Optimal Radio Access for Fully Packet-Switching 5G Networks

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***Abstract*—In the following decades, traffic volumes and the number of devices are projected to increase a ten to thousand-fold. Unfortunately, existing cellular networks adopting closed-loop communications impose too large spectrum overheads, which lead to an unacceptably low spectrum efficiency and unaffordable traffic burdens. To support the extremely challenging and unavoidable massive data exchanges in 2020 and beyond, we fundamentally re-consider an efficient scheme of the fully packet-switching radio access, that is, open-loop communications. To eliminate the concerns on the capabilities of enhancing the spectrum efficiency and reliability to support multimedia transmissions via open-loop communications, in this paper, we develop the optimum transmission repetition scheme to maximize the resource utilization while providing quality-of-service (QoS) guarantees. Our results confirm the efficiency, effectiveness, and practicability of the fully packet-switching radio access as compared with existing closed-loop communications, which suggest a revolutionary system design as the foundations for the fifth generation (5G) networks.**

I. Introduction

IT is predicted that, in 2020, there will be more than 50 billion of heterogeneous devices relying on cellular networks for data exchanges. As compared with only six billion devices in 2011, there will be a ten-fold increase on the number of devices. In addition, it is further expected a thousand-fold increase in traffic volume in 2020 and beyond. To service the projected amount of connected devices and traffic volume, the spectrum efficiency has to be increased a thousand-fold. Unfortunately, the capabilities offered by the fourth generation (4G) network (i.e., LTE-A) [1], currently being deployed worldwide where each device is capable of achieving data rates from the order of hundreds Mbps to several Gbps, may not be enough to support such a heavy burden. The critical challenge lies in the extremely poor spectrum efficiency provided by LTE-A.

Reviewing the history on the development of digital cellular systems from the the third generation (3G) networks (UMTS) [2] to the 4G networks (LTE/LTE-A) [1], the "circuit-switching" operations are adopted in the air interface. Circuit-switch in the air interface suggests *closed-loop* communications to maintain the existence of the wireless link. In closed-loop communications, the optimum (downlink/uplink) data transmission schemes (i.e., dynamic link adaptation, scheduling, and radio resource allocation) for a transmitter is decided after (uplink/downlink) feedback information from the receiver is obtained. For example, the inner-loop power control is performed 1500 times per second in UMTS [2], while the channel estimation is performed 1000 times per second in maximum in LTE/LTE-A [1]. In addition, automatic repeat request (ARQ) and hybrid ARQ (H-ARQ) are adopted in LTE/LTE-A, which need acknowledgement messages sent from a receiver to a transmitter to ensure a successful data transmission. Although each feedback message only contains very few bits, each transmitter needs to spend an extremely large fraction of time in waiting for updated feedback information. As a result, high data rate transmissions for each transmitter-receiver pair may only take place in a very limited fraction of time. Considering the ten-fold increase in the number devices, the existing closed-loop communications invoke unacceptable overheads to severely degrade spectrum efficiency. This fossilized paradigm is the vital defect in UMTS/LTE/LTE-A, which motivates us to forage an unprecedented design.

To significantly enhance the spectrum efficiency, a revolutionary design lies in fully packet-switching radio access in the air interface, which suggests *open-loop* communications. That is, a receiver does not provide any feedback information to the transmitter (i.e., channel state information, acknowledgement, etc.), no matter whether transmitted data is successfully received by the receiver or not. By avoiding feedback overheads, a transmitter does not need to wait for feedback message to save a huge fraction of time. Thus, the spectrum efficiency can be substantially enhanced. However, there are two critical concerns for *open-loop* communications. (i) Without feedback, all link adaptation schemes needing feedback messages are unavailable. As a result, the transmitter may not increase the data rate when the channel quality improves. Thus, whether the spectrum efficiency is actually enhanced is questionable. (ii) A transmitter has no knowledge about the successful/failure reception at the receiver. To provide reliable communications combating fading channels, an effective scheme lies in the transmission repetition [3]–[5], which suggests that a transmitter utilizes duplicates of radio resources for each packet transmission. The repetition can take place in the time domain (packet transmissions are repeated at multiple time slots), in the frequency domain (packet transmissions are repeated at multiple frequency bands), or in the spatial domain (packet transmissions are repeated at multiple communication paths). However,

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the transmission repetition imposes additional overheads, which may also degrade the spectrum efficiency.

To practice the fully packet-switching radio access in the air interface for the 5G networks, above two critical concerns need to be eliminated. In this paper, we shall consequently optimize the performance in the fully packet-switching radio access, in terms of spectrum efficiency and the capability of providing quality-of-service (QoS) guarantees for multimedia (voice/video) sources. By proposing an autonomous scheme for each transmitter to determine the optimum amount of radio resources for the transmission repetition in the time domain and the spatial domain, we show that (i) QoS guarantees can be achieved via open-loop communications, and (ii) open-loop communications provide an outstanding spectrum efficiency as compared with that of the closed-loop scheme (i.e., H-ARQ). This result provides a breakthrough confirmation that the fully packet-switching radio access is a broad way fundamentally enhancing the performance in UMTS/LTE/LTE-A toward the 5G networks.

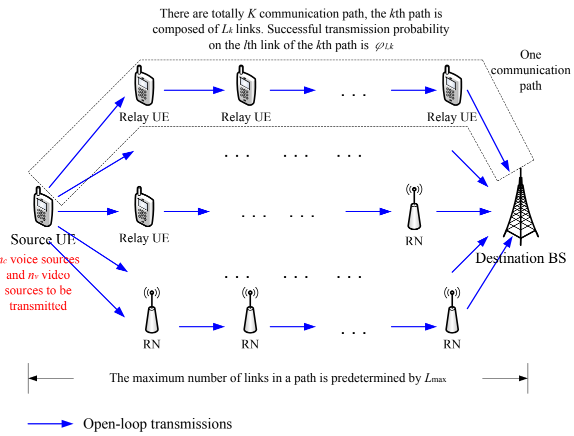

Fig. 1. To form transmission repetition in the spatial domain, multiple UEs (and/or RNs) can form K communication paths between a source UE and a destination BS. Each path can be composed of L_k links ($L_k \leq L_{max}$). By scrambling a transmitter-specific identity with data at each transmitter UE (or RN), each receiver UE (or RN) of each link is able to reject interference from transmitters of other links. Consequently, interference among links and paths can be rejected. The lth link of the kth path suffers from a failure probability $1 - \varphi_{l,k}$.

II. System Model

Considering the uplink transmissions in open-loop communications, a receiver base station (BS) is unable to provide any feedback message to a transmitter user equipment (UE). To combat fading channels, the transmission repetition is adopted. As mentioned above, the repetition can take place in the time, frequency, and spatial domains. For QoS guarantee provisioning, transmission repetitions in the frequency domain and in the spatial domain may be homogeneous. Thus, without loss of generality, transmission repetitions in the time domain and the spatial domain are considered. Due to the lack of knowledge about instantaneous channel condition at the receiver side, the first effort in the standardization progress is to proceed to the link budget calculation. Based on the calculation result, a predetermined modulation and coding scheme (MSC) is adopted by each transmitter UE of each link to provide a targeted successful (transmission with an acceptable error rate) probability φ at each subframe, and a consequently failure (transmission with an unacceptable error rate) probability $1 - \varphi$. The predetermined MCS also results in a information rate of R bis/subframe.

To form the transmission repetitions in the spatial domain, each transmission from a source UE can be received by multiple UEs (or relay nodes, RNs). These UEs (or RNs) can subsequently relay the received packet(s) to neighboring UEs (RNs). By adopting the relay construction protocols (e.g., [6]), multiple UEs (and/or RNs) are able to form K communication paths (indexed by $k = 1, ..., K$) between a source UE and a destination BS, as shown in Fig. 1. Each path can be composed of L_k links ($L_k \leq L_{max}$, where L_{max} is the predefined maximum number of links in a path). Considering the case of only one path between a source UE and a receiver BS, if severe channel fades occur at the path, it may result in continuous transmission failures. Consequently, timing constraints of multimedia transmissions may be violated. This unfavorable case can not be avoided in the single path transmission case. However, if there are multiple communication paths, duplicates of each packet can be transmitted simultaneously via multiple paths. Then, the timing constraint is violated only if packet transmissions fail in all paths. As a result, the probability of QoS constraint violation can be significantly alleviated.

In open-loop communications, the receiver of each link knows the existence of the transmitter, while the transmitter does not know the existence of a receiver. Therefore, each transmitter UE (or RN) needs to scramble a transmitter-specific identity with the transmitted packets at each link. By the scrambled transmitter-specific identity, each receiver UE (or RN) of each link is able to reject interference from transmitters of other links. Consequently, interference among links and paths can be rejected [7]. In the source UE, two key types of multimedia traffic are transmitted: real-time voice and real-time video. Considering n_c voice sources (indexed by $i = 1, ..., n_c$) and n_v video sources (indexed by $j = 1, ..., n_v$) of the same packet size χ, each voice and video source can be characterized by the following parameters.

S1) The ith voice source is characterized by three parameters (λ_i, δ_i, ε_i), where λ_i is the packet arrival rate of the source, δ_i is the maximum tolerable jitter, and ε_i is the acceptable jitter constraint violation probability. Packets of the ith voice source are generated periodically every $1/\lambda_i$ subframes and are stored in a ready-to-transmit (RTT) buffer for this voice source. Jitter is defined as the difference between the time of two successive packet departures and the time of two successive packet arrivals. A voice source with a higher λ_i has a higher priority.

S2) The jth video source is characterized by four parameters (ρ_j, σ_j, d_j, ξ_j), where ρ_j is the average packet arrival rate of the jth video source, σ_j is the maximum burstness (the maximum number of packets in one arrival), d_j is the maximum tolerable delay, and ξ_j is the acceptable delay constraint violation probability. The jth video source regulated by the (σ_j,ρ_j)-leaky bucket is stored in a RTT buffer for this video source. Video sources are with bulk arrivals (that is, multiple packets from upper layers may arrive at the same time). Data is decodable at the destination BS only when the entire bulk of packets are successfully received before the expiration of d_i. Video sources with a smaller d_i have a higher priority.

III. Optimum Fully Packet-Switching Radio Access

Since each transmitter UE (or RN) of each link lacks the knowledge about the channel condition and the existence of the receiver, transmission repetitions have to be performed both in the time and the spatial domains. This principle drives the proposed optimum autonomous scheme as follows.

1) At the end of a packet forwarding period, the source UE scans all its RTT buffers of voice sources. If a packet of a voice source with the highest present priority is found, the source UE transmits at most one packet from this RTT buffer.
2) At the end of a packet forwarding period, if there is no packet arrival in all RTT buffers of voice sources, the source UE then scans its RTT buffers of video sources. If a bulk of packets of a video source with the highest present priority are found, the source UE transmits all packets from this RTT buffer.
3) When a voice packet or a video packet is forwarded from the source UE, duplicates of this packet are simultaneously forwarded via multiple communication paths.
4) For all voice and video packets, it takes $S = \lceil \chi/R \rceil$ subframes to forward a packet through a link. However, since each transmitter UE (or RN) of each link lacks the knowledge about the channel condition, each transmitter UE (or RN) of each link repeats the packet transmission for τ_c/L_{max} subframes ($\tau_c/L_{max} \geq S\rceil$) for each voice packet, and repeats the packet transmission for $\sigma_j\tau_c/L_{max}$ subframes for a bulk of video packets (as σ_j is the maximum burst of a packet arrival). In other words, the source UE reserves τ_c subframes for each voice packet and reserves $\sigma_j\tau_c$ subframes for each bulks of video packets. These reserved subframes are referred to as a *packet forwarding period* in Step 1 and Step 2.
5) Each receiver UE (or RN) of each link relays a voice packet (or a bulk of video packets) to the subsequent UE (or RN) if the voice packet (or the bulk of video packets) is completely received.

It is expected that each voice packet and each bulk of video packets can be forwarded to the destination BS within τ_c and $\sigma_j\tau_c$ subframes, respectively. However, since the packet transmission over the lth link of the kth path suffers from a failure probability $1 - \varphi_{l,k}$, the exact number of subframes required to forward a packet to the destination BS is unclear. Our design goal is consequently to determine τ_c and the number of utilized paths k_a for transmission repetitions to maximize the resource utilization efficiency while providing QoS guarantees for all voice/video sources. This goal can be achieved by solving the following optimization at the source UE.

Definition 1. *The true end-to-end packet forwarding time (in terms of subframes) of a voice packet via the kth communication path, denoted by Θ_c^k, is the sum of the number of subframes of successful transmissions and the number of subframes of failure/redundant transmissions, to deliver a voice packet from the source UE to the destination BS. Thus, the true end-to-end packet forwarding time of a voice packet by leveraging k_a paths, denoted by Θ_c, is*

$$\Theta_c = \min\{\Theta_c^1, \ldots, \Theta_c^{k_a}\}. \tag{1}$$

Definition 2. *The true end-to-end packet forwarding time (in terms of subframes) of a bulk of video packets via the kth communication path, denoted by Θ_v^k, is the sum of the number of subframes of successful transmissions and the number of subframes of failure/redundant transmissions, to deliver a bulk of video packets from the source UE to the destination BS. Thus, the true end-to-end packet forwarding time of a bulk of video packets by leveraging k_a paths, denoted by Θ_v, is*

$$\Theta_v = \min\{\Theta_v^1, \ldots, \Theta_v^{k_a}\}. \tag{2}$$

Optimization 1. *The optimum QoS guaranteed broadcasting communication scheme for voice and video sources is mathematically formulated by*

$$\begin{aligned} &\min \tau_c k_a \\ \textit{s.t. } &\textit{(i) } \Pr[\Theta_c > \delta_i] \leq \varepsilon_i, \textit{ for } i = 1, \ldots, n_c, \\ &\textit{(ii) } \Pr[\Theta_v > d_j] \leq \xi_j, \textit{ for } j = 1, \ldots, n_v, \\ &\textit{(iii) } 0 \leq k_a \leq K. \end{aligned} \tag{3}$$

The objective in (3) minimizes the total amount of repetition resources (in the time domain τ_c and in the spatial domain k_a), under the timing constraints for voice and video sources in (i) and (ii), and the feasibility constraint of k_a in (iii). However, the feasibility constraint of τ_c is still unclear. To solve (3), we should study the relationship among τ_c, k_a, $\Pr[\Theta_c > \tau_c] \leq \varepsilon_i$ for all i, and $\Pr[\Theta_v > d_j] \leq \xi_j$ for all j.

Theorem 1. *By utilizing k_a communication paths, denote*

$$\delta_i^* = \tau_c + \sum_{g=1}^{i-1} \lceil \frac{\lambda_g}{\lambda_i} \rceil \tau_c, \; i = 1, \ldots, n_c, \tag{4}$$

If $\delta_i^* + \tau_c \leq 1/\lambda_i$ and $\delta_i^* < \delta_i$ for all i, the jitter constraint violation probability of the ith voice source is bounded above by $\bar{\Theta}_c/\tau_c$, where $\bar{\Theta}_c$ is the expected value of Θ_c.

Proof: Since packets of the ith voice source are generated periodically $1/\lambda_i$ subframes, by temporarily assuming $\Theta_c \leq \tau_c$, if we can show that the ith voice source has the maximum wait $\tilde{\delta}_i$, the jitter cannot be larger than $\tilde{\delta}_i$.

Furthermore, since each packet of voice sources is allocated by τ_c subframes, if $\tilde{\delta}_i + \tau_c < 1/\lambda_i$, the packet can be delivered to the destination BS before the next packet arrival at the source UE. We prove above arguments by induction with two hypotheses:

$$\text{i) } \tilde{\delta}_i \le \delta_i^*, \text{ and ii) } \tilde{\delta}_i + \tau_c < 1/\lambda_i. \tag{5}$$

Considering the first voice source, the maximum wait of a packet is $\tilde{\delta}_1 = \tau_c = \delta_1^*$ subframes. To ensure the packet of the first voice source to be delivered to the destination BS before the next arrival takeing place at the source UE, the sufficient condition is $\tilde{\delta}_1 + \tau_c < 1/\lambda_i$, which is our assumption $\delta_1^* + \tau_c < 1/\lambda_i$. Suppose that the induction hypotheses hold up to the $i-1$th voice source. We argue by contradiction that $\tilde{\delta}_i \le \delta_i^*$. Suppose $\tilde{\delta}_i > \delta_i^*$, voice sources $g = 1, \ldots, i-1$ must be served. From the induction hypothesis ii), every packet of these $i-1$ voice sources is served before the next packet arrival. Thus, the total number of packets that can be served within $(0,\delta_i^*)$ for these $i-1$ voice sources is at most $\sum_{g=1}^{i-1} \lceil \lambda_g \delta_i^* \rceil$. Therefore, the total number of subframes to serve these packets is bounded above by

$$\sum_{g=1}^{i-1} \lceil \lambda_g \delta_i^* \rceil \tau_c + \tau_c \tag{6}$$

Since $\delta_i^* < 1/\lambda_i$, the quantity in (6) is bounded above by

$$\sum_{g=1}^{i-1} \lceil \frac{\lambda_g}{\lambda_i} \rceil \tau_c + \tau_c = \delta_i^* \tag{7}$$

which follows the definition of δ_i^* in (4). Therefore, all k_a paths cannot always be busy in $(0,\delta_i^*)$ and we reach a contradiction. This shows $\tilde{\delta}_i \le \delta_i^*$ and the packets of the ith voice source will be forwarded to the destination BS before the subsequent packet arrival at the source UE. Above arguments are valid under the assumption $\Theta_c \le \tau_c$. If $\Theta_c > \tau_c$, the packet transmission may violate the maximum tolerable jitter constraint. This probability denoted by $\Pr[\Theta_c > \tau_c]$ is consequently bounded above by $\bar{\Theta}_c/\tau_c$. ■

Theorem 1 fully reveals the relationship between τ_c and $\Pr[\Theta_c > \tau_c]$, by utilizing k_a communication paths. To further incorporating video sources, the following theorem is provided. A video source (say the jth video source) is served by utilizing the remaining time domain resources after serving all voice sources and previous $j-1$ video sources. Therefore, the maximum delay of the jth video source, d_j^*, is affected by all voice sources and the maximum delays of previous $j-1$ video sources. Thus, a recursive form is adopted to provide the following theorem.

Theorem 2. *By utilizing k_a communication paths, recursively denote*

$$d_j^* = \frac{\Theta_v(1 + \sum_{g=1}^{j} \sigma_g + \sum_{g=1}^{j-1} \rho_g d_g^*) + \tau_c(1+n_c)}{1 - \tau_c \sum_{i=1}^{n_c} \lambda_i - \sum_{g=1}^{j-1} \rho_g \sigma_g \tau_c} \tag{8}$$

for $j = 1, \ldots, n_v$. If $\tau_c \sum_{i=1}^{n_c} \lambda_i + \sum_{g=1}^{j-1} \rho_g \sigma_g \tau_c < 1$, then the delay constraint violation probability of the jth video source is bounded above by $\bar{\Theta}_v/\varpi_j$, where $\bar{\Theta}_v$ is the expected value of Θ_v, and

$$\varpi_j = \frac{d_j(1 - \tau_c \sum_{i=1}^{n_c} \lambda_i - \sum_{g=1}^{j-1} \rho_g \sigma_g \tau_c) - \tau_c(1+n_c)}{1 + \sum_{g=1}^{j} \sigma_g + \sum_{g=1}^{j-1} \rho_g d_g^*}. \tag{9}$$

Proof: Let $C_1(t_1, t_2)$ be the number of subframes that can be allocated to the first voice source in an interval $(t_1,t_2]$. From the proof of Theorem 1, the maximum number of packets from n_c voice source that can be served in an interval $(t_1,t_2]$ is at most $\sum_{i=1}^{n_c} \lceil \lambda_i(t_2 - t_1) \rceil$. Applying the inequality

$$\lceil x \rceil \le x + 1 \tag{10}$$

yields the bound $\sum_{i=1}^{n_c} [\lambda_i(t_2 - t_1) + 1]$. Since our design is non-preemptive, the number of subframes that can be allocated to the first video source in $(t_1,t_2]$ is at least $t_2 - t_1 - \tau_c\{1 + \sum_{i=1}^{n_c} [\lambda_i(t_2 - t_1) + 1]\}$, thus,

$$C_1(t_1, t_2) \ge \left[1 - \tau_c \sum_{i=1}^{n_c} \lambda_i\right] (t_2 - t_1) - \tau_c(n_c + 1). \tag{11}$$

Note that the number of departures in $(t_1,t_2]$ from a (σ,ρ)-leaky bucket is bounded above by $\sigma + \lceil \rho(t_2 - t_1) \rceil$. Applying (10) yields the upper bound $\sigma + \rho(t_2 - t_1) + 1$. Let $A_1(t_1, t_2)$ be the amount of work load (number of subframes required for packets arriving at the RTT buffer) within the interval $(t_1,t_2]$ for the first video source. Then,

$$A_1(t_1, t_2) \le \Theta_v[\sigma_1 + \rho_1(t_2 - t_1) + 1]. \tag{12}$$

The delay of an arrival at time t is bounded above by $\inf\{d' \ge 0 : A_1(0,t) - C_1(0, t + d') \le 0\}$. Maximizing over t, we have

$$d_1^* = \sup_t \inf\{d' \ge 0 : A_1(0,t) - C_1(0, t + d') \le 0\}. \tag{13}$$

Applying the upper constraint of $A_1(t_1, t_2)$ and the lower constraint of $C_1(t_1, t_2)$, we obtain

$$d_1^* = \frac{\Theta_v(1 + \sigma_1) + \tau_c(1+n_c)}{1 - \tau_c \sum_{i=1}^{n_c} \lambda_i}. \tag{14}$$

If $d_1^* > d_1$, the maximum tolerable delay constraint of the first video source is violated. For the first video source,

$$\begin{aligned} \Pr[d_1^* > d_1] &= \Pr[\frac{\Theta_v(1 + \sigma_1) + \tau_c(1+n_c)}{1 - \tau_c \sum_{i=1}^{n_c} \lambda_i} > d_1] \\ &= \Pr[\Theta_v > \varpi_1] < \frac{\bar{\Theta}_v}{\varpi_1}, \end{aligned} \tag{15}$$

where ϖ_1 is defined in (9). This completes the argument for the first video source. The argument for the jth video source is essentially the same as that of the first video source. However, the lower constraint required to be modified since the jth video source utilizes remaining (time domain) resources from all voice sources and the first $j-1$ video sources. Parallel to the argument of the first video source, the maximum delay of the j video source is bounded above by $\frac{\Theta_v(1+\sum_{g=1}^{j} \sigma_g + \sum_{g=1}^{j-1} \rho_g d_g^*) + \tau_c(1+n_c)}{1 - \tau_c \sum_{i=1}^{n_c} \lambda_i - \sum_{g=1}^{j-1} \rho_g \tau_c \sigma_g}$. Therefore, $\Pr[d_j^* > d_j] = \Pr[\Theta_v > \varpi_j] < \frac{\bar{\Theta}_v}{\varpi_j}$. ■

With the facilitation of Theorem 1 and Theorem 2, Optimization 1 can be rewritten by imposing three additional constraints for the feasibility of τ_c.

Optimization 2. *The optimum QoS guaranteed broadcasting communication scheme can be mathematically formulated by*

$$\begin{aligned} & \min \tau_c k_a \\ \textit{s.t. } & \textit{(i) } \Pr[\Theta_c > \delta_i] \le \varepsilon_i, \textit{ for } i = 1, \ldots, n_c, \\ & \textit{(ii) } \Pr[\Theta_v > d_j] \le \xi_j, \textit{ for } j = 1, \ldots, n_v, \\ & \textit{(iii) } 0 \le k_a \le K, \\ & \textit{(iv) } \delta_i^* + \tau_c \le 1/\lambda_i \textit{ and } \delta_i^* < \delta_i, \textit{ for } i = 1, \ldots, n_c, \\ & \textit{(v) } \tau_c(\sum_{i=1}^{n_c} \lambda_i) + \sum_{g=1}^{j-1} \rho_g \tau_c \sigma_g < 1, \textit{ for all } i \textit{ and } j, \\ & \textit{(vi) } \tau_c > 0. \end{aligned} \tag{16}$$

To solve Optimization 2, we need to examine all feasible solutions. With the facilitation of Theorem 1 and Theorem 2, (iv) forms the strictest condition among (iv), (v), and (vi) in (16). From (iv), the number of feasible choices of τ_c does not exceed $\lceil 1/\lambda_1 \rceil$. Thus, $0 \le \tau_c \le \lceil 1/\lambda_1 \rceil$. By this observation, (16) can therefore be solved very efficiently by the following procedure.

1) Initially, set k_a=1.
2) For the given k_a, the optimal τ_c is obtained by
$$\tau_c^* = \min_{0 \le \tau_c \le \lceil 1/\lambda_1 \rceil} \tau_c \tag{17}$$
such that the constraints (i) and (ii) in (16) are satisfied.
   a) If τ_c^* can be obtained, the optimal k_a (denoted by k_a^*) is the current value. Therefore, the optimization is reached by $\tau_c^* k_a^*$.
   b) Otherwise, if $k_a < K$, set $k_a = k_a + 1$ and repeat Step 2.

The complexity of solving this optimization is $\mathcal{O}(\lceil 1/\lambda_1 \rceil)$, which is extremely applicable to a UE with limited computing capability. In above optimization, $\bar{\Theta}_c$ and $\bar{\Theta}_v$ are still unclear. In the rest of this section, the derivation of $\bar{\Theta}_c$) and $\bar{\Theta}_v$ is provided. Denote $p_{k,f} \equiv \Pr\{\Theta_c^k = f\}$. Therefore,

$$\begin{aligned} \bar{\Theta}_c &= \mathbb{E}[\min\{\Theta_c^1, \ldots, \Theta_c^{k_a}\}] \\ &= \sum_{g=1}^{\infty} \Pr\{\min\{\Theta_c^1, \ldots, \Theta_c^{k_a}\} \ge g\} \\ &= \sum_{g=1}^{\infty} \Pr\{\Theta_c^1 \ge g, \Theta_c^2 \ge g, \ldots, \Theta_c^{k_a} \ge g\} \\ &= \sum_{g=1}^{\infty} [\sum_{f=g}^{\infty} p_{1,f} \times \ldots \times \sum_{f=g}^{\infty} p_{k_a,f}] \\ &= \sum_{g=1}^{\tau_c} [\sum_{f=g}^{\tau_c} p_{1,f} \times \ldots \times \sum_{f=g}^{\tau_c} p_{k_a,f}]. \end{aligned} \tag{18}$$

To derive $p_{k,f}$, two conditions shall be considered. The true end-to-end packet forwarding time of packet transmissions via the kth path (i) exceeds τ_c with probability Φ_k, and (ii) does not exceed τ_c with probability $1 - \Phi_k$. $p_{k,f}$ is given by

$$p_{k,f} = \Phi_k \Upsilon(f) + (1 - \Phi_k)\Gamma(k, f). \tag{19}$$

For (i), since a voice packet is only allocated by τ_c subframes, if the packet transmission violates the jitter constraint, then $f = \tau_c$ and $p_{k,f} = 1$. Thus,

$$\Upsilon(f) = \begin{cases} 1, & \text{if } f = \tau_c, \\ 0, & \text{otherwise.} \end{cases} \tag{20}$$

For (ii), denote $S_{l,k}$ as the number of subframes to deliver a voice packet through the lth link of the kth path (the number of subframes of failure transmissions and redundant repetitions is not counted). It at least requires $\sum_{l=1}^{L_k} S_{l,k}$ subframes to deliver the packet via the kth path with L_k links. Therefore, $p_{k,f} = 0$ if $f < \sum_{l=1}^{L_k} S_{l,k}$ and $f > \tau_c$. If $\sum_{l=1}^{L_k} S_{l,k} \le f \le \tau_c$, $\Pr\{\Theta_k = f | \sum_{l=1}^{L_k} S_{l,k} \le f \le \tau_c\}$ is given by

$$\begin{aligned} \Omega_k &= \prod_{l=1}^{L_k} \{ \sum_{r_{l,k}=0}^{f - \sum_{l=1}^{L_k} S_{l,k} - r_{l-1,k}} \begin{pmatrix} S_{l,k} - 1 - r_{l,k} \\ r_{l,k} \end{pmatrix} \\ & \cdot \quad (1 - \varphi_{l,k})^{r_{l,k}} (\varphi_{l,k})^{S_{l,k}} \}. \end{aligned} \tag{21}$$

Therefore,

$$\Gamma(k, f) = \begin{cases} \Omega_k, & \text{if } \sum_{l=1}^{L_k} S_{l,k} \le f \le \tau_c, \\ 0, & \text{otherwise.} \end{cases} \tag{22}$$

Finally, $\Phi_k = \Phi_{1,k} + \sum_{f=1}^{L_k - 1} (\prod_{g=1}^{f} (1 - \Phi_{g,k})) \Phi_{f+1,k}$, where $\Phi_{l,k}$ is the probability that the packet transmission via the lth link of the kth path violates the maximum tolerable jitter constraint,

$$\Phi_{l,k} = \sum_{r=\tau_{l,k}^{'} - S_{l,k} + 1}^{\tau_{l,k}^{'}} \begin{pmatrix} \tau_{l,k}^{'} \\ r \end{pmatrix} (1 - \varphi_{l,k})^r (\varphi_{l,k})^{\tau_{l,k}^{'} - r} \tag{23}$$

and $\tau_{l,k}^{'}$ is the number of residue subframes before τ_c is expired. Thus, by (18)-(23), $\bar{\Theta}_c$ can be obtained. By a similar method, $\bar{\Theta}_v$ can be obtained as well.

IV. Performance Evaluations

To evaluate the performance of the proposed scheme, we need to first prove the capability of QoS guarantee provisioning. For this purpose, all the parameters and deployments for simulations are summarized in Table I and Table II. In Fig. 2, simulation results of jitter and delay violation probabilities of 5 voice and 5 video sources by adopting the proposed scheme is provided. We can observe from Fig. 2 that, by utilizing five communication path (k_a=5), timing constraints of 5 voice and 5 video sources can be satisfied when $\overline{\varphi}_{l,k} \ge 0.3$. These results show the effectiveness of our scheme to provide QoS guarantees in fully packet-switching radio access.

Next, we shall evaluate the performance of the resource

TABLE I
SYSTEM PARAMETERS AND ASSUMPTIONS FOR SIMULATIONS

Parameters	Values/assumptions
Carrier frequency	700MHz
Bandwidth	10MHz
Number of resource blocks in a subframe	10
Number of subcarriers in a resource blocks	12
Number of OFDM symbols in a resource blocks	6
Predetermined MCS	64QAM and 1/3 convolutional code
Subframe length	1ms
Total number of communication paths (K)	10
Maximum number of links in each path (L_{max})	10
Number of links in each path	Uniformly distributed in $[1, L_{max}]$
Packet length (χ)	142 Bytes
Failure probability on each link	$(1-\varphi_{l,k})$
Arrival patent of voice source	VoIP model in Table II
Arrival patent of video source	MPEG4 model in Table II

TABLE II
CHARACTERISTICS AND REQUIREMENTS OF VOICE AND VIDEO SOURCES [8]

	Voice1	Voice2	Voice3	Voice4	Voice5
λ[a]	0.05	0.04	0.03	0.03	0.03
δ	20ms	25ms	30ms	30ms	30ms
ε	0.02	0.02	0.02	0.02	0.02

	Video1	Video2	Video3	Video4	Video5
σ	69 pkt	38 pkt	48 pkt	66 pkt	61 pkt
ρ[b]	0.037	0.0012	0.091	0.037	0.0556
d	40ms	40ms	40ms	40ms	40ms
ξ	0.02	0.02	0.02	0.02	0.02

[a] λ is in the unit of subframes/packet arrival.
[b] ρ is in the unit of subframes/packet arrival.

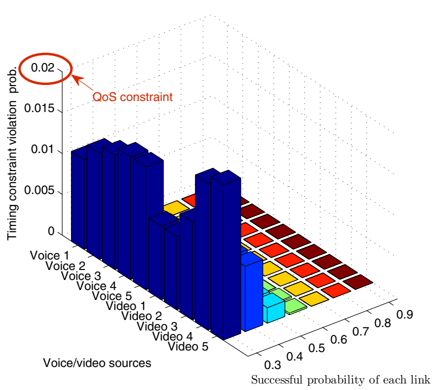

Fig. 2. Simulation results of the proposed scheme on the timing constraint violation probability, where $\varepsilon = 0.02$ and $\xi = 0.02$ for all voice and video sources, k_a=5.

utilization efficiency for transmission repetitions of the proposed scheme. We particularly adopt the repetition scheme of the H-ARQ as the comparison benchmark [9]. By adopting the H-ARQ, each receiver UE (or RN) of each link is able to inform the corresponding transmitter UE (or RN) about the amount of repetition resources for a successful packet transmission. We can observe from Fig. 3 that, to provide QoS guarantees for 5 voice and 5 video sources, our scheme achieves an outstanding resource

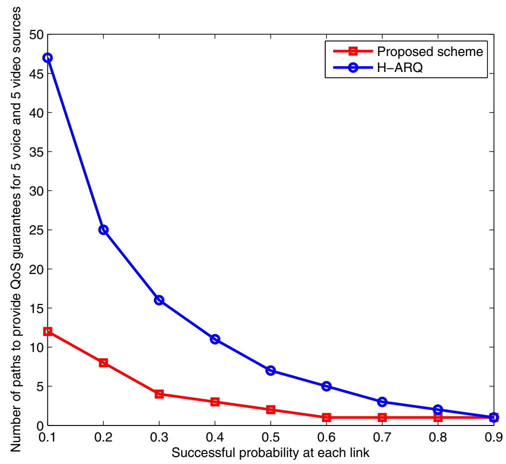

Fig. 3. Number of paths (transmission repetitions in the time and the spatial domains) needed to provide QoS guarantees for 5 voice and 5 video sources.

utilization efficiency as compared with that of H-ARQ. This result demonstrates practical applications of our proposed scheme to QoS guarantees via transmission repetitions for the fully packet-switching radio access.

V. CONCLUSION

In this paper, we optimize the performance of the fully packet-switching radio access (open-loop communications) on supporting real-time voice and video delivery via transmission repetitions. Our scheme confirms the effectiveness of the fully packet-switching radio access on providing QoS guarantees, which also achieves outstanding spectrum efficiency as compared with that of the conventional closed-loop scheme H-ARQ. Our results provide analytical foundations supporting the fully packet-switching radio access as the core technology of the next generation wireless networks.

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