Data Interleaving for Congestion Reduction in Mobile Traffic Transmission

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***Abstract*—In mobile traffic transmission networks, the limited allotted spectrum—involving voice communication and data communication—is faced with the problem of limited bandwidth and consequent traffic congestion. The problem leads to poor communication services like incomplete calls, call drops, slower Internet speed, higher cost of internet service, and other undesirable effects by the service providers. The bandwidth saturation and congestion conundrum still remains critical research area, especially with emerging multitude of edge and intermediate Internet devices. Due to introduction of fog computing such devices become active participants of mobile traffic transmission. In this paper, we consider the Data Interleaving Technique in Mobile Communication (DITMC) and introduce the DITMC-based methodology for reducing the data traffic congestion and bandwidth saturation in GSM networks. The methodology is supported with an Erlang formula model, and our simulation experiments indicate the efficiency. We show that the use of methodology can be extended to other Internet problems like multi-path routing of data streams, active subscription control in smart spaces, and possibility to delegate supplementary processing to edge devices.**

I. INTRODUCTION

Two decades ago the mobile voice traffic significantly dominated the transmission of data traffic in the Global System for Mobile Communications (GSM) telephony. In the present scenario as well as by the end of 2020 (when 5G technology and LTE will be dominating), the mobile data traffic will supersede the voice traffic due to plethora of novel applications being used in the mobile Internet. Consequently, mobile traffic transmission networks are faced now with the problem of high congestion and bandwidth saturation.

In particular, a GSM network involves both FDMA (Frequency Division Multiple Access) and TDMA (Time Division Multiple Access). The effective (time) utilization of the traffic channels plays an important role in optimizing the use of limited bandwidth as well as the data traffic transmission [1]. Indeed, traffic analysis becomes a vital part of mobile voice/data communication. There is a need of performance models for traffic congestion reduction. Such a mathematical model aims at avoiding the expected congestion problem as well as enhancing the capacity of mobile traffic transmission networks [2]. Such a traffic analysis model must be applicable for both real time (voice) and non-real time data. The traffic model design must have the “Goodness-of-Fit” property, i.e., the results obtained from the model must be reasonably close to the actual data (e.g., satisfactory for prediction).

In this paper, we continue our study of the congestion problem in mobile traffic transmission networks, which we started in [3]. We analyzed the Data Interleaving Technique in Mobile Communication (DITMC). This technique forms a strong basis for reducing the traffic congestion in various mobile networks. We consider a reference mathematical model [1] that utilizes such parameters as traffic channels, capacity, cost, Grade of Service (GOS), coverage, available frequencies, and voice quality. The model is based on the well-known Erlang formula and its application to traffic analysis [2].

Based on the mathematical model and our initial simulation experiments, we analyze the role of DITMC in optimizing the congestion in data traffic as well as bandwidth saturation. This paper contributes the generic DITMC-based methodology for data interleaving. The proposed methodology supports better enhanced coverage and reduced traffic congestion. The methodology can also be applied in Edge-centric environment of IoT, i.e., supporting so-called fog computing [4] when the problem is selection of some surrounding devices to delegate them some processing tasks. Also, multi-path routing of stream data [5] can benefit from this methodology since a path in a network can be considered as a channel, similarly to GSM networks. Another application area is mobile agent coordination in smart spaces [6].

The rest of the paper is organized as follows. Section II introduces the congestion problem in mobile traffic transmission networks. Section III considers the reference mathematical model for voice and data traffic. Section IV proposes the DITMC-based methodology for reducing congestion. Section V describes the initial experimental indication of the methodology applicability and the efficiency. Finally, Section VI concludes the paper.

II. TRAFFIC CONGESTION IN GSM NETWORKS

In 2016, global mobile data traffic amounted to 7 ExaBytes (EB) per month. In 2021, mobile data traffic worldwide is expected to reach 49 EB per month at a compound annual growth rate (CAGR) of 47%. According to the Ericsson report on mobile traffic growth, summarized in Table I. In particular, by the end of 2022, there will be 12 times more mobile data traffic in Central and Eastern Europe and Middle East and Africa (CEMA).

In Russia, according to CISCO VNI (Visual Networking Index) Mobile Forecast Highlights, it is predicted the average

TABLE I. EXPECTED MOBILE DATA TRAFFIC (BY ERICSSON)

Mobile Data Traffic by Region	2016 (EB per month)	Multiplier 2016–2022
Asia Pacific	3.6	8
Central & Eastern Europe, Middle East & Africa	1.8	12
Western Europe	1.2	8
North America	1.2	6
Latin America	0.7	8

smartphone will generate 6,350 MB of mobile data traffic per month in 2021, compared with 1,792 MB per month in 2016. The average mobile-connected laptop will generate 6,285 MB of mobile data traffic per month in 2021, compared with 2,821 MB per month in 2016. There will be 97 million total Internet users (68% of population) in 2021, compared with 86 million (60% of population) in 2016. The average mobile-connected tablet or smartphone will generate 7,055 MB of mobile data traffic per month in 2021, compared with 2,584 MB per month in 2016.

In India, consumer mobile traffic will grow 7.4-fold from 2016 to 2021, a compound annual growth rate (CAGR) of 49%. Consumer mobile traffic will reach 1.8 EB per month by 2021, compared with 238.0 Petabytes (PB) per month in 2016. Everyday mobile activities such as mobile social media usage, including mobile chat and voice or video calls, and mobile e-commerce drive mobile traffic but mobile video has the largest growth rate out of all of the mobile content categories. Between 2016 and 2022, smartphone traffic is expected to increase by 10 times and total mobile traffic for all devices by 8 times. By the end of this period, mobile data traffic will preemptively more than voice traffic.

Respectively, the transmission of such amount of traffic inevitably results in a multitude of data and voice flows to manage in GSM networks. An efficient traffic model is required for reducing the congestion in mobile data traffic. The model leads to better network performance and QoS (Quality of Service). Various models and consequent algorithms were proposed based on resource allocation strategies like CP (complete partitioning), PP (Partial partitioning), channel allocation schemes (e.g., SCA, DCA, Centralized DCA, and Distributed DCA) for optimizing and managing traffic congestion have already been developed, see examples in [1], [7], [8].

Nevertheless, the bandwidth saturation and congestion conundrum remains critical research area. It has already been analyzed [3] that most economic and efficient method with minimum hardware complexity to optimize the limited channel capacity and bandwidth in GSM networks is to use the redundancy in voice samples. As well as any silence period can be used during any voice conversation [9]. Such periods can be effectively utilized to transmit data. Our previous simulation study [10] confirmed that this improvement can lead to overall channel utilization up-to 83%.

Now data interleaving is expected to be a promising approach for smart network communication, when data transmission is integrated with data processing, as it happens now for the needs of growing Industrial Internet in software defined networks [11] and smart spaces development [12]. In this case, emerging vision of fog computing [4] supports the use of data interleaving by edge devices (e.g., mobile phones) [6] or intermediate network devices (e.g., Wi-Fi routers) [13] to improve the performance of traffic transmission of continuous data flows. A particular case is multimedia dissemination [14]. Multi-path routing [5] of continuous data flows can benefit from data interleaving since a path in a network can be considered as a channel, similarly to GSM networks.

III. REFERENCE MODEL FOR DATA AND VOICE TRAFFIC

Consider basic details of transmission in GSM networks, e.g., see [8]. The GSM has physical channels and traffic channels. In traffic channels various channel allocation schemes like FCAS (Fixed Channel Allocation Scheme) and DCAS (Dynamic Channel Allocation Scheme) are frequently used. The FCAS has advantages of less implementation complexity, larger coverage, optimized level of signaling load. On the other side, it has the drawback of exhibiting short delay period for call connection, interference and high service cost. Similarly, DCAS has numerous advantages as the channel allocation is dynamic, i.e., as and when its required. This approach provides maximum flexibility and adaptability at the cost if higher system complexity. In this sequence, the DITMC technique seems to be unique one as it does not follow any borrowing scheme from the adjacent cells. It rather finds out the free slots in the channel itself to incorporate more channels for data traffic by finding out the repetition and redundancy in the voice samples.

For our reference case, we consider the simple mathematical model discussed in [1] for the traffic congestion problem in GSM networks. Generic properties of this model for the Internet case are discussed in [2]. For the voice traffic, the calls are assumed to follow the Poisson point process with rate λ_{v}, i.e., the probability of n concurrent calls in the network is defined as

$$\mathbb{P}(N = n) = \frac{\lambda_{\mathrm{v}}^n}{n!} \exp(-\lambda_{\mathrm{v}}) \ .$$

The duration of each call is distributed exponentially with the parameter μ_{v}, i.e., the duration τ_{v} is less than x with the probability

$$\mathbb{P}(0 \leq \tau_{\mathrm{v}} < x) = \mu_{\mathrm{v}} \exp(-\mu_{\mathrm{v}} x) \ .$$

The model is reduced to the Markov chain depicted in Fig. 1. In the TDMA scheme, the total number TS of time-slots is partitioned into the contiguous sets:

- $\mathrm{TS_v}$ time-slots are dedicated to voice calls,
- $\mathrm{TS_{vd}}$ time-slots are shared between voice and data traffic,
- $\mathrm{TS_d}$ time-slots are dedicated to data transmission by GPRS,

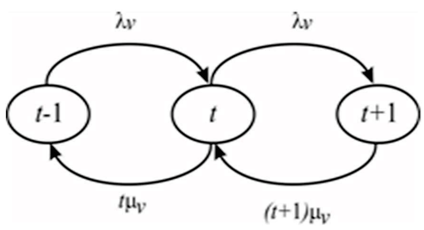

Fig. 1. Markov chain for the voice traffic model with discrete time-slots t.

Fig. 2. Data Interleaving (First Order).

Then $\mathrm{TS_d} + \mathrm{TS_{vd}}$ time-slots are on a single TDMA, which has total of 8 time-slots.

Denote $\rho_\mathrm{v} = \lambda_\mathrm{v}/\mu_\mathrm{v}$. The model has the unique steady state

$$p_\mathrm{v}(t) = \frac{\rho_\mathrm{v}^t/t!}{\sum_{i=0}^{\mathrm{TS_v}} \rho_\mathrm{v}^i/i!}\ , \quad t = 0, 1, \ldots, \mathrm{TS_v}\ . \tag{1}$$

The performance metric can be represented by the blocking probability (also known as Erlang-B formula)

$$B_{\mathrm{v,cp}} = \frac{\rho_\mathrm{v}^{\mathrm{TS_v}}/\mathrm{TS_v}!}{\sum_{i=0}^{\mathrm{TS_v}} \rho_\mathrm{v}^i/i!}\ . \tag{2}$$

This model can be used for data interleaving with DITMC [3]. In the next section, we formulate the DITMC-based methodology that suggests the increase in available data (channels) slots by identifying the repetition and silence in the voice calls. Equations (1)–(2) provide a way for managing the traffic rate and duration and enabling significant increase in time-slots available for data traffic. As a result of this kind of management, the data traffic congestion can be reduced to a significant order. Moreover, the management is used by all mobile units in the cell, and the data traffic optimization covers the overall cell.

IV. DITMC Based Methodology

The 5G networks have the capability of accommodating the enhanced number of users as compared to 4G with faster data access rates. Nevertheless, by the end of 2020 at least, the unsatisfying demand of mobile Internet still exists when the number of users (capacity) is increasing with fast Internet access speed along with the minimum delay, good voice quality, and negligible interference. It will definitely be followed by LTE while the problem of traffic congestion remains insufficiently elaborated. In this case, the DITMC provides a remedial, economical and simple (with less hardware complexity) solution. In the basic case of mobile communication, DITMC is applicable at MS (Mobile Station) side, BTS (Base Transceiver Station), and other appropriate intermediating nodes individually to play a significant role in capacity enhancement and mitigating the traffic congestion without any interference, delay, or voice degradation quality.

DITMC involves the dynamic utilization of the channels by considering the repetition in voice bytes and the bytes generated due to silence period in any call. Whenever the repetition of bytes is 4 (four) or more, a Battery OFF Code (Data Start Code) is transmitted (after the first voice byte). This was, the receiver side is indicated about the repetition. When the repetition is over then a Battery ON Code (Data Stop Code) is sent to indicate the end of interleaving process followed by the normal transmission. The scheme of data interleaving is shown in Fig. 2.

This technique of data interleaving possesses the essential attributes [3]. They include voice delay of 2 samples is only 250 μs in time, better channel utilization enhancement, no handover, suppression of the transmission of redundant (repetitive) message codes, making channel resources available for additional data transmission, detection of much smaller duration voice pauses (which are generally more frequent), negligible inbuilt delay (overhead) in this system is insignificant for real-time applications.

The two bytes delay in transmission is neither noticeable by the listener nor effects the operation of the system in any adverse way. Moreover, the voice quality maintained by interleaved system is far superior to that of the systems using voice interpolation. The interleaved data can also have repetition that may also be considered as redundant. The dynamic utilization of this redundant space also leads to better channel utilization enhancement. Our initial experimental study shows that the enhancement in channel utilization is completely random depending upon the type of data files (see Table II in Section V further).

DITMC exhibits the following advantages as compared to its predecessor DITV (Data Interleaving Technique in Voice Networks).

- Better channel utilization enhancement (47.32%) than DITV (43.253%).
- DITMC does not exhibit any handover.
- Lesser voice delay of 2 Bytes as compared to DITV.
- DITMC suppresses the transmission of redundant (repetitive) message codes.
- DITMC makes channel space available for additional data transmission.
- DITMC can detect much smaller duration voice pauses, which are generally more frequent.
- There is negligible inbuilt delay in this system and is insignificant for real-time applications.
- Unlike voice interpolation processes, DITMC does not make use of any voice detectors.

The signal delay takes place at the originating node only and is independent of network distance. In a GSM network the TDMA is further divided into a specific number of slots (channels) for both voice and data. The DITMC-based methodology is elucidated in Fig. 3 for various voice samples (say S1 to S8) to be taken for modeling and simulation.

Channel Allotted to Voice & Data	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	Number of data channels (due to interleaving)
Channel available for S1	V	V	V	D	D	D	V	V	V	D	D	D	D	D	V	D	9
Channel available for S2	V	V	I	D	D	D	I	V	V	D	D	D	D	D	V	D	11
Channel available for S3	V	I	I	D	D	D	V	I	I	D	D	D	D	D	V	D	13
Channel available for S4	V	I	V	D	D	D	I	I	V	D	D	D	D	D	V	D	12
Channel available for S5	V	V	V	D	D	D	V	V	V	D	D	D	D	D	V	D	09
Channel available for S6	I	I	I	D	D	D	I	I	I	D	D	D	D	D	I	D	15
Channel available for S7	V	I	I	D	D	D	I	V	I	D	D	D	D	D	I	D	14
Channel available for S8	V	V	V	D	D	D	V	V	I	D	D	D	D	D	V	D	10

Fig. 3. Increased channels for data due to the data interleaving concept. V: voice channel, I: interleaved channel available for data traffic, D: data channel

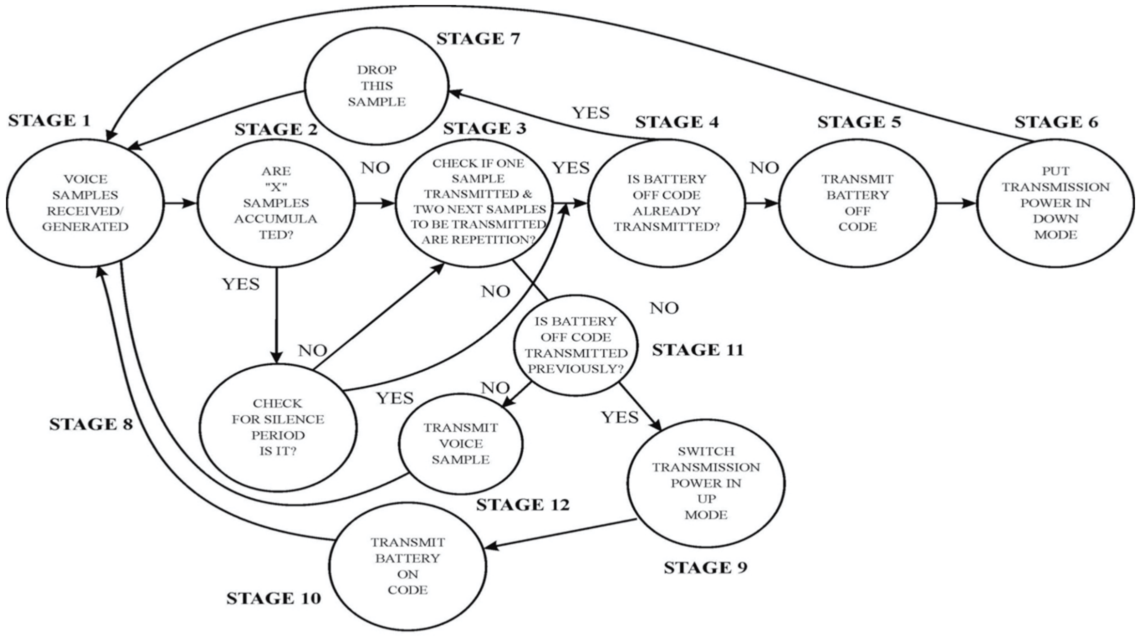

Fig. 4. State Transition Diagram for DITMC at mobile station

Since the data interleaving concept is applicable for only one channel the DITMC-based methodology is intended to be used for allocating as well as increasing in the number of data channels (that are fixed for a particular traffic channel). For example, Fig. 3 shows that the minimum number of data channels allotted is 9 (randomly). Since interleaving is possible in voice communication only (for the channel in use) then there will be increase in the number of data channels available. Hence more data can be accommodated leading to the significant reduction in data congestion.

The data interleaving concept is not dependent upon the congestion, it can rather reduce the congestion. The concept can also be utilized by all the users allotted for voice communication thereby reducing congestion in the whole data traffic.

The state transition diagram for generating and transmitting the signal at the mobile station (MS) is shown in Fig. 4. The model is derived from [10]. At this MS side the redundancy in voice samples or silence period is detected. The decision is made about sending the BATTERY OFF, BATTERY ON code and sending the voice sample itself. There are 12 stages (Stage 1 to Stage 12) to perform this operation.

First of all, the voice samples are received or generated. From these samples the decision is made regarding the accumulation of say n samples. When the n samples are accumulated they are checked for silence bytes, i.e., whether these are indicating silence period or voice. If these are silence bytes then they are checked for the continuous repetition of more than three bytes. The BATTERY OFF code is sent and the transmission power is put in DOWN mode provided that BATTERY OFF code is previously not transmitted. In this case, if BATTERY OFF code is found to be sent previously then this sample is dropped and the next voice samples are received or generated.

If the accumulated n samples are not silence bytes then the repetition is checked in these declared voice samples/bytes. After this voice samples detection they are checked for the continuous repetition of more than three bytes. If there is continuous repetition of more than three bytes and BATTERY

OFF code is already transmitted then the sample is dropped and the next voice samples are received or generated. If BATTERY OFF code is not transmitted previously then the BATTERY OFF code is sent and the transmission power is put in DOWN mode. However, if there is no continuous repetition of three or more bytes in the voice sample the decision is to be made regarding BATTERY OFF code. That is, if this code is transmitted previously then the transmission power is switched in UP mode, BATTERY ON code is transmitted, and the next samples are received or generated. Otherwise, the voice samples are transmitted and the next voice samples are received or generated.

The methodology behind allocating the channels for data is to make use of voice channels. They can be optimized due to redundancy and silence period of any call. Moreover, the data interleaving concept can also be applied to increase the capacity of GSM networks. This way, the data traffic congestion can further be reduced.

The simplified mathematical equation for enhancing the capacity in a GSM network using DITMC can be described as follows. The enhancement in time utilization in a voice conversation in mobile communication (one side only) is expressed as the following mathematical model. Let

$$\nu = T_{\text{on}}/T_{\text{tot}} \ ,$$

where ν is estimation of the time utilization without DITMC, T_{on} is the ON time of the conversation. Then

$$T_{\text{tot}} = T_{\text{on}} + T_{\text{off}} \ ,$$

where T_{off} is the OFF time observed on channel including silence period.

The time utilization with DITMC is evaluated as

$$\nu_{\text{DI}} = T_{\text{eff}}/T_{\text{tot}} \ ,$$

where T_{eff} is the reduced T_{on} with DITMC.

Consequently, the enhancement in time utilization can be defined as the difference

$$\delta = \nu - \nu_{\text{DI}}.$$

This means that if ν_{DI} is smaller than ν then more enhancement in time utilization can be obtained. It will further leads to increased channel utilization enhancement. In GSM networks the enhancement in channel utilization, capacity (number of calls/channels in a cell) can be achieved by the implementation of DITMC.

The blocking probability P_{b} with the implementation of data interleaving technique can be mathematically expressed as follows.

$$P_{\text{b}} = \frac{(A_{\text{DI}+A_{\text{h}}}) \sum_{k=c}^{c+g} A_{\text{h}}^{k-c}/k!}{\sum_{k=0}^{c-1} \frac{(A_{\text{DI}+A_{\text{h}}})^k}{k!} + (A_{\text{DI}+A_{\text{h}}})^c \sum_{k=c}^{c+g} A_{\text{h}}^{k-c}/k!} \ , \quad \text{where}$$

A_{DI} is the increased traffic due to arrived calls with data interleaving technique, A_{h} is traffic due to handover calls, g is guard channels, c is the number of access channels, i.e., $c = N - g$, where N is total number of channels per cell.

The proposed DITMC-based methodology uses mathematical modeling on the basis of queuing theory and blocking probability, as we showed in Section III. Our further direction is simulation modeling to demonstrate the "goodness-of-fit" property and the reduction level of traffic congestion.

V. Initial Experimental Indication

The experimental results obtained from our simulation experiments for GSM network settings are shown in Table II and Fig. 5. More details can be found in [10], [3]. The results provide the strong basis for expected significant reduction in data traffic congestion leading to more subscribers accommodation in a cell. Since the channel utilization enhancement is already 47.32% leading to overall channel utilization (voice) of 83.32% (36 + 47.32) with negligible overhead in terms of bytes, the data traffic congestion can be expected to be reduced at least in the order of 40% (yet to be examined).

The total number of bytes in the random 120 voice samples is 2.536 GB. The total number of interleaved bytes is 1.2 GB. Data interleaved in Voice is 47.32%. Te overhead involved in the interleaving process is 0.034394%. The channel utilization without interleaving is 36%. Therefore, channel utilization with interleaving is $36 + 47.32 = 83.32\%$.

The application scope of DITMC-based methodology is not limited by GSM networks or their LTE/5G successors. The following problem domains of distributed systems indicate the applicability of data interleaving.

Problem 1: Multi-path routing [5] when streaming data flows, e.g., multimedia in edge-centric and mobile environments. In this case, each path can be treated similar to a GSM channel, and the data sender selects which path to use to sent the current packet of the flow. In particular, we experimentally showed that random selection by the data sender of periods to

TABLE II. Experimental Results of DITMC

#samples	File type	Interleaving %	Overhead %	Enhancement %
120	.wav	47.32	0.034	83.32
25	.doc/.docx	11.90	1.25	47.9
25	.pdf	1.41	0.28	37.41
25	.ppt	6.22	1.34	42.22
25	.xls	15.43	2.68	51.43
25	.jpg	0.041	0.015	36.041

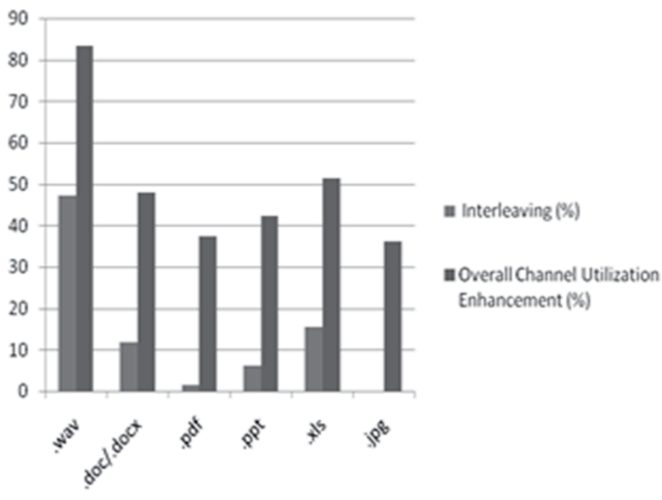

Fig. 5. Measurement simulation results for DITMC usage

use a path from many can lead to better performance in the term of packet reordering at the data receiver.

Problem 2: Active subscription control in smart spaces [6] for effective remote detection of information update by mobile participants (subscribers). In this case, each information subset that is interested to mobile participant (i.e., subscription to the subset) can be treated similar to a GSM channel, and the participant selects the waiting period for update in each subscription. In particular, we experimentally showed that rational selection of periods between subsequent checks reduces the number of notification losses at the subscriber while preserving low bandwidth utilization in terms of active check sent from the subscriber.

Problem 3: Supplementary processing on an edge device for information sharing in smart spaces [13] when in addition to its primary function the target device performs data processing delegated by surrounding participants (clients). In this case, each participant can be treated similar to a GSM channel, and the target device selects the no-service period for each client. In particular, we experimentally showed that for a wireless router in a local area network about 40% of its capacity can be used for supplementary processing, while keeping satisfactory service quality for the primary function—packet routing.

Problem 4: Selection of appropriate devices among surrounding volunteers [14] when an underutilized device can make some processing, i.e., active involvement into the system even small devices. In this case, each volunteer device can be treated similar to a GSM channel, and its underutilized period is used to perform supplementary work. This problem is close to Problem 3 above. In contrast to Problem 3 (where the target device performs the tasks needed for other participants), the target device rationally delegates its processing tasks to some occasional participants.

In sum, the above application domains show the direction towards smart communication. Traffic transmission becomes more effective when participants are actively involved into the transmission process as well as data are transformed (processed) during the transmission.

VI. Conclusion

We can conclude that the proposed DITMC-based methodology can be efficiently applied in reducing the traffic congestion in GSM networks and in other cases with mobile communication. The network transmission performance is improved (with minimum hardware complexity) to a good extent (in the order of 40%) leading to enhanced capacity in a GSM network. In the GSM case, the simulation experiments showed that the overall channel utilization becomes 83% from the original 36%. More development on the underlying mathematical model is needed to elaborate the "goodness of fit" property with verification within real-life scenarios. Development and experimental study of possible extensions of the DITMC-based methodology to other application scope form the direction of our further research.

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