On Selecting Channel Parameters for Public Safety Network Applications in LTE D2D Communications

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***Abstract*— The Third Generation Partnership Project (3GPP) defines various pre-configured channel parameters for the Long-Term Evolution (LTE) Device-to-Device (D2D) communications with Physical Sidelink (SL) Channels. In this paper, we investigate the impacts of channel parameter settings on the performance of content deliveries for Public Safety Network (PSN) applications in Out-of-Coverage (OOC) scenario. We first measure the reliability of the SL channels under various sets of channel parameters using Monte Carlo simulations. Then, for a given PSN application, the acquired reliability results are utilized to help determining the amount of delay that is to be introduced to the system, such that the throughput requirement for the application is assured during the transmissions. To the best of our knowledge, this is the first LTE D2D work that focuses on OOC mission-critical communications performance in group traffic settings. Our results are valuable to both network operators, for using them as references in selecting a best set of channel parameters, and to future studies on more complex transmission patterns and network scenarios using D2D communications in PSNs.**

I. Introduction

A. Background

During disasters, network infrastructures become susceptible to damages, and sometimes the entire network connection could be cut out [1]. To resolve this challenge we investigate the mission-critical communication (MCC) performances of the LTE Device-to-Device (D2D) communication protocol, ProSe (Proximity Services), when operating in Out-of-Coverage (OOC) environment under communication Mode 2.

Mode 2 communications in ProSe utilize the LTE Uplink resources to form Sidelink (SL) communication channels known as the ***P**hysical **S**idelink **C**ontrol **Ch**annel* (PSCCH) and the ***P**hysical **S**idelink **S**hared **Ch**annel* (PSSCH) that allow User Equipment (UEs) to establish direct communications, broadcasting in nature, without any backhauling. Due to this reason, it is considered as a prominent technology for group voice/data deliveries in OOC scenarios, both in the current LTE PSN settings and for ubiquitous applications in the future Fifth Generation New Radio (5G NR) settings.

However, since in OOC scenario there is no centralized coordination of the UEs, individual UEs allocate their own resources in the SL Resource Pool (RP) autonomously. This competitiveness among devices can cause severe packet losses. In this work, we adopt the packet loss models proposed in [2], [3], where packet loss is caused by Collision and Half-Duplex Effect (HDE). Later, we review the packet loss model and how to detect packet loss in the channels in Section III.

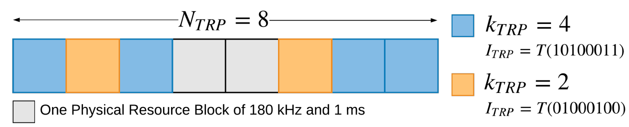

Fig. 1. **Time Resource Pattern (TRP)**: I_{TRP} is the index of the TRP mask which is a binary bit map of length N_{TRP}; when a mask bit is 1 then that bit is to be used for transmitting one data TB in the PSSCH; k_{TRP} is the number of subframes a UE can allocate its TBs within one N_{TRP} length, hence representing the number of 1's in a TRP mask. We fix $N_{TRP} = 8$.

In addition, due to its broadcasting nature, D2D communication does not have acknowledgment. To mitigate potential packet losses, two methods are specified by 3GPP [4]: i) by transmitting duplicate copies of the Transport Blocks (TBs); specifically, two copies for the Sidelink Control Information (SCI) TBs, and four copies for the associated data message TBs. As well as, ii) by limiting, k_{TRP}, the number of subframes each UE can allocate in every $N_{TRP} = 8$ subframes in the PSSCH, where TRP stands for Time Resource Pattern. The concept of TRP is illustrated in Fig. 1 with more details.

B. Problem Description

To test the performance of the proposed D2D solution in mission-critical settings, we considered an OOC group communication scenario with Mode 2 communications among N_u Half-Duplex UEs held by First Responders (FRs), all within each others' proximity. Now assume the group leader, denoted by UE-1, has an MCC message that must be disseminated to ALL the other $(N_u - 1)$ UEs in the group.

§ First, we examine the successful transmission probabilities of the SL channels by defining the **PSCCH Reliability** $\mathbb{P}\{\mathcal{S}_C\}$ and the **PSSCH Reliability** $\mathbb{P}\{\mathcal{S}_D\}$ as the probabilities of the following two events, respectively:

- $\mathcal{S}_C$: **successful reception of leader's SCI TB(s)**, "the event where all the other $(N_u - 1)$ UEs would receive at least one copy of UE-1's non-collided SCI TB in the PSCCH of a single independent SL period."
- $\mathcal{S}_D$: **successful reception of leader's data TB(s)**, "the event where all the other $(N_u - 1)$ UEs would receive at least one copy of UE-1's non-collided data TB in the PSSCH of a single independent SL period where event $\mathcal{S}_C$ took place."

§ Secondly, we consider a PSN application with a certain throughput requirement of $\mathcal{Q}$ bits/sec that is to be fulfilled, i.e., the Guaranteed Bit Rate (GBR) of that application. We are interested in the question that: while guaranteeing event $\mathcal{S}_D$ in the proximity, what is the best channel configuration that is to be used by the system such that the UE-1 reaches $\mathcal{Q}$ in the transmissions? We compare the efficiencies among each set of channel parameters through the notion of **Amount of Delay** that is introduced to the system during the transmissions.

We propose to examine the above metrics within the domain defined by the pre-configured SL channel parameters [5] presented in Table I. This is because, in consideration of FR UEs' precious yet limited battery life, it is reasonable to argue that the FR UEs should be set up before being deployed to a disaster site. Even if the settings are to be adjusted on-the-fly, it would still be more power efficient to switch among these pre-configurations. We discuss these pre-configured channel parameters and their impacts in later sections.

To the best of our knowledge, this is the first LTE D2D communications work that focuses on OOC MCC performance in group traffic settings, where the channel parameters are meaningful from 3GPP standard perspective; and could have practical impacts on promoting further research and development of the ProSe in public safety domains.

The rest of the paper is organized as follow. In Section II, we briefly review related work. Then, in Section III and IV, we present our main results on the two identified problems and the methodologies on acquiring these results. Lastly, we conclude our paper and briefly talk about future work in Section V.

II. Related Work

In [2], [3], Griffith et al. analytically modeled the channel reliability problem for PSNs and conducted extensive validations through simulating the SL channels. However, their work did not consider the throughput factor of the UEs; hence has yet to address the impacts of channel reliability onto the higher layers. Also, in their simulations, unlike ours, the channel parameters were rather artificial.

In [6], Cipriano and Panaitopol did investigate the throughput factor; however their work did not focus on the mission-critical aspect. More specifically, both their criteria for a successful transmission and throughput were aggregated over all the UEs. However, we consider the delivery of the leader's message of utmost importance. Also, best-effort throughput is not the most proper metric for limited PSNs, since the extra throughput would be "wasted" after the GBR is met.

To the best of our knowledge, most existing literature discuss performances utilizing the aggregated throughput notion. Hence, by considering the MCC Quality-of-Service perspective, our work is rather novel; and our results will be more prospective and beneficial to the public safety community.

III. Channel Reliability

A. Packet Loss in SL Channels

Now, we briefly review the two factors causing packet losses in SL channels; the general idea is being illustrated in Fig. 2. Then we discuss the steps on how to detect packet losses in the PSCCH and PSSCH.

First of all, since the UEs being considered are using Half-Duplex transmissions, i.e., if a UE is transmitting at one subframe, it won't be able receive at that subframe. Thus, in this case, as shown in the left-most of Fig. 2, the UE with its TB labeled with orange will not be able to hear the leader's transmission at that subframe, and vice versa. However, all the other UEs who are not transmitting at that subframe will be able to hear both.

Secondly, if two UEs' allocated TBs reside in the same subframe and occupy the same set of resources, then a "hard" collision happens to these two set of TBs. In this case, unlike in HDE, not only the two UEs will not be able to hear each other, but also, since these transmitted signals would add up with each other, none of the other UEs who can hear the TBs can differentiate nor successfully decode them. Thus, these two TBs and the encoded bits are to be completely discarded.

There also exists the "soft" collision scenario, where two TBs are partially overlapped. However, since all UEs are within each others' proximity, considering the worst case that the interference would still be too much for anyone to separate the two signals. Hence, these two TBs are still to be discarded. More delicate decisions could be done in the future for this part, such that when the signal-to-interference ratio is high enough, the collided TB(s) could be somewhat recovered.

Also note that, without the instructions encoded in the SCI, UEs will not be able to monitor the correct Physical Resource Block (PRB) spectrum that contains the associated data. Hence, a successful reception of the SCI is the premise of a successful reception of the data.

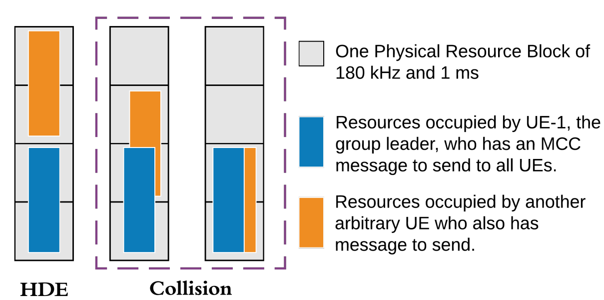

Fig. 2. **Packet Loss Model**: from left to right, each sub-figures showcases the Half-Duplex Effect, the "Soft" Collision, and the "Hard" Collision; where each TB occupies 2 PRBs.

§ Now, for event $\mathcal{S}_C$, in order to detect packet loss in the PSCCH based on our model, we employ the following steps:

1) Since a collision in the PSCCH only happens when two UEs choose the exact same random number n_{PSCCH}, to check for collisions, we simply compare UE-1's random numbers with those of the other UEs. If any UE chooses the same random number as UE-1 does, then, by our assumption, this whole SL period is going to be discarded.

2) Then to check for full HDE, we first compute the SCI allocation subframe numbers (b_1, b_2) from n_{PSCCH} [4] for each UE; and check if there is any UE who has the exact

same subframe pair as UE-1 does; i.e., if any UE will miss both SCI TBs of UE-1.

§ Now, given event $\mathcal{S}_C$ took place in an SL period, for event $\mathcal{S}_D$ in that period, to detect packet loss in the PSSCH, we employ the following steps:

3) First we perform a quick pre-check for full HDE by inspecting if any UE chooses the same I_{TRP} as UE-1 does; i.e., if any UE will miss all four data TBs of UE-1.

4) Then to check for collisions for UE-1, we do the following: for an non-leader UE, at subframes where it shares allocations with UE-1, we check the overlapping situation their TBs in the frequency-domain; if at a subframe these two TBs are colliding, then for UE-1, its TB at this subframe is going to be discarded. We perform this for all non-leader UEs, and check if all of UE-1's TBs are collided.

5) Lastly, for UE-1's remaining non-collided TBs, we iteratively check for HDE by comparing the TRP masks again.

B. PSSCH Reference Measurement Settings

From the packet loss model, it can be seen that given a fixed channel bandwidth and some number of simultaneous transmitters N_u, in the time-domain, the likelihood of HDEs taking places is affected by the lengths of the channels and how the TBs spread out in time within the corresponding channels; while in the frequency-domain, the likelihood of collisions taking places is affected by the allocation size of the TBs.

For SCI messages, 3GPP specifies all the TBs are to occupy only one PRB [4, Table A.6.4-1]; and the spread between the two TBs in time is deterministic. Hence $\mathbb{P}\{\mathcal{S}_C\}$ is only affected by N_u and the length of the PSCCH, L_{PSCCH}.

On the other hand, for data TBs, the *Fixed Reference Measurement Channel for PSSCH* [5, Subclause A.6.5], shown in Table I, defines five different TB schemes with varying Allocated RBs amounts and Modulation of Coding Scheme (MCS) Indices pairs for 10 MHz bandwidth channels. Hence, $\mathbb{P}\{\mathcal{S}_D\}$ is affected by not only N_u, the length of the PSSCH, and $\mathbb{P}\{\mathcal{S}_C\}$, but also the CD (which is an indexing prefix, not an acronym) setting used, as well as k_{TRP}, which stochastically determines how the four data TBs spread in time.

TABLE I
PSSCH REFERENCE MEASUREMENT CHANNEL

Ref. Channel	Allocated RBs	TB Size	MCS Index
$CD.1$	10	872	5
$CD.2$	10	2536	14
$CD.4$	50	12960	14
$CD.5$	2	328	10
$CD.7$	50	25456	23

C. SL Channel Resource Pool Settings

Before the evaluation on channel reliability, we configure our channel RP accordingly. We assume all the 50 PRBs are utilized for SL during public safety events. Also, in the evaluation, we assume all UEs are having good enough signal strengths for demodulating data TBs with varying CD settings; hence we categorize the references into three classes based on the allocation sizes, namely, $CD.5$, $CD.1/2$, and $CD.4/7$. Other RP-related settings are configured as follow:

1. Length of the PSCCH, L_{PSCCH}, is set to 24 subframes per period; based on the result in [2, Fig. 10], such that when $N_u \leq 15$, $\mathbb{P}\{\mathcal{S}_C\}$ is at least the target reliability of 95 %.

2. k_{TRP} is set to either 2 or 4 subframes per TRP mask length (8 subframes). This is because when $k_{TRP} = 1$, too few masks are available, hence the chance of getting HDE is going to be higher; whereas when $k_{TRP} = 8$, HDE will always happen when there are more than 1 transmitters [3].

3. SL Period Length, L_{Period}, is chosen from the values in $\{40, 80, 160, 320\}$ subframes per SL period [7], and is derived from the combined lengths of the PSCCH and PSSCH. Since this factor only affects throughput, for the channel reliability part, we simply simulate under the 40 subframes setting; i.e., there are 16 subframes in the PSSCH, which fits in exactly two sets of TRP masks. This is important since if $k_{TRP} = 2$, two sets of TRP masks are required in order complete one transmission of the four copies of the data TB. Whereas, if $k_{TRP} = 4$, two non-duplicate transmissions can be made.

Later, when throughput is involved, the period length matters in the same way that it determines how many TRP masks there are within one SL period; hence affects the number of non-duplicate transmissions that can be done per period.

D. Evaluation on Channel Reliability

Since currently there is no D2D-enabled chipset publicly available, conducting hardware tests is not an option. Hence, given RP settings, and varying UE and channel parameter settings, we evaluate the channel reliability by conducting Monte Carlo simulations, as follow: $i)$ at each SL period, UEs' SCI TBs and data TBs are generated and allocated in corresponding channels based on the SL UE procedures; and then $ii)$ using the packet loss detection schemes described above, we observe whether events $\mathcal{S}_C$ and $\mathcal{S}_D$ take place at that period. Overall, for each set of channel parameter settings, we check $\mathbb{P}\{\mathcal{S}_D\}$ out of 20 000 (sufficiently large) independent successful $\mathcal{S}_C$ events. In the experiment, all the UEs are transmitting with the same set of parameters. The reliability results over all the possible channel parameter setting combinations are shown in Fig. 3.

In Fig. 3 on the left, we observe that, for a given N_u, the $\mathbb{P}\{\mathcal{S}_C\}$ is not affected by simulation variables. We also note that the PSCCH reliability of at least 95 % with the given RP structure is consistent with the results in [2]. On the right, when $N_u > 2$, different parameter pairs result in drastically varying PSSCH reliability results. As we can observe from the results, by choosing different CD and k_{TRP} pairs, there is a trade-off between having a higher throughput and having a higher reliability. We will address this trade-off in the next section.

Here, we only present the reliability results obtained from one of many runs of our Monte Carlo simulations. The differences among the results of the runs, for the same settings, are no more than 0.5 %. Hence, the results are representative.

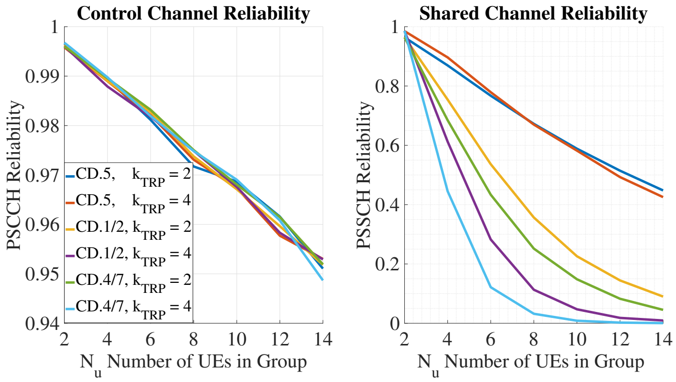

Fig. 3. Simulated results of **PSCCH Reliability** $\mathbb{P}\{\mathcal{S}_C\}$ (left) and **PSSCH Reliability** $\mathbb{P}\{\mathcal{S}_D\}$ (right) for different chosen CD Reference Measurements and k_{TRP} combinations under various N_u values.

IV. Throughput Planning and Delay

In this section, we incorporate the channel reliability results acquired above with the notion of PSN application GBR $\mathcal{Q}$ and try to derive the amount of delay metric. In order to do so, we propose a duplicative transmission scheme that assures the success of group deliveries of each second-worth data; and based on the amount of duplications needed for each SL period under different channel parameter settings, we calculate the amount of delay.

In our experiments, the PSN application GBR values are originally conducted by the Minnesota Department of Public Safety [8] and later modified by the Communications Technology Laboratory (CTL) at the National Institute of Standards and Technology (NIST) with additional network-layer overhead added. Since the reference channel parameters act between the network-layer and the transport-layer, the GBRs we utilize are compatible with the cross-layer settings.

A. Data Broadcasting and Throughput

We observe that, in order for UE-1 to successfully broadcast its application content in the SL channel and attain a throughput at least the amount $\mathcal{Q}$, it needs to transmit in multiple periods within a one second window. The number of transmissions, given an SL RP, then depends on $i)$ the period length L_{Period} and $ii)$ the k_{TRP} setting of the UEs, as explain in the following.

$i)$ For example, if the SL period length is 40 ms, then there are going to be $\lfloor 1000/40 \rfloor = 25$ full SL periods that can be utilized for transmissions; comparing with if the SL period length is 80 ms, then there are fewer, 12, full periods that can be utilized. However, when using a longer period, the data rate (not the actual throughput) is going to increase. When the period length is 80 ms, besides the 24 subframes of control region, the remaining 56 subframes can accommodate 6 TRP masks per period; comparing with only 2 per period in the 40 ms period length case. Later, we denote the number of full periods per second as ρ, and number of TRP masks within one period as γ.

$ii)$ Meanwhile, given the same RP, if $k_{TRP} = 2$, then within each TRP mask, only 1 transmission can be done, since in this case only half of the 4 copies are going to be allocated within each of 8-subframe set; comparing with when $k_{TRP} = 4$, within each period, 2 transmissions can be done, where the second transmission has the exact same allocation as the first one; i.e., the PSSCH reliability stays the same. However, this does not necessarily mean the throughput is increased when using the tighter mask, since as we can see from the reliability results, in most cases, the reliability is significantly lower when $k_{TRP} = 4$ comparing to that when $k_{TRP} = 2$ under the same N_u and CD settings. Thus when $k_{TRP} = 4$, it is more likely to have packet loss taking places, which will jeopardize the whole transmission. Thus, by having a longer period, the penalty for packet loss is going to be higher than that of having a shorter period.

Hence, both factors play important roles in attaining a desired throughput for UE-1, along with the actual allocation choices, i.e., the CD options for PSSCH reference measures.

B. Performance Evaluations

Now we begin to discuss the throughput planning part of our study, where given a fixed number of N_u FR UEs, a PSN application with GBR $\mathcal{Q}$, we want to check if it would be possible for UE-1 to achieve latency-free transmissions, and if not, what is the amount of delay that should be expected?

We introduce the notion of *overall transmission successful probability*, $\mathcal{P}$, as the transmission successful probability over i contiguous periods within each independent second; and if the single period channel reliability is p, then we have $\mathcal{P} = p^i$. We define the target probability as $\hat{p}$; hence as long as $\mathcal{P}$ reaches the target threshold, the second-wise transmission is considered to be successful.

Next, we investigate how the factors mentioned above, i.e., L_{Period}, k_{TRP}, and the selected CD Reference Measurement affect the transmission performance, measured through the delay amount δ. In reality, this evaluation is conducted before the deployment taking place, such that the preferable parameter setups can be implemented for the channel and UEs beforehand; and also after the deployment a UE can switch to a different CD setup that is more appropriate for its own demand based on the obtained results.

We demonstrate the process on how to obtain the amount of delay for an application under various channel and UE settings, by giving a walk-through for the "Next Generation 911 (NG911) Video Medium Resolution" streaming application, with $\mathcal{Q} = 274,562$ bits/sec, when $L_{Period} = 40$ ms, as follow. The formal algorithm description for obtaining the delay amount δ is given by **Algorithm 1**.

1. We first assume all ρ periods within a second are to be used for transmission, to get the minimum throughput that must be met for each transmission and use it to decide what are the choices among the CD settings that are valid for our current parameter settings, by averaging $\mathcal{Q}$ over the total number of transmissions within one second. In the example,

Algorithm 1 Amount of Transmission Delay per Second

Input: GBR of Application Q, Period Length L_{Period}, UEs' k_{TRP}, Threshold for Overall Transmission Successful Probability $\hat{p}$;
Output: Amount of Delay δ

$\rho \leftarrow \lfloor 1000/L_{Period} \rfloor$;
if ($k_{TRP} == 2$) **then**
  $\gamma = \lfloor (L_{Period} - L_{PSCCH})/(2 \times N_{TRP}) \rfloor$;
else
  $\gamma = \lfloor (L_{Period} - L_{PSCCH})/N_{TRP} \rfloor$;
end if
$\omega \leftarrow \lceil Q/(\rho \times \gamma \times k_{TRP}/2) \rceil$;
for (each CD. Measurement that has TB size $\sigma \geq \omega$) **do**
  $\eta \leftarrow \lceil Q/(\sigma \times \gamma \times k_{TRP}/2) \rceil$;
  if ($(p)^{\eta} < \hat{p}$) **then**
    Find smallest $\tau \in \mathbb{N}$, such that $(1-(1-p)^{\tau})^{\eta} \geq \hat{p}$;
    $\delta \leftarrow (\tau \times \eta - \rho) \times L_{Period}$;
  else
    $\delta \leftarrow 0$;
  end if
  if ($\delta < 0$) **then**
    $\delta \leftarrow 0$;
  end if
  Record current δ value;
end for

there are $\rho = 25$ full periods. If $k_{TRP} = 2$, the number of 16-subframe pairs is $\gamma = \lfloor (40-24)/16 \rfloor = 1$; and if $k_{TRP} = 4$, the number of 8-subframe sets is 2. Then, the total number of transmissions would equal to $(25 \times 1 \times k_{TRP}/2)$.

If $k_{TRP} = 2$, then each period can only allocates one transmission; hence, if all the 25 periods are being used for the transmission, each period must attain a TB of size $\omega = \lceil \mathcal{Q}/25 \rceil = 10983$ bits/allocation, which maps to the $CD.4$ Reference Measurement. Then, if we divide the GBR by the product of the TB size of $CD.4$ and the number of transmissions per period, we would find that if $CD.4$ is used as the allocation scheme, then $\eta = 22$ full periods are actually going to be needed. Here, we assume when all the periods are transmitting, the choice is $CD.4$. If we plan to use $CD.7$ which has a larger TB size, even fewer periods, $\eta = 11$ are going to be needed. Let us denote when $k_{TRP} = 2$ and using $CD.4$ as **Case a)**, and when $k_{TRP} = 2$ and using $CD.7$ as **Case b)**.

If $k_{TRP} = 4$, which doubles the throughput in each period, then the minimum required TB size would be about halved, $\omega = 5492$ bits/allocation, and this again maps to the $CD.4$ Reference Measurement. Similarly, we can choose a higher TB size. If we choose to transmit with $CD.4$, then $\eta = 11$ periods are needed; if $CD.7$ is chosen, then $\eta = 6$. We denote when $k_{TRP} = 4$ and using $CD.4$ as **Case c)**, and when $k_{TRP} = 4$ and using $CD.7$ as **Case d)**.

2. Now, given number of UEs in the group, say, $N_u = 2$, we want to compute $\mathcal{P}$ as p^{η}, i.e., using only η periods.

Hence, for **Case a)**, where $p = 96.41$ %, $p^{11} = 44.74$ %, which is going to be significantly lower than $\hat{p}$, the threshold for having a successful transmission over one second. Similarly, for **Case b)** and **Case c)**, after the exponentiation, we have $0.9641^{11} = 66.89$ % and $0.9861^{11} = 85.75$ %, respectively, and both fall short to meet the goal $\hat{p}$.

However, for **Case d)**, since it requires much less transmission periods, the overall probability, $0.9861^{6} = 91.94$ %, meets the goal, and, as a result would cause no delay at all.

3. For the former three cases, in order to mitigate this exponentially add-up attenuation effect, for each of the periods, we propose that more than one copies are to be transmitted. When the modulated information in a period is transmitted for τ times, call it **trials**, the probability that at least one copy is successfully transmitted is calculated as $(1-(1-p)^{\tau})$.

For **Case a)**, if $\tau = 2$, then the successful probability over a second is $(1-(1-p)^{\tau})^{11} = 97.20$ %, and thus the success of transmissions is going to be assured. This is very similar to the idea of transmitting multiple copies within a period, but now the duplication is regarding the periods instead of the allocations within the periods. For the other cases, $\tau = 2$ as well.

4. However, as we just discussed, by transmitting extra copies of the periods, the delivery for the content within each second is going to be delayed.

In both **Case b)** and **Case c)**, since all the 11 periods are going to be used for transmissions, and each periods are going through 2 **trials** to assure the success of transmission, in total $11 \times 2 = 22$ periods are going to be transmitted, which still falls into the one second (25 periods) range; and hence also would not cause any delay. On the other hand, for **Case a)**, 44 periods are needed, where 25 of them fall within the one second range, and the other 19 are going to be counted as the delay portion; hence, for the content of the first second, the delay is going to be $\delta = 19 \times 40 = 760$ ms.

Hence, delay-wise, we can see **Case a)** would be the least favorable scheme among the four to choose for this application, under currently tested settings. For **Case d)**, it can also transmit multiple copies to further improve its $\mathcal{P}$.

In general, by adjusting the three parameters, which results in different η and τ values, will change the amount of delay. Hence, for various PSN applications with different GBR values, for various N_u values, and among all the valid combinations of the three parameters, we conduct extensive experiments to collect the resulting delay amount δ using our proposed **Algorithm 1**. Our experiments generate sets of tables on the δ values with completeness; hence network operator can then set up the SL channel and the D2D UEs according to these pre-computed δ values so that the delay budget is met. The results can also be used to draft new latency budget policies, since none of the existing ones were made for OOC scenario in particular.

Due to space limitation, we only present part of our experimental results, as shown in Fig. 4 for the "NG911 Video MQ" application which has a relatively high GBR value; and in Fig. 5 for the "Phone Voice" application who has the intrinsic highest priority in communication, with a moderate GBR of $30,440$ bits/sec. and for both applications we focus on $N_u = 4$, where zero delay is achievable. When N_u becomes larger, it is most likely that at least some degrees of delay is to be introduced, and network operators can evaluate which schemes are better under circumstances by considering the trade-offs.

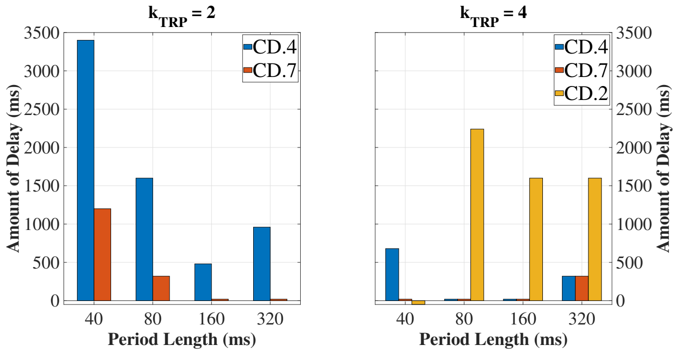

Fig. 4. Amount of delay for "NG911 Video MQ" application with $\mathcal{Q} = 274,562$ bits/second, for $N_u = 4$; within each group of bars, from left to right, each bar represents $CD.4$, $CD.7$ and $CD.2$ (if exists), respectively.

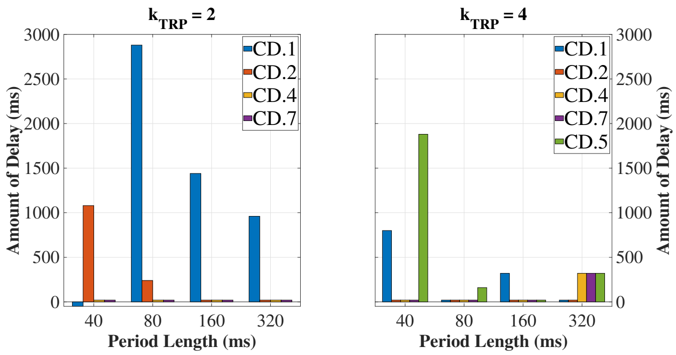

Fig. 5. Amount of delay for "Phone Voice" application with $\mathcal{Q} = 30,440$ bits/second, for $N_u = 4$; within each group of bars, from left to right, each bar represents $CD.1, 2, 4, 7$, and $CD.5$ (if exists), respectively.

In both Fig. 4 and Fig. 5, we use bars with very small heights to represent that there are no delays for the communications under certain parameter settings. Also, the yellow and blue bars with negative values indicate invalid data, since those settings will not be able to achieve $\mathcal{Q}$ at $L_{Period} = 40$ ms.

When comparing the two figures, we can see that when $\mathcal{Q}$ becomes smaller, more reference measurements with smaller TB sizes become valid for transmission. However, it is obvious that, in most cases, although channel parameters with smaller TB sizes have much higher channel reliability, they do **NOT** produce smaller delay values, due to the exponentially add-up attenuation on the reliability. Two exceptions being that, in Fig. 5 when $L_{Period} = 160$ and 320, settings with smaller TB sizes would actually produce smaller amount of delays. On the other hand, channel parameters with larger TB sizes usually require receiving UEs to have better channel qualities in order to actually demodulate the data TBs, which obviously is not always achievable in real-life situations. Thus, it is up to network operators to balance the trade-off accordingly.

V. Conclusion and Future Work

In this work, we thoroughly investigate the channel parameters that will affect the channel reliability, and consequently, the amount of delay introduced during communication, with varying FR group sizes and GBRs of PSN applications. We introduce an approach to assure the successful delivery of content under a rigorous packet loss model, and derive the amount of delay introduced by applying this approach under all the possible parameter combinations.

It can be seen from the amount of delay results that, although in general, settings that carry out larger TB sizes per allocation, despite of having much lower channel reliability, would produce smaller amount of delays; exceptions do exist, and hence the problem is indeed worth studying. Thus, our results have the potential to benefit network operators and researchers in the future operations and studies on SL channels.

As we all know, standards are evolving as technologies develop. Therefore, some of the parameters we investigate in this work may be extended and end up having more values to be utilized to increase the channel reliability, UE throughput, and thus reduce the amount of delay being introduced. We plan to generalize the domains of certain parameters, which will likely make it impossible to construct a table that contains all the parameter information. We will investigate the development of an algorithm that would yield an optimal setting without having to generate the complete table.

Furthermore, there are plentiful of topics that can be expanded into and studied based on our work. For example: **Role-based D2D UEs** where UEs may choose different parameters based on their demand and behavior/roles; and, **Multi-hop D2D Communication** where messages being passed through link paths between base station and remote UEs via multiple D2D-enabled relay UEs.

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