

MMTC Communications - Frontiers

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Message from MMTC Chair

Dear MMTC colleagues:

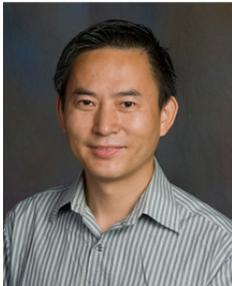
Happy holidays! It was very nice meeting some of our MMTC members at the IEEE Globecom 2016. I would like to thank all the past chairs/directors and volunteers for the work they have accomplished and their success of promoting MMTC in all aspects. I believe that our new MMTC leadership team will continue this success. In recent years, there are more and more demands for multimedia applications such as online movies, TV shows, and virtual reality (VR), which have fostered a significant growth of advanced multimedia technologies. This trend can also be supported by our multimedia conferences such as IEEE ICME in this community. I do feel there are strong needs to further expand our MMTC community and promote the communications among MMTC members. Our new MMTC leadership team will work with all MMTC members to grow our community. As the Vice chair of MMTC Letters & Member Communications, I would like to announce two important publications: Communications – Frontiers and Communications – Review, led by this new leadership team in the next two years.

The publication of Communications – Frontiers (was E-letter) will be led by Directors Guosen Yue (Huawei USA), Danda Rawat (Howard Univ.), Hantao Liu (Cardiff Univ.) and Dalei Wu (Univ. Tennessee at Chattanooga)

The publication of Communications – Review (was R-letter) will be led by Directors Pradeep K. Atrey (University at Albany, SUNY), Qing Yang (Montana State University), Jun Wu (Tongji University, China), and Wei Wang (San Diego State University).

I would like to call for more contributions to the Communications – Frontiers and more recommendations to the Communications – Review. I would like to extend my invitation to all our MMTC members to consider the opportunity of serving our community. I look forward to your participation!

Happy new year 2017!



Vice chair of MMTC Letters & Member Communications
Multimedia Communications TC of IEEE ComSoc

SPECIAL ISSUE ON Recent Advances of Multimedia Communications in Internet of Things

Guest Editors: Lei Chen, Georgia Southern University, Qing Yang, Montana State University

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The Internet of Things (IoT) can be defined as the internetworking of various types of devices, buildings and other things with the capability of collecting and exchanging data among them utilizing embedded electronics, sensors, actuators, software and network connectivity¹. The fast adoption of IoT in various network applications has witnessed the explosive demands of multimedia data processing and multimedia network traffic in various applications, from video and audio streaming over smartphones to real-time transportation traffic monitoring. There exists an increasing demand for handling ever-growing multimedia data in a secure and energy and resource efficient manner.

The five papers included in this special issue on multimedia communications of IoT aim to address a number of noteworthy challenges and present the corresponding solutions and suggestions. Most of these contributions are made by authors who are renowned researchers in the field, and the audience will find in these papers the research advances for enhanced multimedia communication performance in terms of better multimedia quality, less delivery delay and reduced delay jitter, and higher Peak Signal-to-Noise Ratio (PSNR), among many other metrics. Each of these five papers is briefly introduced in the following paragraphs.

Multimedia Sensing as a Service (MSaaS) has become a popular communication pattern in modern cloud edge IoT and fogs, yet energy efficiency is still a critical challenge due to limited resources on sensing devices. “*Energy Efficient Multimedia Sensing as a Service (MSaaS) at Cloud-Edge IoTs and Fogs*” presents the contribution made by Qin Wang, Wei Wang and Kazem Sohraby, where a new premium-based energy efficient resource allocation scheme to improve media quality is introduced. In this research, resources of transmission power, retry limit, and modulation size at various layers are adaptively allocated in a cross-layer manner. Simulations show that this joint optimization of resources achieves significant performance in multimedia quality gain.

Multimedia has been dominating networking applications and services in the recent years. Rami Haddad proposed, in his contribution “*Optimizing Internet of Multimedia Things Communication in Passive Optical Networks*”, a new exhaustive grant sizing using Feed-Forward Bandwidth Indication (FFBI) as the queue size prediction for Ethernet Passive Optical Networks (EPONs). This video source method feeds forward future video bandwidth requirements to be used by network devices to improve resource allocation. The experimental results in this research attest that the proposed approach provides accurate forecast when used as an exhaustive grant sizing method for reducing the video queueing delay and video delay jitter.

Recognizing the explosive growth of video traffic in IoT in the recent decade, in contribution “*Delay-Aware Fountain Code Strategies for Video Communication*”, Kairan Sun and Dapeng Wu introduced a novel delay-aware fountain code scheme for video streaming, deeply integrating channel coding and video coding. It is a pioneer work exploiting the fluctuation of bit rate in video data at the level of channel coding and incorporating it towards the optimal design of video streaming-oriented fountain codes. Simulation results evidence superior performance of the proposed scheme over the state-of-the-art schemes in terms of decoding ratio and PSNR.

Jun Huang, Huifang Yan and Qiang Duan presented a network-coding-based multi-hop Device-to-Device (D2D) communication model for IoT in their contribution “*Enhancing Capacity for Multimedia Communications in Internet of Things*”. In this research, they reviewed applications of network coding in wireless broadcasting, emphasized the feasibility of encoding packets by network coding at relaying nodes, and investigated four different

¹ <http://www.itu.int/en/ITU-T/gsi/iot/Pages/default.aspx>

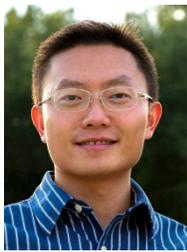
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and paradigmatic packet re-transmission schemes in the literature that are either equipped with the techniques of network coding or without. Simulation results clearly indicated that the schemes with network coding outperform those without, shedding light on and invoke future research work in this area.

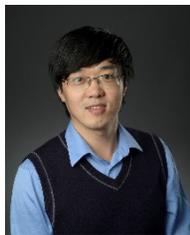
The IoT has emerged as a new paradigm deeply changing every aspect of daily lives, including daily transportation needs. Transportation IoT however may experience various security challenges preventing its wide deployment. In the contribution “*Toward Secure Transportation Internet of Things: A Trust Management Perspective*”, Wenjia Li, Yehua Wei and Feng Zeng study the problem of better securing transportation IoT via trust management approach by evaluating different behaviors of nodes and building trust based on behavior assessment observations among nodes. This research is valuable to the understanding of IoT security in transportation applications.

Due to the time constraint and volume limit, this special issue has no intent to present a complete scope of recent advances of multimedia communications in IoT. Nonetheless, we hope to bring to the audience the essence of selected innovative and original research ideas and progress for the purpose of inspiring future research in this fast growing area.

The guest editors would like to give our special thanks to all the authors for making contribution to this special issue. We are also thankful to the MMTC Communications – Frontier Board for providing helpful support.



Lei Chen received his Ph.D. degree from the Department of Computer Science and Software Engineering at Auburn University, Alabama, USA in 2007. He then joined Sam Houston State University where he was promoted to Associate Professor with tenure. In 2015 he joined Georgia Southern University. His research focuses on network security, cloud and information security, digital forensics, and networking. Dr. Chen is the editor-in-chief of book “Wireless Network Security: Theories and Practices” published by Springer and HEP in 2013. He has served as an editor for multiple special issues and in the editorial board for various journals and magazines. He has also chaired workshops and sessions with prestigious conferences such as INFOCOM, GLOBECOM, ICNC, WASA, and Mobile Cloud.



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Energy Efficient Multimedia Sensing as a Service (MSaaS) at Cloud-Edge IoTs and Fogs

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1. Introduction

Due to the popularity of smart devices and multimedia sensors, big-volume-of-data (BVD) multimedia content poses severe burden on wireless access channels in Internet of Things (IoT) [1]. Multimedia Sensing as a Service (MSaaS) [9] becomes a popular communication pattern in modern cloud edge IoTs and fogs, however, energy efficiency is still a critical challenge due to limited resources on sensing devices.

Past research have been done to reduce multimedia traffic for cloud media applications [2][3]. Different source data compression techniques such as exploring wavelet zero tree roots [4] and tree structure nodes [5] were also proposed to reduce redundancy of data. Based on scheduling methods [6][7], various resource allocation strategies regarding Unequal Error Protection (UEP) for prioritized media packets were discussed in [8][9].

However, most of the above research ignored the energy efficiency considerations for multimedia at the cloud edge fogs. Furthermore, traditional packet prioritization methods were not able to handle the big media data generated by MSaaS source devices. In this paper, we utilize application layer premium-regular concept among video packets, while exploring energy efficient resource allocation strategies at the cloud edge and the fog. Resources of transmission power, packet length, channel coding rate and packet retry limit are jointly optimized for quality driven MSaaS multimedia communications.

2. Quality-driven Resource Allocation

There are two essential challenges of big media data communication at cloud edges and fogs. The first is to decide packet priority according to their importance levels in media streaming reconstruction. Importance of video packets can be obtained in transform, spatial or temporal domain [9]. Then, network resource including transmission power, retransmission limit, channel coding rate and modulation rate should be allocated unequally among video packets based on their priorities.

Considering N pictures waiting for transmission, the model is to maximize media quality with energy consumption constrained. If we allocate more resource to important packets, these packets get better transmission quality, while the energy cost is also increased. Thus, the goal of resource allocation methodology is to find a balance between video quality and resource cost.

Let λ_i denote the threshold of important image blocks for image i , i.e., the truncation point between premium packets and regular packets. Let $P, b, S, C,$ and m denote transmission power, modulation constellation size, source data packet length, redundancy encoding bits in Forward Error Correction (FEC) scheme, and Automatic Repeat reQuest (ARQ) limit. $R=[P, b, S, C, m]$ denotes resource vector for image blocks. Video quality at picture level is defined as the successfully delivered distortion reduction quality gain per unit of energy consumption, i.e., the aggregated information quality (AIQ). The quality-driven energy-constrained resource allocation problem can be formulated as

$$\{\lambda_i, R_{premium_i}, R_{regular_i}\}_{\forall i} = \arg \max \left\{ \sum D(1-\omega)/E \right\}$$

$$\text{s.t. } \sum_i E_i \leq E_{max} .$$

Energy consumption can be expressed as

$$E = \frac{P \times (S + C + L_{oh})}{B \times b} \times \frac{1 - \omega^{m+1}}{1 - \omega}$$

where $D, \omega, B,$ and L_{oh} denote the distortion reduction, the packet error rate, the bandwidth, and the packet header overhead, respectively.

3. Performance evaluation

Figures 1 and 2 illustrate the video quality per micro-joule of energy consumption against channel amplification/loss using various schemes. Due to its effectiveness in controlling bit errors, prioritized FEC scheme that assigns unequal channel coding rates to different video categories has the biggest impact on improving quality per unit of power, followed by adaptive modulation ranking as the second choice. The performance gain of allocating diverse retransmission times among prioritized frames is similar to that of packetizing frames to different sizes. Retransmitting large frames with more reattempts has similar effect on improving video quality with packetizing video data to small coding sizes with fewer retransmission times.

Figure 2 shows that removing any kind of resource allocation strategy from the proposed priority-based energy efficient resource allocation scheme causes performance decline. Dismissing ARQ strategy leads to the least performance reduction. It means that we can jointly use the remaining four strategies to optimize video quality by removing ARQ strategy for the purpose of saving computing and communication cost. However, removing power control strategy causes the biggest performance loss. Compared with the result in Figure 1, we conclude that the performance gain is minimal if we only use power control strategy without any other resource allocation methods. But it occupies the most dominant position to improve system performance when jointly optimizing with other four resource allocation strategies.

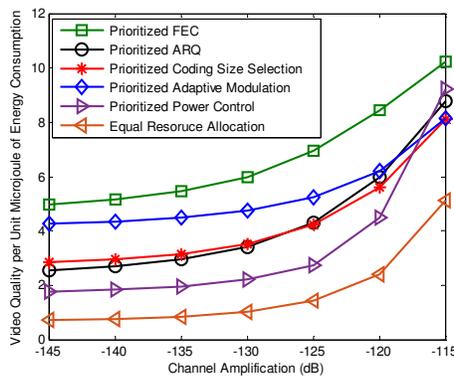


Figure 1. Video quality per micro-joule.

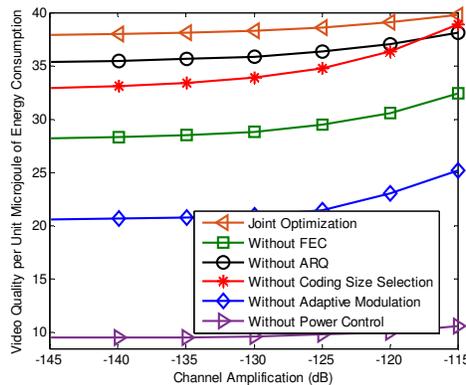


Figure 2. Video quality per micro-joule.

4. Conclusion

In this paper we have presented a new premium-based energy efficient resource allocation scheme to improve media quality. Resources of transmission power, retry limit, and modulation size at various layers are adaptively allocated among packets in a cross-layer manner. Simulations show that joint optimization of resources achieves significant performance multimedia quality gain.

Acknowledgement

This research was support in part by NSF grants CNS-1463768 and CNS-1423192 on energy efficient wireless multimedia communications.

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Optimizing Internet of Multimedia Things Communication in Passive Optical Networks

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1. Introduction

The recent advancements in designing low-cost small scaled devices have triggered a surge in the number of Internet-enabled devices. These smart Internet-enabled devices equipped with the capability to observe and/or interact with physical environment are extending the Internet towards the so called 'Internet of Things' (IoT) [1]. With multimedia dominating the applications and services, Cisco Virtual Network Index predicts that by the year 2020 the global video IP traffic will constitute over 82% of all consumer Internet traffic [2]. A significant portion of this video traffic will be generated by IoT devices, hence the notion Internet of Multimedia Things 'IoMT'. To accommodate this exponential growth in traffic due to IoMT and other multimedia services, passive optical networks (PONs) are being deployed as the next generation access networks since their bandwidth significantly exceeds the bandwidth of the current copper access network infrastructure [3]. The passive optical network structure has an optical fiber that connects the Optical Line Terminal (OLT) at the service provider side with multiple Optical Network Units (ONUs) at the user side through a passive splitter. This star topology mandates the use of a centralized medium access protocol to control the scheduling of the upstream traffic from the ONUs to the OLT. This centralized protocol operates using a cyclical polling process to service all the optical network units. However, this polling process can increase the upstream queueing delay which is very critical when streaming delay-sensitive traffic such as video traffic. Therefore, in this paper, an exhaustive grant sizing using queue size prediction for Ethernet PONs (EPONs) was proposed as a solution to eliminate this additional delay. To realize this, Feed-Forward Bandwidth Indication was proposed as an accurate forecasting approach.

2. Feed-Forward Bandwidth Indication

Feed Forward Bandwidth Indication (FFBI) [4,5] is a mechanism that feeds forward future video frame sizes and incorporates them into traffic ready to be sent as illustrated in Fig. 1.

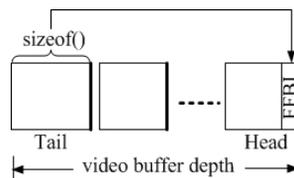


Figure 1. Feed-Forward Bandwidth Indication process.

This provides the network devices with valuable bandwidth information that can help them better provision their resources. This is viable since the majority of the video content used over networks is pre-recorded video, which means that the entirety of the video is available ahead of time at the point of streaming. Even for live video broadcast, FFBI can still be applicable by delaying the video stream by a very short period of time and creating a backlog of video frames. This will not affect the viewing experience since the viewer would not have a frame of reference to notice this small induced delay. However, when streaming live interactive video, there is a limitation to how much induced delay can be added to this video traffic set by the ITU G.114 specification [6]. This specification clearly states that one way delay of 150ms is considered acceptable. This will limit the number of cached video frames to 4 or 5 depending on the video frame rate, but will not render FFBI useless. FFBI relieves the network devices from having to predict the future incoming video traffic and instead provides a 100% accurate indication of the expected traffic.

3. Performance evaluation

In this section, we validate the effectiveness of IoT multimedia communication using the FFBI mechanism over simulated EPONs. The simulated EPON has 32 attached Optical Network Units (ONUs) with a 1 Gbps transmission rate. Each ONU has a data queue and a video queue. A shared limited grant sizing mechanism between the data and video queue was used. In this grant sizing mechanism, the video and data share the grant with the video traffic given the priority over data to utilize the grant up to a maximum limit.

In this experiment, a Poisson traffic source was used as input to the data queues, for rapid convergence, and video traces as input to the video queues. The Poisson traffic packet size distribution was based on the following quad modal: 60% 64 bytes, 4% 300 bytes, 11% 580 bytes, and 25% 1518 bytes. Six MPEG4 Part-2 video traces were utilized from the Arizona State University video trace library as the source of video traffic. The six video traces were: The Matrix (I, II, and III) and Lord of the Rings (I, II and III) [7]. All the videos had a common intermediate video format (CIF) (352×288 resolution) with a frame rate of 25fps and a GoP structure of G12B2. Video frame sizes were scaled to an average video traffic load of 100 Mbps, while the data load was varied from 0.1 to 0.8 Gbps.

Video packet queuing delay

Fig. 2 shows the average video packet queuing delay. It is observed that using FFBI as the bandwidth forecast grant sizing method provides a lower average video queuing delay compared to using no forecasting.

The average video queuing delay at 0.8Gbps data load when no forecast grant sizing is used is ~1.4ms. However, this delay is reduced to ~650µs when FFBI forecast grant sizing is used; this is a 50% reduction. This is the case since the lower bound on the video queuing delay is approximately a grant cycle length when no forecasting is used. This lower bound technically becomes zero when an accurate forecasting method is used to provide exhaustive service.

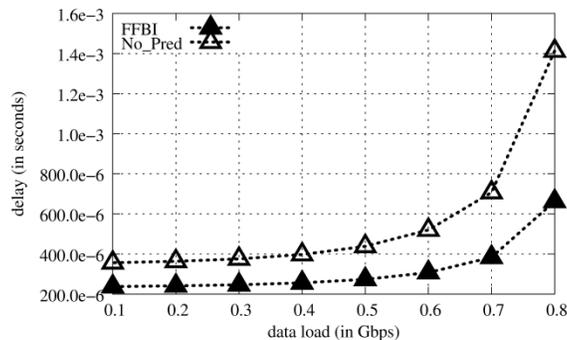


Figure 2. Average video packet delay

Video packet queuing delay variation

Fig. 3 shows the average video packet delay variation (i.e. delay jitter).

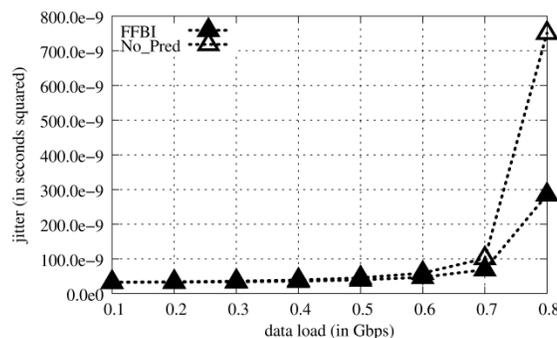


Figure 3. Video packet delay variation (jitter)

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As depicted, FFBI provides a significantly smaller queueing delay variation than no forecast grant sizing especially when the network is congested. The average queueing delay variation at 0.8Gbps data load when no forecast grant sizing is used is $\sim 750\text{ns}^2$, while FFBI results in a queueing delay variation of only $\sim 285\text{ns}^2$. Since FFBI always serves the video packets in the first granted transmission window after their arrival, this results in the least variation possible in the video queueing delay.

4. Conclusion

In this paper, a new exhaustive grant sizing using Feed-Forward Bandwidth Indication (FFBI) as the queue size prediction for EPONs was proposed. FFBI is a video source method which feeds forward future video bandwidth requirements to be used by network devices to improve resource allocation. FFBI provides accurate forecast when used as an exhaustive grant sizing method which reduces the video queueing delay and video delay jitter. The experimental results using FFBI to provide exhaustive queue service to ONUs of an EPON resulted in a 50% reduction in video queueing delay compared to the use of no forecasting, and a 62% reduction in video queueing delay variation compared to no forecasting.

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Delay-Aware Fountain Code Strategies for Video Communication

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1. Introduction

Internet of Things (IoT) has witnessed the explosive growth of video traffic over the recent decade, especially after the prevalence of smart mobile devices and surveillance cameras. At the same time, IoT nowadays relies more and more on wireless networks.

In order to deal with the stochastic loss in a wireless network, the Forward Error Correction (FEC) codes are developed. One important class of FEC codes are fountain codes, such as Luby transform (LT) code and Raptor code. However, traditional fountain codes are designed for file transfer rather than video communication. In order to accommodate fountain codes to video streaming, Delay-Aware Fountain codes (DAF) [1] are proposed. It is based on sliding window schemes. However, different from the existing ones, DAF codes do not treat the sliding windows as homogeneous. By adaptively selecting the window length and optimally adjusting the sampling pattern according to the ongoing video bit rate, DAF codes deliver significantly higher video decoding ratio than existing schemes.

This paper will introduce the basic concepts of DAF. Also, some strategies that apply to DAF will also be briefly introduced, in order to provide more functionalities, such as real-time video streaming [2] and unequal error protection (UEP).

2. Delay-Aware Fountain Codes

DAF codes [1] are designed for streaming video over lossy wireless network. The objective is to encode videos (which are source-coded with unstable bit rates) over the network, so that the channel data rate is constant. It is achieved without using rate control, which will degrade the quality of videos. Instead, by making use of the coding structure of sliding window, DAF is able to provide stable video quality while maintaining constant channel data rate.

Basic concepts of DAF.

The basic idea of DAF codes is to segment the video file into overlapping windows, and then encode and send them consecutively with fountain codes. While the non-overlapping block coding scheme has a relatively small block size, the overlap between sliding windows allows the decoded packets in a previous window to help the decoding of future windows. By doing so, the block size is virtually extended.

Within each window, the fountain codes algorithm randomly chooses the native packets and combines them into a coded packet, according to degree distribution and sampling probability. Because the windows are overlapping, the total sampling probability of a frame is related to all the windows that covers it, as defined in (1).

$$ASP(t) = \frac{1}{n(t)} \sum_{\omega \in \text{all windows cover frame } t} \left(\sum_{i=1}^{n(t)} p_{\omega}(i) \right), \quad (1)$$

where $n(t)$ denotes the number of packets in frame t , and $p_{\omega}(i)$ denotes the sampling probability of i^{th} packet in window ω . So, $ASP(t)$ denotes the total probability accumulated on every packet in frame t through all the sliding windows covering that frame.

However, with the sliding window, even if the sampling distribution of every window is uniform, the overall ASP may still be nonuniform. As a result, in order to obtain a uniform overall ASP, the sampling probabilities in each window need to be adjusted.

Slope-only description for sampling distributions.

A proper representation of the sampling distribution is required. Storing the entire sampling distribution in each packet header would be impractical due to both communication and computation overhead. Alternatively, the *slope-only* description is proposed: the distribution is approximated by a linear function defined by the slope factor a . The

slope factor is a real number ranging from -1 (forming a backward triangular distribution like the green line in Fig. 1) to 1 (forming a forward triangular distribution like the red line in Fig. 1). When $a = 0$, it represents a uniform distribution like the blue line in Fig. 1.

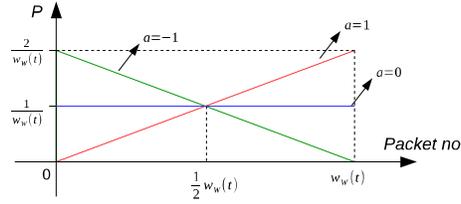


Figure 1. The distribution functions when slope factor $a = 1$, $a = 0$, and $a = -1$.

Given bit rates \mathbf{s} and slope factors \mathbf{a} , the resulting ASP for each frame can be computed by (2).

$$ASP_{\mathbf{s},\mathbf{a}}(t) = \mathbf{B}_{\mathbf{s}}(t) \cdot \mathbf{a} + D_{\mathbf{s}}(t), \quad (2)$$

where “ \cdot ” denotes the dot product of the two vectors of $(T - W + 1)$ elements. $\mathbf{B}_{\mathbf{s}}(t)$ and $D_{\mathbf{s}}(t)$ are the vector and value that only relevant to t given bit rates \mathbf{s} . Their definitions are omitted due to space limitation.

Optimization solution for DAF.

If we assume that each packet has the equal importance, the objective is to minimize the variance of ASP. Given the total number of frames T , the window size W , and number of packets in each frame $s(t)$, DAF codes want to find a set of slope factors \mathbf{a} , for which the variance of the sampling probabilities of all packets attains its minimum value. The optimization problem is defined in (3).

$$\arg \min_{\mathbf{a}} \sum_{t=W}^{T-W+1} (ASP_{\mathbf{s},\mathbf{a}}(t) - \overline{ASP}_{\mathbf{s},\mathbf{a}})^2 \quad s. t. \quad -1 \leq a_t \leq 1, \forall t, \quad (3)$$

where $\overline{ASP}_{\mathbf{s},\mathbf{a}} = \frac{1}{T-sW+2} \sum_{t=W}^{T-W+1} ASP_{\mathbf{s},\mathbf{a}}(t)$.

3. MPC-based DAF

Because the original DAF scheme has the disadvantages of high computational complexity and incapable of coping with real-time video application, a low-complexity and online version of DAF, MPC-based DAF, is proposed in [2]. In order to have an online algorithm, the optimized slope factor for the current sliding window should be generated in real-time, given the history of previous sampling distributions. We denote τ as the index of current time, and denote the slope factor for the current sliding window as \mathbf{a}_{τ} .

The basic idea is to limit the optimization process to a finite number of frames, or horizon, denoted as H . Then, Model Predictive Control (MPC) technique is used to compute the optimal slope factor for the current window. The objective function is similar to the global optimization problem in (3), which minimizes the variance of ASP. With a finite horizon imposed, the objective function is limited to a local range as in (4).

$$\arg \min_{\mathbf{a}_{\tau}^H} \sum_{t=\tau}^{\tau+T+W-2} (P_{\mathbf{s},\mathbf{a}_{\tau}^H}(t) - \overline{\mathbf{P}}_{\mathbf{s},\mathbf{a}_{\tau}^H})^2 \quad s. t. \quad -1 \leq a_t \leq 1, \forall t, \quad (4)$$

where \mathbf{a}_{τ}^H denotes the H -length slope vector for the windows starting from τ . \mathbf{P} denotes the vector of ASP in the range of $[\tau, \tau + H + W - 2]$, including both past and predictive values. $\overline{\mathbf{P}}_{\mathbf{s},\mathbf{a}_{\tau}^H}$ is the average value over \mathbf{P} . \mathbf{P} is the combination of the three components:

$$\mathbf{P}_{\mathbf{s},\mathbf{a}_{\tau}^H} = \mathbf{P}_{init} + \tilde{\mathbf{P}}_{\mathbf{s},\mathbf{a}_{\tau}^H} + \hat{\mathbf{P}}_{\mathbf{s}}. \quad (5)$$

In a nutshell, \mathbf{P}_{init} denotes the initial ASP that was already sampled by previous windows; $\tilde{\mathbf{P}}_{\mathbf{s},\mathbf{a}_{\tau}^H}$ denotes the range where ASP are affected by the currently optimized slopes within the horizon; $\hat{\mathbf{P}}_{\mathbf{s}}$ denotes the range where ASP will be affected by the slopes out of the horizon range in the future.

With the proposed scheme, the complexity is lowered to a linear algorithm for time T . Besides, the scheme does not require the knowledge of the bit rate outside the H -length horizon, which enables MPC-based DAF to cope with real-time video streaming.

4. UEP-based DAF

Because the packets in video data do not typically have uniform importance, a very natural extension of DAF with equal error protection (EEP) would be unequal error protection (UEP). In DAF, because the sampling probability is already non-uniform by design, once the importance values for each frame (denoted as $I(t)$) is provided, we can achieve UEP by simply adjusting the objective of optimization problem defined in (3) into the function in (6).

$$\arg \min_a \sum_{t=W}^{T-W+1} \left(ASP_{s,a}(t) - ASP_{s,a}^{TAR}(t) \right)^2$$

$$s. t. \quad -1 \leq a_t \leq 1, \forall t, \quad (6)$$

where $ASP_{s,a}^{TAR}(t)$ represents the target ASP for every frame, which could be computed by (7).

$$ASP_{s,a}^{TAR}(t) = \overline{ASP}_{s,a} \times I(t) / \bar{I}, \quad (7)$$

where $\overline{ASP}_{s,a}$ and \bar{I} represent the average values of ASP and importance profile, respectively, over the current sliding window.

5. Conclusions

This paper introduced a novel delay-aware fountain code scheme for video streaming, which deeply integrates channel coding and video coding. This is the first work to exploit the fluctuation of bit rate in video data at the level of channel coding, and to incorporate it towards the optimal design of video streaming-oriented fountain codes. Based on this idea, we developed three coding strategies: EEP-DAF, MPC-DAF and UEP-DAF. The simulation results in the corresponding articles show that the decoding ratio of our scheme is 15% to 100% higher than the state-of-the-art delay-aware schemes in a variety of settings. The PSNR of the videos transmitted by proposed schemes are also the highest among existing algorithms.

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Enhancing Capacity for Multimedia Communications in Internet of Things

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1. Introduction

The rapid adoption of the Internet of Things (IoT) in various networking scenarios brings in higher performance demand for multimedia applications in IoT. However, communications in typical IoT environments often provide low throughput due to limited available bandwidth, especially in wireless mobile and ad hoc sensor networks. Therefore, enhancing transmission capacity using limited bandwidth becomes a key issue to support multimedia communications in IoT. Device-to-Device (D2D) communications [1] allow users within a short range to engage in direct communications by reusing cellular resources, thus may significantly enhance the network capacity. Multi-hop over relay nodes is often employed in order to overcome the distance limitation of D2D communications. However, multi-hop relaying may potentially form a bottleneck that limits D2D throughput.

In this letter, we tackle this problem by employing the network coding technique. Network coding [2], by allowing relay nodes to encode arrival packets before forwarding them, offers a promising approach to further enhancing network capacity in order to support multimedia communications in IoT. To this end, we propose a model for D2D communications with a class of relay nodes running network coding and evaluate capacity improvement introduced by network coding especially for D2D communications with data retransmission.

2. System model

We consider a one-to-many, long-distance, relay-assisted, two-way communication model in a single cell. As shown in Fig. 1, UE 1 wishes to establish a D2D communication with all of the destination users UE 2, UE 3, ..., UE d , and the distance between UE 1 and any UE i , $i \in \{2, \dots, d\}$ is beyond the maximum range allowed by D2D communications. Due to the long distance between the source user and the destination users, some relaying users (Relay 1, Relay 2, ..., Relay k) between UE 1 and the destination users are needed to help them to accomplish the communication. At these relaying nodes, packets are first encoded and then transmitted to end users using the standard techniques of half duplex and frequency-division multiplexing.

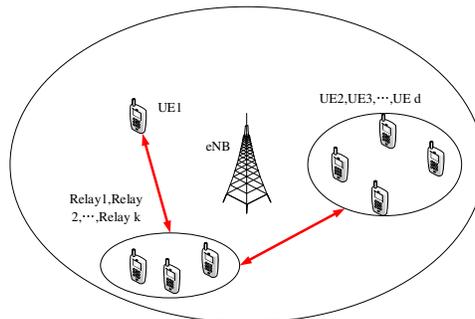


Fig. 1. System model.

The *objective* of this work is to enhance the system capacity while maintaining the required communication quality.

3. Network coding based solutions

Substantial results in network coding have been obtained since 2003. For instance, Li, Yeung, and Cai [3] proved that the upper bound of multicast problems with one or more sources can be achieved with the scheme of linear network coding, and Ho et al. [4] showed how random network coding can be used in broadcast networks where links are not error-free. Due to the notable advantage of random linear network coding in the sense that it does not need to know the topology of the network and is able to deal with the dynamic changes of links in networks, we adopt the random linear network coding as the coding scheme for relaying nodes in our system model. The essential idea of random linear network coding is that messages will be encoded by linear combinations of inputs with randomly chosen coefficients, and be decoded by solving a system of linear equations by the means of Gaussian Elimination.

Packet Retransmissions

To ensure the correctness of data transmission in wireless broadcast communications, every receiver must correctly receive all packets sent by the sender. As such, some error-control mechanisms, such as Automatic Repeat-Request and Forward Error Correction, must be employed to enhance the reliability of wireless broadcast. This makes the study on the efficiency of packet retransmissions particularly important in wireless networks. Recently, the technique of network coding has started being used in reducing the number of packet retransmissions in wireless communications, and remarkable results have been reported. For instance, Xiao et al. [5] suggested to reduce the number of information packet exchanges by using network coding; Katti et al. [6] proposed a new wireless mesh network architecture COPE using the idea of network coding, and demonstrated that it can remarkably reduce the average number of data transmissions by experiments on a 20-node 802.11a testbed; and Nguyen [7] et al. presented a network coding assisted packet retransmission strategy for the scenario of two receivers.

A notable work on utilizing the technique of network coding in reducing the number of data retransmissions in wireless communications is Nguyen et al.'s study [7], where they examined four different scenarios of packet retransmissions in which the network coding technique is either used or not used. In their first scenario of packet retransmission, scheme A, where the network coding is not utilized, if a receiver misses a packet in the current time slot, it will immediately send a negative acknowledgement (NAK) to the sender, regardless of whether it has correctly received the same packet in some previous time slots. In other words, all receivers are memoryless. Under this scheme, for any packet p , the sender must retransmit p until all receivers have correctly and simultaneously received p at some time slot. The second scenario, scheme B, does not utilize network coding either. In scheme B, a receiver signals a NAK to the sender only if it loses a packet in the current time slot and has never received this packet correctly before. Clearly, scheme B is an improvement over scheme A in that a receiver will not ask the sender to resend packets it has already received, which will lead a less number of total packet retransmissions. Unlike schemes A and B, schemes C and D leverage the technique of network coding. In scheme C, the sender does not immediately retransmit the required packet after receiving the NAK signal; rather, it maintains as a buffer a list of sent packets and their status (received or lost) with regard to all the receivers, and waits until N (a predefined number) packets have been sent and start the retransmission by XORing lost packets and broadcasting them. If an encoded (i.e., XORed) packet is lost at some receivers, then the sender will keep resending this encoded packet until all receivers have successfully received this encoded packet. Also, the decoding information is sent along with the encoded packet to all receivers. Scheme D is an improvement over scheme C in the sense that it dynamically changes the combination of packets to give rise to new encodings, based on the feedback information on retransmitted packets from receivers.

Although packet retransmission using network coding is able to reduce the number of retransmissions and improve the utilization rate of bandwidth, it is not free of issues. For example, the selection of N is a tricky issue: if N is too small, then the advantage of using network coding will not be shown; if N is too large, then more space will be required to store the packets before they are retransmitted, the complexity of decoding encode packets will be increased, and longer transmission delays will likely to occur.

4. Performance evaluation

Simulation experiments are conducted to evaluate the performance of the four different packet retransmission schemes discussed in the previous section. Fig2 shows the bandwidths consumed by the four schemes with respect to different number of receivers (up to 20), where the packet loss probability from a relaying node to any receiver is 0.05 ($p_i = 0.05, i = 1, 2, \dots, 20$). Clearly, schemes C and D (in which the network coding technique is used) require much less bandwidths than schemes A and B (in which the network coding technique is not used), and this advantage becomes more obvious with the increase of number of receivers. The reason that there is little change in required bandwidths for schemes C and D over the increase of the number of receivers is the assumption that the signal losses encountered from the relay node to all receivers are the same.

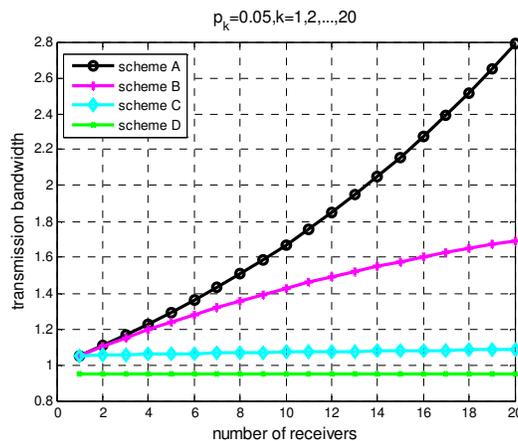


Fig 2. Required bandwidths of schemes A, B, C, and D.

Fig. 3 further demonstrates the advantage of network coding technique over the traditional method by comparing the performances of schemes B and D in terms of packet retransmissions.

In Fig 3, network coding gain is defined to be the ratio of the bandwidth consumed by scheme B to that by scheme D, and is used as the evaluation yardstick. Also, only two receivers are considered in this simulation, and p_1 and p_2 are the packet loss probabilities incurred at the respective links to these two receivers. Obviously, all three curves in Fig 3 show that scheme D requires less bandwidth than scheme B. Moreover, the network coding gain reaches its pinnacle when $p_1 = p_2$. This is due to the fact that the number of lost packets for these two receivers are the same at this moment, which maximizes the number of encoded packets at the relaying nodes and thereby reduces the number of packet retransmissions.

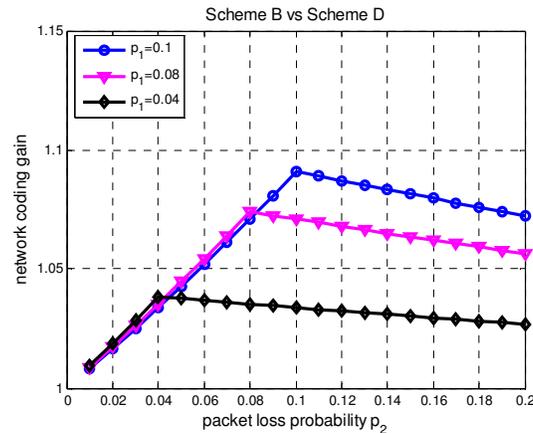


Fig 3. Network coding gain over traditional packet retransmission method.

5. Conclusion

In this letter, we have presented a network-coding-based multi-hop D2D communication model to address the multimedia communications in the IoT, reviewed applications of network coding in wireless broadcasting, and emphasized the feasibility of encoding packets by network coding at relaying nodes. We have investigated four different and paradigmatic packet retransmission schemes in the literature that are either equipped with the technique of network coding or not. Our simulation results clearly indicate that packet retransmission schemes with network coding outperform the ones without network coding, which may shed light on and invoke future research work in this area.

Acknowledgement

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Toward Secure Transportation Internet of Things: A Trust Management Perspective

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1. Introduction

According to a report released by World Health Organization [1], the number of road traffic deaths globally has plateaued at 1.25 million a year, which makes road traffic injuries a leading cause of death globally. Hence, it is critical to improve road safety and enhance the efficiency of transportation systems. For instance, the exchange of traffic-related information in a timely and accurate fashion in road transportation system is critical to accident prevention because prior knowledge of future collisions in as little as one-half second before an actual accident, could lead to a decrease in traffic accidents by as much as 60% [2]. Therefore, the concept Internet of Things (IoT) has been emerged recently as a new paradigm which could revolutionize the transportation systems, in which various devices, such as vehicles, road-side sensors, handheld mobile devices are interconnected together so that they can collect and exchange transportation related data in real-time fashion.

However, the exchange of information also makes the system prone to attacks and more general security-related issues. For example, in July 2015, automotive cyber security researchers Charlie Miller and Chris Valasek demonstrated to Wired Magazine about how they could remotely and wirelessly hack into a Jeep Cherokee from 10 miles away while it was on the highway [3]. Their demonstration led Chrysler to recall 1.4 million affected vehicles, the first known automotive recall for a cybersecurity vulnerability [4].

To address the security issues in transportation IoT, trust management has been studied to enhance the resilience to various security threats, which aims to evaluate different behaviors of other nodes and build a trust for each node based on the observations made for behavior assessment, such as the research works discussed in [5], [6], [7].

In this paper, we will discuss a trust management scheme which is able to better secure transportation IoT.

2. Secure Transportation IoT Using Trust Management

2.1 System Preliminaries

In general, the *trustworthiness* of a node N_K can be defined as a vector $\Theta_K = (\vartheta_K^{(1)}, \vartheta_K^{(2)}, \dots, \vartheta_K^{(N)})$, in which $\vartheta_K^{(i)}$ stands for the i -th dimension of the trustworthiness for the node N_K . Each dimension of the trustworthiness $\vartheta_K^{(i)}$ corresponds to one or a certain category of behavior(s) $B_K^{(i)}$ (such as packet forwarding or true recommendation sharing), and $\vartheta_K^{(i)}$ can properly reflect the probability with which the node will conduct $B_K^{(i)}$ in an appropriate manner. $\vartheta_K^{(i)}$ can be assigned with any real value in the range of $[0, 1]$, i.e., $\forall i \in \{1, 2, \dots, n\}, \vartheta_K^{(i)} \in [0, 1]$. The higher value of $\vartheta_K^{(i)}$, the node N_k is more likely to conduct $\vartheta_K^{(i)}$ properly.

Each dimension of the trustworthiness $\vartheta_K^{(i)}$ for the node N_K is defined as a function of the misbehaviors $M_K^{(i)}$ that are related to $B_K^{(i)}$ and have been observed by the neighbors of the device N_K . Different dimensions of the trustworthiness may correspond to different functions, and the selection of different functions should coincide with the basic features of $M_K^{(i)}$, such as severity of the outcome, occurrence frequency, and context in which they occur.

In particular, the trustworthiness of a device is represented in a vector $\Theta_K = (\vartheta_K^{(1)}, \vartheta_K^{(2)})$, and each element in the vector stands for *functional trust* and *recommendation trust*, respectively. In the future, if it is necessary to introduce new element to the trust vector, the new element can be added easily.

2.2 Adversary Model

First of all, the Road-Side Units (RSUs) are assumed to be trustworthy since they are usually better protected. The connected vehicles, on the other hand, are generally more susceptible to various attacks, and they can be compromised at any time after the vehicular network is formed.

The adversary can be an outsider located in the wireless range of the vehicles, or the adversary can first compromise one or more vehicles and behave as an insider later. The adversary is able to eavesdrop, jam, modify, forge, or drop the wireless communication between any devices in range. The main goals of the adversary may include intercepting the normal data

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transmission, forging or modifying data, framing the benign devices by deliberately submitting fake recommendations, etc. More specifically, the following malicious attacks are considered in this paper.

- Simple Attack (SA): An attacker may manipulate the compromised nodes not to follow normal network protocols and not to provide necessary services for other nodes, such as forwarding data packets or propagating route discovery requests. However, the compromised node will not provide any fake trust opinions when it is asked about other node's trustworthiness.
- Bad Mouth Attack (BMA): In addition to conduct simple attack, the attacker can also spread fake trust opinions and try to frame the benign nodes so that the truly malicious nodes can remain undetected. This attack aims to disrupt the accurate trust evaluation and make it harder to successfully identify the malicious attackers.
- Zigzag (On-and-off) Attack (ZA): Sometimes sly attackers can alter their malicious behavior patterns so that it is even harder for the trust management scheme to detect them. For instance, they can conduct malicious behaviors for some time and then stop for a while (in that case the malicious behaviors are conducted in an on-and-off manner). In addition, the sly attackers can also exhibit different behaviors to different audiences, which can lead to inconsistent trust opinions to the same node among different audiences. Due to the insufficient evidence to accuse the malicious attacker, it is generally more difficult to identify such sly attackers.

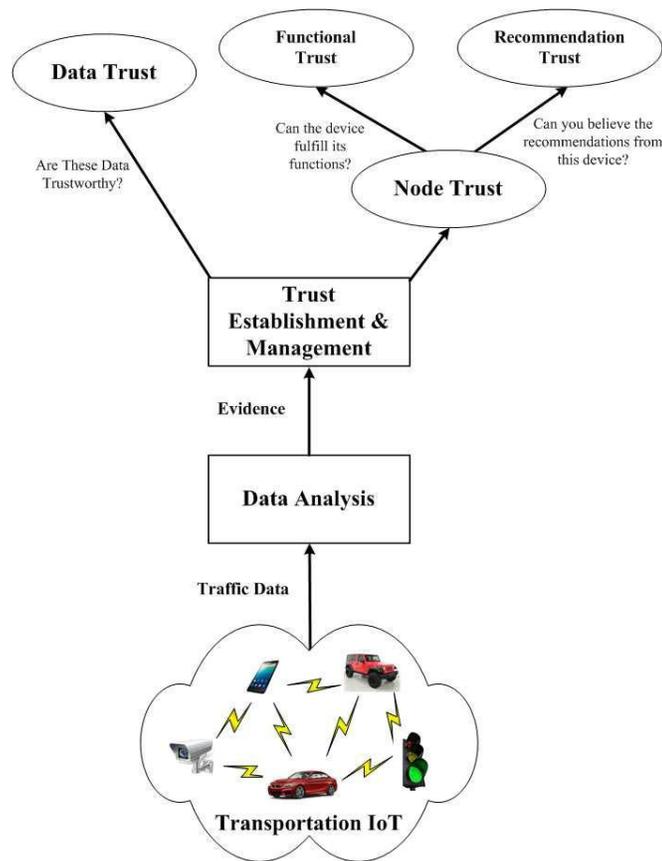


Fig. 1. Trust Management Scheme to Secure Transportation IoT

2.3 Scheme Overview

The trust management scheme is composed of two phases, namely data analysis and trust management. The schematic diagram of the scheme is depicted in Figure 1.

In the trust management scheme, we first collect traffic data from VANETs for data analysis. Second, we summarize the findings from the data analysis as evidences for trust management schemes to evaluate the trustworthiness. For the data analysis part, Dempster-Shafer Theory (DST) can be used to fuse together multiples of sensory data [8]. We then use the Collaborative Filtering technique to evaluate the trustworthiness of each device in transportation IoT [9].

3. Conclusion

It is well understood that Internet of Things (IoT) has recently emerged as a new paradigm which can deeply change every aspect of our daily lives, including daily transportation needs. However, transportation IoT may experience various security threats which can possibly prevent its wide deployment. In this paper, we study of the problem of how to better secure transportation IoT via trust management approach.

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CFP: Special Issue for IEEE Transactions on Multimedia

Video over Future Networks: Emerging Technologies, Infrastructures, and Applications

The current Internet faces numerous challenges as a platform for media delivery: the high bandwidth demand is fueled by online video streaming services and the advent of ultra-high-definition (UHD) video; widespread use of social media fosters instant sharing of user-generated video; rising consumer interests in augmented and/or virtual reality (AR/VR) underscore the need to support richer media forms with lower latency. In the meantime, design and experimentation of the Internet itself is evolving at its own pace, as demonstrated by recent advances in software-defined networking (SDN), network function virtualization (NFV) technologies, and information-centric networking (ICN) --- technologies with potentially far-reaching impacts on the future Internet. This special issue aims to highlight research works that investigate future Internet technologies through the prism of its most prevalent application: video distribution. Intriguing research questions abound: how can named-data-networking (NDN) support live video streaming? What is the most efficient distribution mechanism for social sharing of user-generated video? What are proper performance metrics for novel networked multimedia applications based on augmented/virtual reality?

We invite submissions of high-quality papers on either original research or survey/overview, which have not been published previously. Topics of interest include, but are not limited to:

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- Network-assisted rate adaptation in SDN-enabled networks
- In-network caching and caching for mobile video delivery
- Video distribution over information-centric networking (NDN) architectures
- Cost and economic models for video distribution over future networks
- Network support for emerging novel applications, e.g., based on augmented reality (AR)
- Networking and distributed systems for augmented reality (AR)
- Distribution of ultra-high-definition (UHD) video over next-generation networks
- Distribution of user-generated media content over future Internet
- Network and cloud support for real-time video analytics
- Integration of video distribution and multimedia computing

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IEEE COMSOC MMTC Communications - Frontiers

MMTC Communication - Frontier CFP: Content-Driven Communications and Computing for Multimedia in Emerging Mobile Networks

Due to continuing advances in wireless communications and mobile devices, we are entering an era of rapid expansion in multimedia applications and services called as multimedia -based services, such as video streaming (Youtube, Netflix) and content sharing (Instagram, Snapchat). Obviously, these multimedia -based services are content-centric and dominant driving forces behind the expansion. Today's mobile network architectures, however, are designed to be connection-centric, which have become a barrier to meet the diverse application requirements and the quality expectation of the end users. Although the bandwidth and data rate increase, current mobile networks are still facing poor user experience and low service quality.

A paradigm shift is needed to meet the proliferation of content-centric services. On the one hand, the developments of multimedia transmission systems and services call for new understanding and evaluation of user's perceived quality of experience (QoE). There is tremendous demand for objective, online QoE prediction and monitoring for content-centric services. The large-scale data sets of online video streaming and content sharing have made it possible to reveal the true relationship between QoE and traditional metrics as well as network conditions. On the other hand, the increasing data rates, bandwidths, as well as processing capabilities of base stations and user end devices lays the foundation for efficient and massive content distribution. It is envisioned that content-driven communications and computing technologies will break the bottleneck of current connection-centric network architectures, and lead to a clean-slate redesign of network architecture.

Topics of interest include, but are not limited to:

- Content-driven heterogeneous multimedia networks architecture
- Content-driven software-defined multimedia networks architecture
- Content-driven communications technologies in wireless multimedia networks
- Content-driven computing technologies in mobile multimedia networks
- Content-driven caching for multimedia in wireless networks
- QoE-aware content delivery for multimedia in mobile networks
- QoE-aware 5G multimedia network architecture/techniques and performance analysis
- QoS-QoE relationship modeling
- Real experiments and testbeds for QoE evaluation
- Efficient video codec design with guaranteed QoE

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