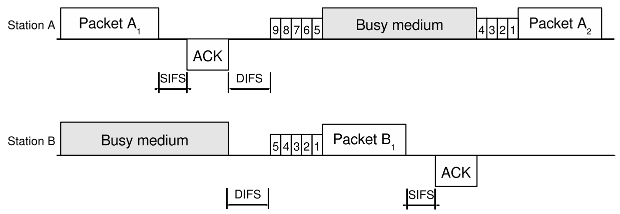

Fig. 2. IEEE 802.11 access scheme based on the DCF.

802.11 standard to access the wireless medium. According to the DCF scheme a station that wants to transmit a packet monitors the channel until an idle period equal to the Distributed Inter-Frame Space (DIFS) is detected. The station then transmits the packet. If no acknowledge (ACK) is received, i.e. collisions or errors affected the transmission, the station starts then the exponential backoff procedure setting a counter to a random number of slots. As long as the channel is sensed idle, the backoff timer is decremented at a fine granularity, and it is frozen when a transmission is detected, and reactivated when the channel is measured free for a DIFS interval. Thus, during the backoff procedure, the transmission of alien stations has to be accounted for the per-packet service time of a transmitting stations, with a large overhead due to backoff at the increase of the number of terminals transmitting in radio range [7].

IV. Aggregation in Error-prone Channels

Aggregating multiple higher layer packets into a single burst at the MAC level allow to both reduce the encapsulation and backoff overhead and increase the system throughput. The last claim is supported by the analytical model introduced in [5]. In this paper the author provides an accurate estimation of the saturation throughput for both the basic and the RTS/CTS access scheme. It is worth pointing out that such model does not take into account packet loss due to transmission errors; in reality, real world channel conditions are far from been error free and packets are lost due to both collision *and* transmission errors. In such a scenario exploiting the maximum burst length allowed by the underlying networking technology leads to degradation of system's performances (e.g., throughput, latency, packet loss). In fact, very long frames have an higher probability to get corrupted and retransmissions will vanish the service time reduction coming from aggregation.

A. Optimal burst length

Routing protocols developed for ad hoc networks often aim at minimizing the hop count. However, while the

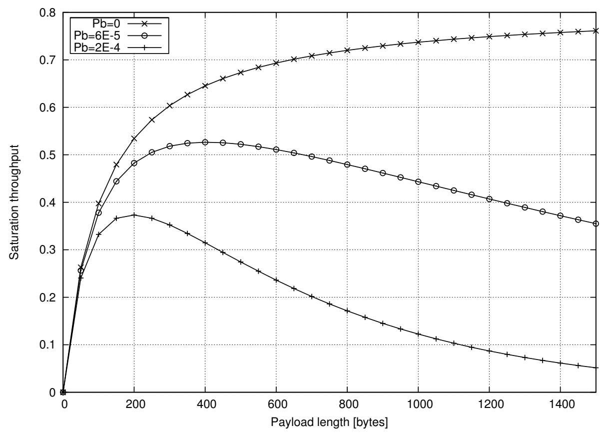

Fig. 3. IEEE 802.11 Saturation throughput versus an increasing length of the payload for different values of the ETX metric.

original work in this area was motivated by the need to support highly mobile applications (e.g., battlefields). However, WMNs have limited or no mobility and no constraints on battery life, and requirements to routing protocols are different. In fact, it is widely acknowledged that, even under light load conditions, routing metrics that minimize the hop count do not lead to good performances [14]. A simple counterexample is a two hop path over reliable and fast links, leading to better performances than a single hop route over an unreliable link. In this paper we will focus our attention on a specific metric, the Expected Transmission Count (ETX) metric and we aim at cross-layer techniques able to match routing and MAC layer parameters. We have chosen ETX being because it is used as basic building block for other routing metrics (e.g., ETT [2], [15], WCETT [15]) making our adaptive aggregation policy suitable for a wide rage of deployment scenarios. For a detailed evaluation of the performance of different routing metrics, including ETX, we refer to [14].

ETX estimates the average number of retransmission needed to successfully deliver a packet over a given link. In order to compute ETX, each node periodically broadcasts probes at data rates. Each probe contains the overall number of probes received from each of the neighbors during a specific observation window. Based on such counters, each station computes the unicast loss rate over a specific link. Considering that a successful unicast transmission in IEEE 802.11 requires sending the data packet and receiving the corresponding ACK, the ETX metric between nodes A and B, can be computed as

$$M_{\mathrm{ETX}} = \frac{1}{(1 - P_{AB})(1 - P_{BA})} = \frac{1}{P_{Uni}}, \qquad (1)$$

where P_{AB} and P_{BA} are, respectively, the loss rate from A to B and from B to A and P_{Uni} the overall probability of a successful unicast transmission. Being P_e the Frame Error Rate (FER) and p the collision probability, it follows

$$P_{Uni} = (1-p)(1-P_e). \tag{2}$$

Assuming also that errors after decoding are i.i.d. over the frame bits, and being P_b the Bit Error Rate (BER), is stands:

$$P_e = 1-(1-P_b)^L. \tag{3}$$

By combining (1), (2) and (3), we obtain

$$P_b = 1 - e^{-\frac{\log[M_{\text{ETX}}(1-p)]}{L_{\text{Probe}}}}, \tag{4}$$

where L_{Probe} is the length of the probe packet used for computing M_{ETX}. The network throughput computed according to the Bianchi's model presented in [5] can then be extended to account for error prone channels

$$\begin{aligned} S &= \frac{P_s(1-P_e)E[P]}{\sigma P_i + P_s P_e T_e + P_s(1-P_e)T_s + (1-P_c)T_c} \\ &= \frac{P_s(1-P_e)E[P]}{\sigma P_i + P_s T_s + P_c T_c} = \frac{AB^L L}{C+DL}, \end{aligned} \tag{5}$$

where for the ease of reading we put

$$A = \frac{P_s}{R}, \qquad B = 1-P_b, \tag{6}$$

$$C = \sigma P_i + P_s T_{s0} + P_c T_{c0}, \quad D = \frac{P_s+P_c}{R}. \tag{7}$$

Also, we denoted

$$T_{s0} = H + SIFS + \delta + ACK + DIFS + \delta \tag{8}$$

$$T_{c0} = H + DIFS + \delta \tag{9}$$

$$T_s = T_{s0} + E[P], \quad T_c = T_{c0} + E[P^*], \tag{10}$$

being and T_e the time the channel is sensed busy for a failed transmission due to a channel error. Notice that in (5) we assumed that $T_s = T_e$, which is due to the particular timeout settings in IEEE 802.11, namely the EIFFs. Figure 3 plots the system saturation throughput versus an increasing length of the payload for different values of the routing metric ETX. As we can see, the argument that is the basis for the packet aggregation technique dscribed in the following is that if we increase the payload length, we decrease system overhead. But, there exist a trade off, since frame errors become more likely with the packet length, i.e., the number of aggregated packets. Assuming as a first approximation constant length packets, the optimal packet length L_{Opt} writes as

$$f(M_{\text{ETX}}) = -\frac{C}{2D}\left(1-\sqrt{1+\frac{4DL_{\text{Probe}}}{C\log M_{\text{ETX}}(1-p)}}\right). \tag{11}$$

B. Runtime measurements and adaptation

Our aggregation scheme exploits the *monitor mode* provided by the Atheros-based WiFi interfaces building our testbed. Madwifi [16] is used as device driver. The monitor mode, or RFMON mode, allows a computer with a wireless NIC to monitor all traffic received from the wireless network, which gives us a good estimation of the channel load since it consider also traffic from other WiFi sources not participating the WMN. Monitor mode is similar to promiscuous mode used for packet sniffing in wired networks. This feature has been expploited to monitor the number of active neighbors in the last N seconds. This parameter is used in order to compute both the p and τ values that appear in the Bianchi's model [5], and the C, and D parameters used in (11). In order to speed-up computation, a look-up table is used. Numerical values are pre-computed for $1 \leq N \leq 30$. Using (11), we then approximate the optimal burst length as follows:

$$L_{Opt} = \min\left[f(M_{\text{ETX}}), B_{Max}\right], \tag{12}$$

where B_{Max} is the maximum frame size supported by the local area network technology. This parameter has been set to 1500 bytes which is the MTU (Maximum Transmission Unit) supported by an Ethernet LAN. Based on (12), and estimating the parameters described before, a node will then be able to update L_{Opt} according to the link conditions.

V. Implementation details

Our aggregation scheme concatenates several MAC Service Data Units (MSDUs) to form the data payload of a large MAC Protocol Data Unit (MPDU). The PHY header and the MAC header together with the Frame Check Sequence (FCS) are then appended in order to build the Physical Service Data Unit (PSDU). The frame format for an Aggregated MSDU (A-MSDU) is sketched in Fig. 4.

The building blocks of our MSDU aggregator and their relationships are sketched in Fig. 5. Incoming MAC frames are first classified according to the destination address. Each flows is then fed to a different queue. For each queue, an A-MSDU is generated when either an aggregation timer is expired or at least a burst of length L_{Opt} can be generated.

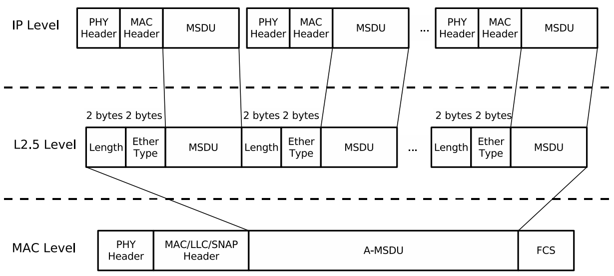

Fig. 4. Aggregated MSDU (A-MSDU) frame format.

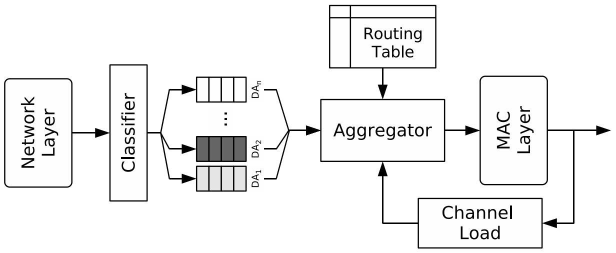

Fig. 5. Block diagram for the packet aggregator.

VI. Experimental Setup

A. Network Configuration

In order to asses the performance of our packet aggregation policy we have exploited a 4-nodes wireless testbed deployed in a typical office environment implementing a two-tier structure. Testbed's nodes are all Fujitsu notebooks model P7010D equipped with a 1.20 GHz Intel Pentium M processor and 512 MB of memory. All nodes run Debian GNU/Linux with kernel 2.6. Each node has a single IEEE 802.11b/g wireless NIC supporting data rates up to 54 Mbps. All measurement are run with the IEEE 802.11 interfaces operating in "b" mode with RTS/CTS disabled. Moreover in order to show the performance of our traffic aggregation scheme under different channel load conditions, an additional interfering node tuned on the same channel of the WMN have been placed close (less than 50 cm) to each mesh node. Interfering nodes are Linksys WRT54G wireless routers running the OpenWRT GNU/Linux distribution.

Our testbed is based on Roofnet [17], an experimental IEEE 802.11-based WMN deployed at Cambridge, Massachusetts (USA) by the Massachusetts Institute of Technology (MIT). Roofnet routes packets using a modified version of DSR [18] called SrcRR [2] exploiting ETX as routing metric. Routing is implemented using the Click modular router [19], developed at MIT. A Click router is built by assembling several packet processing modules, called *elements*, forming a directed graph. Each *element* is in charge of a specific function such as packet classification, queuing, and interfacing with networking devices. Click comes with an extensive library of elements supporting various types of packet manipulations. Such a library enables easy router configuration by simply choosing the elements to be used and the connections among them. We extended the default Roofnet configuration by implementing three additional elements responsible for packet aggregation and de-aggregation and for channel load evaluation.

Codec	Packet Interval	Bitrate	Payload
G.729.3	30 ms	8 kbps	30 bytes

TABLE I
KEY FEATURES OF THE G.729.3 CODEC

B. Evaluation methodology

In our tests, we focused our attention on VoIP applications due to: (i) their widespread use (e.g., Skype) and (ii) their strong requirements in terms of Quality-of-Service. Being very demanding in terms of loss and delay constraints, VoIP services are an interesting benchmark case, especially when dealing with mesh structures, where multihop communication could introduce unpredictable delays. A typical VoIP source tends to transmit a large number of packets with a small payload, and such a combination is known to lead to large protocol overheads [3]. Table I summarizes the key features of the G.729.3 VoIP codec being tested in our experiments. We made our probes through synthetic traffic generation, and we resorted to the E-Model [20], which provides an objective method to evaluate speech quality in VoIP systems. The outcome of an E-Model evaluation is called R-factor (R). The R-factor is a numerical measure of voice quality, ranging from 0 to 100, with 70 as lower bound for a VoIP call of acceptable quality. In the E-Model several different parameters affecting the quality of a conversation are taken into account. The main assumption is that various impairments at the psychological scale have an additive behavior (dB-like behavior), $R = R_0 - I_s - I_d - I_e + A$.

In particular, R_0 is the basic signal-to-noise ratio (environmental and device noises), I_s accounts for the impairments on the coded voice signal (loud connection and quantization), I_d represents the effect of delay, I_e the effect of low bit rate codecs and A is the advantage factor, corresponding to the user allowance due to the convenience in using a given technology. By choosing the default values suggested by ITU-T [20], the R-factor equation can be further simplified to $R = 94.2 - I_d - I_{ef}$, where I_d is the impairment factor caused by the end-to-end delay and I_{ef} is the equipment impairment factor:

$$
\begin{aligned}
I_d &= 0.024 T_a + 0.11(T_a - 177.3)H(T_a - 177.3) \\
I_{ef} &= I_{e_{opt}} + C_1 \ln(1 + C_2 P). \qquad (13)
\end{aligned}
$$

In (13), T_a and P are respectively the one-way delay in ms and the loss rate, $H(x)$ is the step function, and $I_{e_{opt}}$, C_1, C_3 are codec specific parameters. Their values for G.729 codec under random packet losses are respectively 11, 40 and 10.

C. *Traffic Generation*

VoIP flows were generated using Jugi's Traffic Generator (JTG), a freely available synthetic traffic generator [21]. Beside being able of generating and injecting different traffic patterns over TCP and/or UDP sockets, JTG can read the information about packet transmission intervals and sizes from files, allowing us to create an exact duplicate of a trace starting from a pre-recorder stream. Traffic is then collected at the receiver side where suitable tools are available for analysis.

The tests, whose results are reported in the next section, refer to downlink traffic only. In our settings, each mesh node sustains the same traffic, consisting in an increasing number of VoIP sessions. We have emulated each VoIP call by transmitting a UDP flow modeled according to the parameters of the G.729.3 VoIP codec (see Table. I). Background traffic, generated by the interfering node is modelled according with a TCP socket working in saturation regime. In order to collect reliable measures of delays, before each experiment we synchronized each node with a common reference using NTP [22]. All measurements were performed over 3 minutes intervals; results are averaged over 10 runs. Network topology and traffic flows are sketched in Fig. 6 where VoIP bundles and TCP flows are represented respectively by solid and dashed lines.

VII. Performance Measurements

Our measurements campaign aimed at evaluating the number of concurrent VoIP flows that can be sustained, i.e. the voice capacity of the system. According to ITU recomendations, we considered $R = 70$ as the minimum required value of R for acceptable quality VoIP calls. The system's performances were analyzed with and without background interference. We reported on the results for the scenario in Fig. 8. In this case, our adaptive aggregation policy provides a factor 3 performance increase, since the number of sustained sessions reaches 53, whereas the plain IEEE 802.11 protocol allows for just 11 VoIP sessions. However, as we can see from Fig. 7, when background interference is present the relative performance boost provided by our packet aggregation scheme is lower than the previous scenario, but it is still providing a factor 2 benefit.

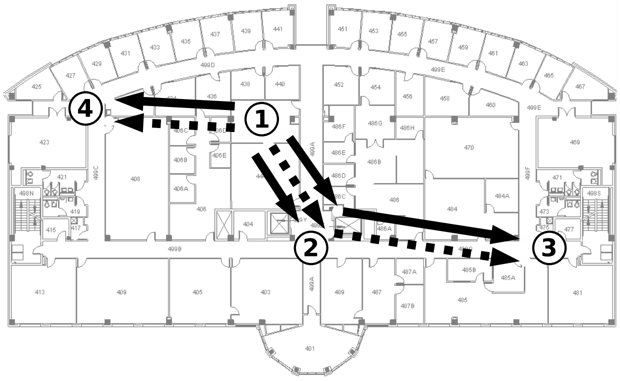

Fig. 6. Logical network topology.

VIII. Conclusions

In this paper we evaluated a novel scheme for traffic aggregation in WMNs, where service time reduction at the MAC layer is possible if several packets are compunded into a unique burst frame. The scheme receives as input some measurable link metrics, and each node can evaluate the optimal burst length, based on a closed formula under the saturation approximation. The scheme proved to increase the scalability of IEEE 802.11-based WMNs in the specific case of VoIP traffic. The entire measurements campaign was performed on a testbed deployed at UFL premises.

Overall, the results show that the aggregation scheme proposed here has a very large gain. With respect to the specific case of VoIP traffic, in fact, we experienced a factor $2/3$ increase in the number of simultaneous VoIP flows. Future research will be devoted to a measurement campaign on larger topologies to obtain further insight in the scalability properties of the solution proposed in this paper.

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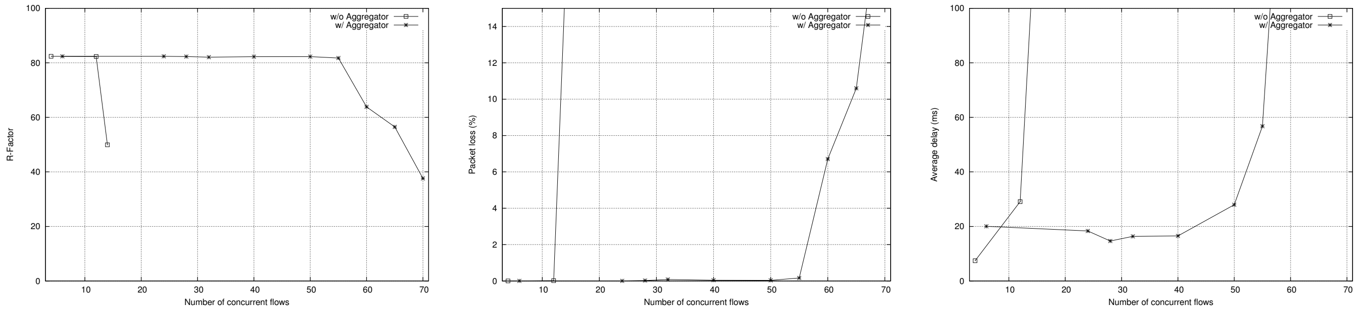
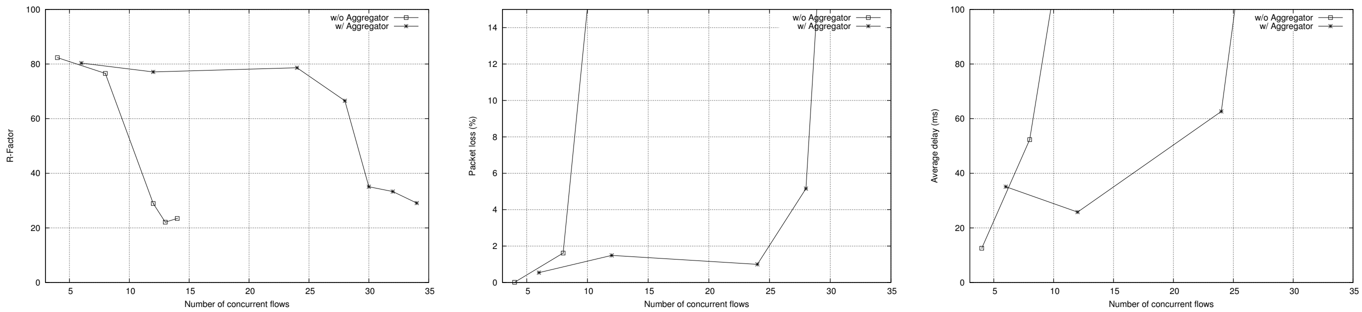


(a) Average R-factor. (b) Average packet loss. (c) Average delay.

Fig. 7. Performance measurements results without background interference.



(a) Average R-factor. (b) Average packet loss. (c) Average delay.

Fig. 8. Performance measurements results with background interference.

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