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JOURNAL OF TELECOMMUNICATIONS AND INFORMATION TECHNOLOGY

Preface

We are immensely pleased to introduce you to the latest issue of the *Journal of Telecommunications and Information Technology* quarterly. The articles presented in this edition cover some important problems experienced in the field of broadly understood telecommunications (and specifically in wireless communications). The current issue of the Journal contains twelve papers.

The first six articles bring up issues related to various problems experienced in wireless communications.

The first article, titled *A Study of Wireless Sensor Networks to Comprehend their Relevance to Different Applications* is written by J. Agarkhed, P. Y. Dattatraya, and S. Patil. In this paper, the significance of WSNs is illustrated, with the focus placed on demands and understanding of the research problem and on analyzing it based on literature review.

The second article, titled *Passive Node-Assisted Wireless Networks Forming Raptor Codes with Random Access*, by D. Kusuma Hendraningrat, I. N. Apraz Ramatryana, and B. Narotama, considers wireless networks with passive nodes forming a pre-code of Raptor codes, also known as Raptor code-structured wireless networks (RCS-WN).

The next article, titled *Simulations of the MAC Layer in the LoRaWAN Networks* (M. Piechowiak and P. Zwierzykowski) considers communications-related problems in IoT (Internet of Things) systems. The article provides an overview of major LoRa MAC network simulators. It presents and comments on the most important research results obtained by the authors of the aforementioned software.

The next paper by A. Jamoos is titled *Estimation of UFMC Fading Channels Using H_∞ Filter*. The estimation and equalization of UFMC fading channels based on the comb-type pilot symbols arrangement, are considered in this paper.

The paper titled *First and Second Order Characteristics of a Wireless Three-hop Relay Channel with the Presence of Rayleigh Fading* is written by D. Krstic, P. Nikolic, Z. Popovic, and M. Stefanovic. This paper analyses a wireless three-hop relay communication system operating over the Rayleigh multipath fading environment.

A new hybrid method of solving integral equations concerning electric field in the time domain, as used for modeling linear antennas with pulse excitation, is proposed in the article titled *Extrapolation of Electromagnetic Response from Linear Antennas in Time Domain without Late-Time Instabilities in Numerical Solution of EFI Equation* (A. Witenberg). The method consists in a mixed, numerical-analytical description of the process that helps in keeping the stability of calculations over late time.

Three subsequent papers are concerned with problems related to analytical modeling of telecommunications systems. The paper titled *Performance Evaluation of the Threshold Call Admission Policy in Multi-rate Loss Systems* by I. Moscholios, I.-A. Chousainov, P. Panagoulas, P. Sarigiannidis, and M. Logothetis considers a multi-rate loss system of a single link that services random or quasi-random traffic under the TCA (Threshold Call Admission) policy. The analysis presented is based on reversible Markov chains.

A new multirate loss model for the call-level analysis of a two-link system with restricted accessibility that accommodates Poisson or quasi-random arriving calls of different service classes is proposed in the next article titled *Performance Evaluation of Two-Link Multirate Loss Models with Restricted Accessibility* (I. P. Keramidi, I. D. Moscholios, P. G. Sarigiannidis, and M. D. Logothetis). In the system under consideration, each link may support calls offloaded from other links.

The last article from this group is titled *Modeling of Complex Non-Full-Availability Systems* by S. Hanczewski, M. Stasiak, and M. Weissenberg. This article presents an analytical model of a complex non-full-availability telecommunications system. The model may be used, in the future, for analyzing cloud computing systems.

The next paper, titled *V2X Communications for Platooning: Sensor Inaccuracy Aspects* by M. Sybis, P. Sroka, and P. Kryszkiewicz, discusses the impact of sensor inaccuracy on the performance and behavior of autonomous vehicle platooning that relies on wireless communications aided with context information sourced from various databases and maps. Platooning is a potential future application of autonomous driving.

The next paper, titled *An Expert System for Automatic Classification of Sound Signals*, by K. Tyburek and P. Kotlarz, presents the results of research concerning classification methods used for recognition/classification of audio signals. In this work, a particular emphasis was placed on fine-tuning the fuzzy classifier model and on neural classifier applications, taking into account new neural networks that have not been studied so far for sound classification purposes.

The last article is titled *Retransmission as a Replacement of Forward Error Correction in Noiseless LV Networks Used for Narrowband PLC*. Its author, P. Kiedrowski, describes a project under which PLC technology is used in railway light signals. The solution proposed increases reliability of communication between railway light signals and railway automation centers, using the frame retransmission approach instead of deploying the FEC (Forward Error Correction) technique.

We would like to thank all the authors and reviewers for the effort they have put into preparing this issue of the *Journal of Telecommunications and Information Technology*.

Sławomir Hanczewski
Guest Editor

A Study of Wireless Sensor Networks to Comprehend their Relevance to Different Applications

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Abstract—Wireless sensor networks (WSNs) have experienced enormous growth, both in terms of the technology used and their practical applications. In order to understand the features of WSNs that make the solution suitable for a specific purpose, one needs to be aware of the theoretical concepts behind and technological aspects of WSNs. In this paper, the significance of WSNs is illustrated, with a particular emphasis placed on their demands and on understanding research-related problems. A review of the literature available is presented as well. Detailed discussions concerning sensor node architecture, different types of sensors used and their relevance for various types of WSNs is presented, highlighting the need to achieve application-specific requirements without degrading service quality. Multipath and cluster-based routing protocols are compared in order to analyze QoS requirements they are capable of satisfying, and their suitability for different application areas is reviewed. This survey highlights the performance of different routing protocols, therefore providing guidelines enabling each of the routing techniques to be used, in an efficient manner, with factors such as specific network structure, protocol operation and routing path construction taken into consideration in order to achieve better performance.

Keywords—clustering, energy efficiency, multipath-based routing, wireless sensor network.

1. Introduction

WSNs are an evergreen technology due to the fact that they support different types of applications, which makes them perfectly suited for use in such areas as security and surveillance, environmental monitoring, health monitoring, factory process automation, home automation, as well as tracking animals, objects, vehicles, and humans [1], [2]. The sensors used in WSNs are very small and offer limited processing power and computing capabilities. These sensors may collect or process information, and transmit it to the sink node [3].

When sensors are deployed at remote and difficult-to-access locations, wireless communication is used to transfer the

data to a base station (BS). A battery is the primary source of power for a sensor node. Alternatively, power may be harvested using solar panels. Considerable amount of research has been conducted on the design of various data routing mechanisms, with strict WSN resource-related restrictions taken into consideration. The latest advances in the field of micro-electro-mechanical systems (MEMS) and the ever-changing application-related demands require that the manner in which all layers of a WSN protocol stack are designed be modified as well. Quality of service (QoS) demands pose a serious challenge while designing routing mechanisms that boost performance, due to the unique characteristics and specific features of WSNs.

1.1. Characteristics of WSNs

The routing protocols of a WSN are unique in order to satisfy the requirements of various applications. These include, for instance:

- limited resource availability (power, storage capacity, computing capacity),
- deployment in a hostile environment causes difficulty with recharging batteries of sensor nodes,
- self-organizing nature of sensor nodes causes frequent topology changes,
- heterogeneous nature of WSNs may cause interoperability problems,
- routing mechanism-based reconfiguration in the case of topology changes due to node failures, link breakups,
- scalability is an issue in massive network deployments,
- processing of unattainable operation is necessary in various application environments.

Common WSN-related research issues focus on ensuring high bandwidths, low energy consumption, QoS provisioning, node mobility, congestion control, congestion detection, congestion mitigation, reliability, end-to-end and hop-by-hop packet recovery, cache ACK/NACK, scalability, synchronization, data cache, data aggregation, computational overhead, availability, data security, and integrity [4], [5]. The provision of QoS support in WSNs while ensuring energy efficiency is an emerging area of research that needs to take into consideration numerous challenges [6].

The following WSN-related challenges have been identified while working on an efficient protocol design [7], [8]:

- **Limited resource constraints.** An efficient routing protocol needs to consider the energy status of the node, available bandwidth, storage, limited buffer size and computing capabilities in order to achieve high data rates with limited transmission power.
- **Heavy node-to-sink traffic.** This demands efficient utilization of available bandwidth and load balancing.
- **Redundant data transmission.** Data redundancy helps achieve reliability but increases power consumption.
- **Network dynamics.** Routing protocols need to periodically route packets which may suffer from a node or a link failure, and must establish links dynamically.
- **Node energy conservation.** Data transmission through the same path, retransmission of lost data and control messages depletes node energy.
- **Transmission power.** A heterogeneous environment with multimedia data demands minimization of transmission power.
- **Scalability.** The network has to attain desired performance levels, even with an increase in network size.
- **Time constraint transmission.** Critical and non-delay tolerant applications demand the transfer of data within a specified deadline.
- **Security.** For secure data transmission, several cryptographic, steganographic and other techniques are used.

Due to the many constraints in the traditional WSNs, new technologies need to be considered to overcome these and many other issues [9] and to offer a better understanding of the underlying sensor network.

The remaining part of the paper is constructed as follows. Section 2 presents details about sensor network architecture. Section 3 details the classification of WSNs based on node deployment type, delivery model, location, type, and nature. Section 4 specifies the applications of WSNs

and in Section 5, routing is discussed. Section 6 describes two broad routing categories and security aspects, while Section 7 presents the conclusions.

2. Sensor Network Architecture

WSNs consist of a vast number of sensor nodes that are distributed throughout the areas of interest, as shown in Fig. 1. WSNs comprise lightweight, low powered sensor nodes with the capability of acquiring, computing and communicating data (wirelessly). These sensor nodes communicate with each other and with the destination node that is also referred to as a base station or sink.

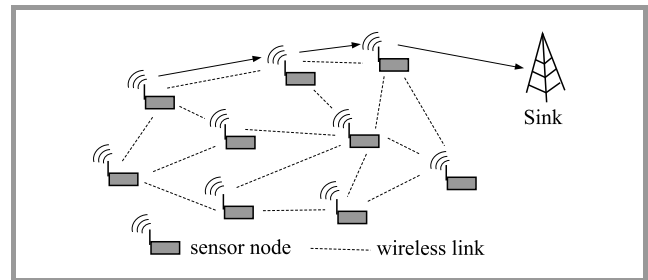


Fig. 1. Structure of WSNs.

Sensor nodes have some computing capability as well. They may be stationary or mobile, homogenous or heterogeneous. Homogenous sensor nodes are characterized by the same capacity in terms of power, computation and communication. Heterogeneous sensor nodes may perform sensing activities and other tasks, such as data relaying.

The basic components of sensor nodes are shown in Fig. 2.

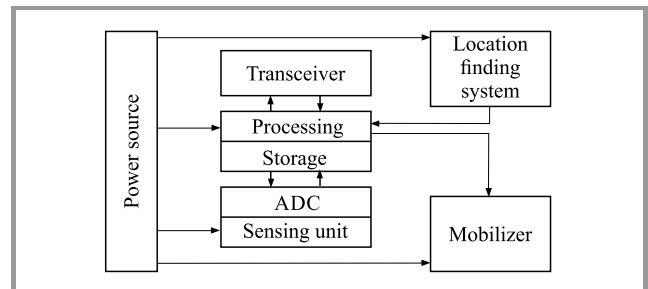


Fig. 2. Basic structure of sensor node.

The battery may be a disposable or a rechargeable power source. Since the nodes are deployed at a remote location, the design must ensure that the network is capable of functioning over long periods of time, with power preserved to extend the network lifetime [3].

The transceiver unit facilitates the radio link with neighboring nodes, or internally, within a subsystem of a nodes. This unit supports several operational states (sleep, idle or active) for efficient resource utilization. The transceiver unit may support different transmission media, such as radio frequency (RF), infrared, and laser [10].

The processing unit is the core element of a node that determines its computational and performance capabilities. It contains a microcontroller, a digital signal processor or field-programmable gate arrays (FPGAs). It performs various functions, such as turning the sensor on, converting data to the digital format, aggregating data, as well as processing data [11].

The sensing unit consists of a sensor and an analog-to-digital converter (ADC). It collects data from the environment. The ADC serves as an interface between the sensor and the processor [12].

The sensors are classified as physical, chemical and biological sensors:

- **Physical sensors** are used for measuring vibration, acceleration, ultra-sound, water level, stress, flex, bend, and strain.
- **Environmental sensors** are used to measure air temperature and humidity, soil humidity, humidity effect on leaves, wind direction and speed, air pressure, etc.
- **Gas sensors** allow to measure e.g. CO, CO₂, CH₄, NH₃, O₂, NO₂, SH₂ and pollution levels.
- **Optical sensors** are used to measure sunlight, infrared and ultraviolet radiation. These sensors detect human presence using the IR spectrum. They are also used in agriculture, where sunlight, ultraviolet and radiation sensors measure the amount of energy and sunlight absorbed by plants.
- **Biometric sensors** perform electrocardiograms (ECG) and measure perspiration and pulse to prevent heart attacks.

Data sensed by the sensor node will be collected and routed to a more powerful node called the sink node.

3. Classification of WSNs

WSNs are categorized based on deployment type, delivery model, location of deployment, type and nature of node. Two deployment types are distinguished: deterministic and non-deterministic.

Non-deterministic deployment means that nodes are scattered randomly and creates an ad-hoc infrastructure, thus forming an unstructured WSN. Detecting sensor node failures and managing connectivity is difficult in unstructured WSNs. In this scenario, sensor nodes communicate with each other directly establish a link with BS.

Deterministic deployment allows the sensor nodes to be manually placed in a pre-planned manner at specific locations, in order to guarantee the coverage of a specific region and forming a structured WSN. In this scenario, data is routed through predetermined nodes.

Data is transferred using a specific data delivery model that is application dependent and classified as continuous,

query-driven, event-driven or hybrid (Fig. 3). In the continuous data delivery model, each node sends the data periodically. The event-driven model uses triggering based on a specific event determined in the target tracking application. In the query-driven model, data is transmitted in a reply to a query generated by the sink, in environment or habitat monitoring applications. The hybrid model combines the aforementioned features.

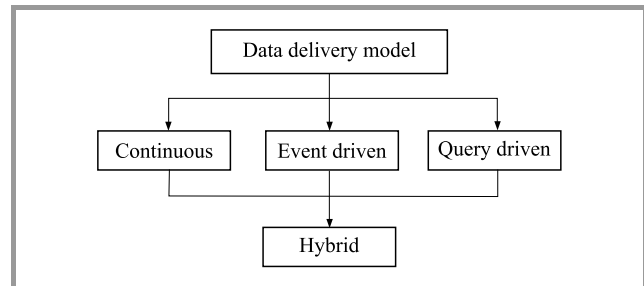


Fig. 3. Data delivery model.

WSNs are classified into different types based on the location of their deployment [2], [13]:

- **Terrestrial.** WSNs of this particular type rely on either ad-hoc or pre-planned deployment. In an ad-hoc deployment, the nodes are randomly dropped into the specific target area. Pre-planned implementation is based on an optimal placement grid;
- **Underground.** The network is formed by sensor nodes placed in mines or below ground to monitor underground conditions. The equipment required here has to support reliable communication through rocks, water, soil, etc.;
- **Mobile.** This network is formed by sensor nodes positioned in vehicles. Long propagation delays, limited bandwidth and signal fading are the most important issues here;
- **Body area.** Networks of this type are formed by the deployment of disposable, inside-the-body or on-the-body, low cost sensor nodes equipped with cameras.

Next, we may classify WSNs based on the type of nodes used.

In a homogenous network, all nodes have an equal capacity in terms of power, communication and computation. A heterogeneous network, in turn, may sense different parameters, such as humidity, temperature and pressure. Its sensors may also detect motion and capture images, videos, etc.

Finally, networks may be of the static variety, with nodes fixed at a predetermined location, or of the mobile variety, using frequent topology changes.

4. WSN Applications

WSNs are applied in a wide range of fields. Specific applications may be divided into two broad categories, as shown in Fig. 4 and summarized in Table 1 [14]–[24].

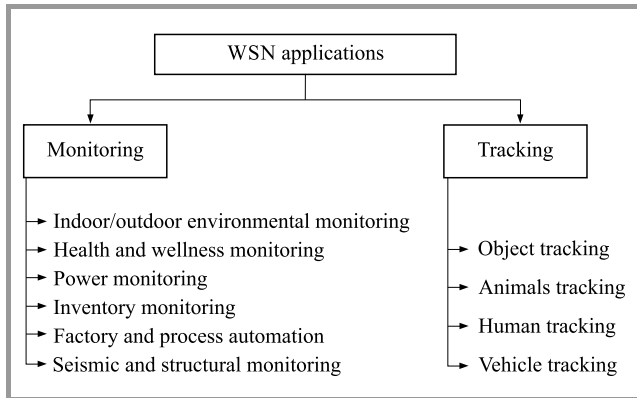


Fig. 4. WSN applications.

The performance of a WSN system may be evaluated by the use of a testbed, with specific performance parameters considered [9]. A WSN testbed is costly, difficult to design and time-consuming [25]. Hence, simulations may be performed using either an emulator or the simulator discussed in [26].

In consideration of the above, the design of a WSN depends, to a significant degree, on its specific application. It should also consider such factors as the environment, design objectives, hardware required, cost and overall system constraints [3], [27]. As WSN applications require the deployment of sensor nodes in a harsh environment, continuous functioning of the nodes is needed even in the event of a network infrastructure failure [4]. This necessitates that the network be designed in a proper manner, from deployment to data routing, with energy conservation being one of the most critical and challenging issues.

5. State-of-the-Art Routing

A vast literature survey concerning WSNs helps understand that an energy-efficient routing protocol is desirable in WSNs, as the lifetime of the network depends on the battery life of its sensor nodes. The essential energy conservation technique consists in choosing a proper network topology. Increased reliability [28] of the transmitted information may be achieved by routing it through a flexible path, with the ability to re-establish the route in the event of a node or link failure.

In the flat network topology (Fig. 5), all nodes perform identical functions [29] and communicate directly with the sink node, increasing transmission-related energy consumption and the communication overhead. Flat network topologies include, for instance, sequential assignment routing (SAR) [30], sensor protocols for information via negotiation (SPIN) [31] and greedy perimeter stateless routing

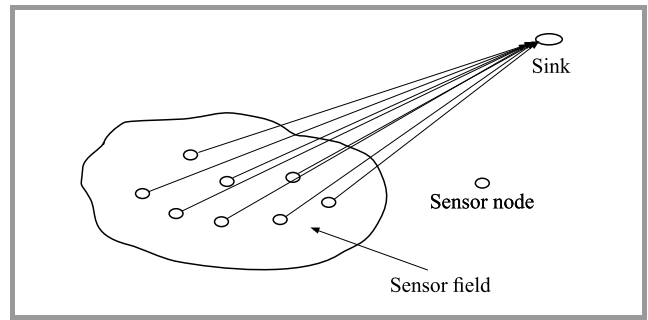


Fig. 5. Scheme of WSN with flat-based routing protocols.

(GPSR) [32]. All of them are based on application-related requirements and rely on star, mesh or tree topologies. Cluster-based routing (Fig. 6) is formed by subdividing a flat network into a set of small individual clusters. Each cluster selects a node with the maximum energy level as its cluster head (CH), and all remaining nodes are considered to be respective cluster members (CMs). The primary task of the CH is to collect data from CMs within the cluster and to transmit the aggregated data to the next hop neighboring CH, until it reaches the BS [33]. Such an approach allows to save energy, as the CH is the only node that needs to be active all the time and the CMs are only used for sensing and may be periodically turned on and off to save energy. The role of the CH is assumed, alternately, by different CMs to avoid an early shutdown of a single CH due to it being continuously used for transmission purposes. Time-driven CH rotation is simple and offers a fixed cluster setup time. Time-driven CH rotation is a highly efficient solution.

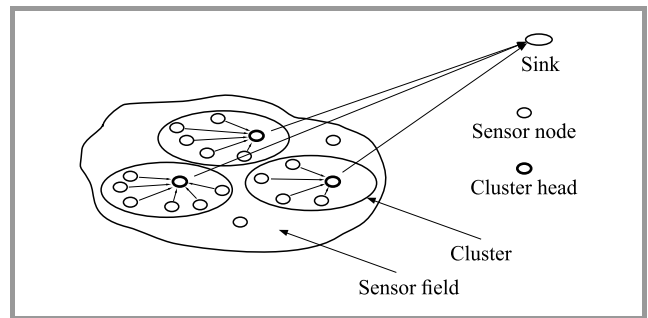


Fig. 6. An example of a WSN with cluster-based routing.

Reliability of information transmission is improved by multipath routing. This technique is used for balancing the load to avoid congestion, and to reduce the repeated route inquiries. It offers the lowest overall routing overhead and provides enhanced robustness against failure of a node, to achieve reliability and energy efficiency [34]. A lot of routing protocols are cluster-based or multipath-based in order to meet the service quality requirements. Designing a routing protocol in a manner that satisfies the requirements of a given application, simultaneously taking into consideration energy conservation and service quality, is a challenging task.

Table 1
WSN applications

Application area	Monitoring	Tracking
Military	Friendly troop status [19], vehicles and their movements	Availability of equipment on the battle-field [20], [21]
Industrial control	Monitoring applications follows the data-driven model to monitor material flows, quality, and safety in order to reduce human-made errors and the cost of manufacturing [22]	Changes in plant security, machine processes, or mishandling by operator [23]
Surveillance	Used to monitor airports, public parks, and railway stations [24]	Tracking of intruders
Environmental	• detect volcano eruptions [14], • detect forest fires by collaboration of solar-powered sensor nodes [15], • estimate floods by monitoring water level in rivers [16]	• identifying air pollution levels caused by increasing traffic volumes, • movement of birds, animals and insects
Agriculture	Monitor water, humidity and temperature levels in order to achieve good quality farming produce [17]	Tracking the movement of animals
Health	Sensors monitor blood oxygen level, blood pressure, body temperature and the glucose level [18]	Assisting doctors in emergencies

The main objective of routing in WSNs is to conserve energy. Therefore, energy-efficient routing protocols are desired in WSNs, as the lifetime of the network relies on the battery life of each node. The well-known underlying protocol, referred to as low energy adaptive clustering hierarchy (LEACH) [35], allows CHs to communicate with sink directly. A cluster formed by a set of sensor nodes is based on the signal strength of the nodes and uses CH as a router-to-sink, with the constraint that a maximum of 5% of nodes may serve as CHs of a given network. LEACH offers lower energy consumption than direct communication, as it establishes a distributed system without any global knowledge. LEACH is not relevant for large size networks where selecting the CH is difficult. Its specific variant, LEACH-F, has a fixed number of clusters and alternately assigns the role of the CH to different nodes within the clusters. However, it does not support energy saving, meaning that it is not suitable for mobile applications.

An increase in the distance between the CH and the sink, in turn, increases communication cost and shortens network lifetime. Communication distance between the nodes is minimized by dividing the network into an optimal number of clusters, as specified in the optimal number of clusters algorithm (ONCA) [36]. Power-efficient gathering in sensor information systems (PEGASIS) is a technique that uses multi-hop routing by forming a chain of sensor nodes and selecting only one node to transmit to the sink, instead of multiple nodes, as in LEACH. It outperforms LEACH by aggregating data from the nodes. One drawback of PEGASIS is the increased delay caused by the chain formation process. This challenge is overcome by the hierarchical variety of PEGASIS. This version uses simultaneous trans-

mission of messages, which may lead to collisions that are avoided by the use of signal coding.

Chain-based CDMA-capable protocols support parallel transmission of data to minimize delay, but need dynamic topology adjustments to conserve energy [37]. A threshold-sensitive energy-efficient sensor network (TEEN) is an example of a hierarchical protocol that responds to a sudden change in the sensed attributes, such as temperature. It is useful in time-critical applications relying on networks operating in the reactive mode. The closest nodes from a given cluster communicate with each other and the process is repeated from the second level, until the base station is reached. The protocol uses either a soft or a hard threshold to trigger the sensor node to switch its transmitter on. Data transmission is controlled by varying these thresholds. TEEN is not useful for periodic reporting applications [38]. The adaptive threshold-sensitive energy-efficient sensor network protocol (APTEEN) captures periodic data and responds to time-critical events. The performance of TEEN is better than that of LEACH and APTEEN [39]. The hybrid energy-efficient distributed protocol (HEED) is useful in ad-hoc sensor networks. HEED assumes that all nodes have the same energy initially. Then, all nodes appoint themselves cluster heads based on their remaining power and based on the degree of a given node relied upon for cluster election. It is a cluster-based multi-hop routing protocol. HEED is characterized by good load balancing and uniform CH distribution. HEED does not offer balanced energy consumption due to the fact that a higher number of CHs is generated [40].

A useful way to save energy is to schedule data transmission processes that reduce the amount of sensing power and

preserve the quality of sensed data. One such algorithm, known as CIES, shares sensing error information and controls sensing errors through neighbor node coordination, therefore achieving high data accuracy and throughput [41]. It uses clustering and groups the sensor nodes into different clusters in order to meet scalability requirements, to balance the load in high-traffic networks, to minimize the size of the routing table at each node, and to efficiently utilize the communication bandwidth by relying on inter-cluster interaction with the CH [42].

In order to conserve energy, CH may schedule the cluster's activity by toggling between the node's active and sleep modes [43], [44]. The author in [45] presented a technique to optimize sensor node energy utilization by clustering, using a genetic algorithm. The paper [46] considered each sensor node's residual energy, the amount of energy used in sensing, the node's distance from the base station, and the number of neighboring nodes – all in order to identify an optimized, dynamic network structure.

Some of the routing protocols referred to above use single path routing, where each sensor node routes its data through the shortest path to the sink. Single-path routing may cause the routing path to break up due to the failure of the nodes and demands that a new route be established. It also increases energy consumption and the probability of a failure of a node which results in the packets being dropped and cause increased delay in the transmission to the sink. In such a case, it is desirable to avoid node failures by choosing an alternative routing path, thus forming multipath routing that increases the number of potential routes, guaranteeing robustness of transmission and increased throughput.

Single path routing does not offer a sufficient bandwidth for data transmission, as WSNs have a limited bandwidth capacity. Multiple routing paths facilitate simultaneous communication, and the aggregated bandwidth of the multiple routing paths is capable of satisfying significant bandwidth demand of any particular application. An increase in bandwidth overcomes delay and guarantees load-balanced energy utilization, along with increased network lifetime [47], [48]. Service differentiation deployed to differentiate and prioritize traffic flow is another mechanism relied upon to meet application-related requirements.

6. Classification of Routing Protocols

Based on extensive work on routing techniques in WSNs, routing protocols may be classified into the following broad categories: flat, hierarchical, location-based, multipath based, query-based, negotiation-based, QoS-based, coherence-based, proactive, reactive and hybrid.

The classification of routing protocols distinguishes also data-centric based, location-based, cluster-based, node mobility-based, multipath-based and service quality-based varieties [4]. The best routing protocol must offer the maximum residual energy of nodes, negligible packet drop rates, lower energy consumption, lower delays and load fairness.

6.1. Multipath Routing Protocols

One of the most significant advantages of multipath routing is that it minimizes route updates, balances the traffic and reduces the energy consumption of each sensor node. The comparison of multipath routing protocols presented in Table 2 highlights the strong points of the solution. In [49], the author proposed a maximally radio-disjoint multipath routing (MR2) that uses an incremental approach which helps choose alternate paths but leads to higher delays with increased overhead compared to hop count multipath routing (HCMR). HCMR is characterized by higher energy usage compared to the energy efficient interference-aware multipath routing (EEIAMR) protocol. It determines short alternate paths to minimize delay and energy consumption by relying on the large network size. With an increase in node count, MR2 and EEIAMR generate higher overhead compared to HCMR. Sensor protocol for information via negotiation (SPIN) provides robustness in the event of path failures, by relying on topology changes [44]. Reliable, real-time routing protocol (3R) considers both multipath and time constraints. This protocol [45] uses packet reception rate estimations to calculate the necessary number of forwarding paths. The disjoint path routing protocol [46] finds the first k-shortest route to the sink and sends periodic messages to keep these paths alive. Performance improvements and fault-tolerance are achieved by equal division of load among the nodes.

Table 2
Multipath-based routing protocols

Protocol	Number of paths built	Interference awareness	Mean delay	Energy consumption	Over head
MR2 [49]	6–8	Yes	Higher	Lower	Higher
HCMR [9]	2–4	No	Lower	Higher	Lowest
EEIAMR [9]	1	Yes	Lowest	Lowest	Lower
SPIN [44]	1 or more	No	Higher	Lower	Lowest
3R [45]	More	No	Lowest	Higher	Lower
Disjoint path [46]	K	No	Lower	Higher	Higher

6.2. Cluster-Based Routing Protocol

In WSNs, the nodes are grouped into a cluster to enable load balancing, network scalability, to minimize the routing table size at the level of an individual node, to spare the communication bandwidth by inter-cluster interaction with CHs, and to ensure a stabilized network topology [45]–[48]. For energy conservation purposes, activities within the cluster are scheduled by the CH which toggles between sleep and active state of the nodes. Another advantage of clustering consists in the fact that CH aggregates data from all sensors in its respective cluster to reduce the count of packets to be sent [49]. Advantages of the cluster-based routing protocols are presented in Tables 3 and 4.

Table 3
Cluster-based routing protocols with varying packet delivery ratio and end-to-end delay

Protocol	Feature supported	Delivery ratio	End-to-end delay
LEACH [11]	Minimizes energy consumption but incurs more delay by using large buffer size	Medium	Higher
CBEEQR [10]	Uses large buffer size, leading to lower packet drop rates	Increased	Lower
ONCA [12]	Increases network lifetime by dividing the network into an optimal number of clusters	Higher	Lower
TEEN [38]	Responds to a sudden change in a sensed attribute, based on a hard or soft threshold. Supports event-driven applications and is not suitable for periodic applications	High	Lower
APTEEN [39]	Captures data on a periodic basis and reacts to the time-critical events. Supports three types of queries historical, one-time and persistent	High	Lower

Table 4
Cluster-based routing protocols with varying energy consumption

Protocol	Time [s]	Number of nodes alive	Routing type	Energy consumption
LEACH [5]	800	0	Proactive	High
	500	20		
Leach-C [36]	800	0	Proactive	Medium
	500	20		
TEEN with a hard threshold [37]	800	2	Proactive	Lower
	500	95		
TEEN with a soft threshold [38]	800	95	Proactive	Lowest
	500	100		

The majority of location-based protocols are not very energy efficient or reliable and sacrifice QoS requirements, which leads to transmission delays. They are not applied as frequently in healthcare, but may be useful for environmental monitoring or target tracking, provided that newly designed protocols place a greater emphasis on energy efficiency and reliability-related aspects. Cluster-based and mobility-based routing protocols are best suited for healthcare and military applications which require reliable transfers and a continuously operational environment. This requirement is met by mobile devices with batteries whose power capacity is higher compared to the conventional sensors.

Multipath based protocols are suited for military and medical applications, since they ensure high reliable data delivery rates. Appropriate cluster design and the selection of CH minimize energy consumption while communicating and aggregating messages and are highly significant design features. Data transmission is performed to conserve energy and achieve QoS with an efficient routing technique, in order to ensure efficient use of the sensor.

Application-related QoS requirements may be satisfied by assigning different priority levels to different types of applications, users, frames, data flows and packets through resource sharing. High quality of service is ensured by such parameters as delay, bandwidth, and packet loss [50]–[59].

Service quality takes into consideration hard and soft QoS factors. Routing protocols with hard QoS must meet strict requirements concerning delay, packet loss, and bandwidth. On the other hand, routing protocols with soft QoS may violate these provisions. Both hard and soft QoS are guaranteed through service differentiation. The two service differentiation models are the integrated service model (IntServ) [60] and the differentiated service model (DiffServ) [61]. They are presented, along with their key features, in Table 5.

Table 5
Service differentiation model

Integrated service model	Differentiated service model
Service quality maintained on the per-flow basis	Service quality maintained on the per packet-basis
The approach relied upon reservation-based and is either data-centric or host-centric	The method used is reservation-less and relies on the multi-hop strategy to conserve energy
Cannot achieve guaranteed service quality due to the varying capacity of the channel, needs to maintain sensor state on the per-flow basis, scalability issues	Higher memory requirement due to the fact that each node acts as a source or intermediate node

The security of data sensed by a particular sensor is an issue that has been continuously gaining in importance. Security algorithms must not boost memory requirements and must not lead to increased power consumption during processing and transmission, as the sensor node suffers from resource constraints. The various security requirements include confidentiality, authentication, integrity, and availability [62]. The various cluster-based WSN protocols that apply different security methods to prevent different types of attacks, are presented in Table 6 and Table 7.

Advancements in sensor network technologies are an enabler of the Internet of Things (IoT) – a technology that shapes the world anew by offering the ability to measure, infer and understanding environmental indicators. Recent developments and technological improvements have increased the efficiency and lowered the cost of devices, enabling

Table 6
Cluster-based security protocols

Protocol	Security method	Type of security achieved	Computation overhead	Communication overhead
CHiMAC [63]	Message divided into two blocks and encrypted. Each block added to message with source, destination, and cluster ID	Authentication and integrity	High due to encryption and decryption	Lower due to clustering
Sybil attack model [64]	A malicious node uses all of its Sybil identities to join each cluster. RSSI and positioning using three points to defend against the novel attack has been considered	Detects 99.8% of Sybil nodes with 0.08% false detection rate	High	Lower
HMAC [65]	Hash-based message authentication code (HMAC) authentication followed by pair-wise key establishment used during data aggregation for two different types of node set up processes. It uses a pair-wise key generation process in which each node randomly generates a unique key with combining private and public key components	Authentication	Lower	Lower
KCLP [66]	To protect the location of the real source and the sink, a fake source and a fake sink have been used along with k-means clustering to elongate the routing path and to boost safety. The use of different routing patterns of the fake and the real packet has reduced network delay	Jamming attack, node collaboration	High	High
DH method [67]	Security guaranteed at each hop by using a binary hop count and end-to-end node authentication relying on Huffman coding. Dijkstra's algorithm finds the nodes with maximum energy level and optimum distance path	Message and node authentication. Integrity achieved by encryption	Medium	Lower

Table 7
Different types of attacks

Type of attack	Classification into specific attack type	Description	Security methods applied
Common attack	Eavesdropping	Attacker retrieve information from transmitted data	Encryption technique and time stamp
	Message modification	Packet data modified while data in transmission	Provide authentication and identification of traffic, pushback
	Message replay	Adversary retransmit the contents and packet	
Denial of service attack [63]	Node collaboration	Some nodes act as maliciously and prevent message transmission to other nodes in the network	Use multiple disjoint routing paths and diversity coding
	Jamming attack	An adversary jams communication channel	
	Exhaustion power	Repeated packet request lead to depletion of battery power	
Node compromise attack	Node compromise	The attacker gain control of sensor node and launch attacks from compromised nodes	Trusted key server checks for node identification
Impersonation attack		Malicious node impersonates a legitimate node and uses its identity to mount an active attack such as Sybil [64] or node replication	BS or key server authenticates every node on the network
Protocol specific attack	Spoofed routing	Adversary spoof routing information by corrupting routing table content	Provide access to the authenticated node
	Selective forwarding	Selectively forwarding of packets that traverse a malicious node depending on some criteria	Use multiple disjoint routing paths
	Wormhole attack	Creation of wormhole that replays information from one location to other location	Each message is forwarded individually, choosing the next-hop node to be the neighbor closest to the ultimate destination
	Hello flood attack	Hello flood attack create false control packets	Check bi-directionality of the local links

their large-scale use in remote sensing applications. Our smartphones offer a diverse range of sensors as well and, consequently, they may operate a variety of mobile apps relevant for the IoT [68]. The most challenging task is to process huge amounts of data generated by sensors, taking into account security-related issues, energy and network limitations, as well as various types of uncertainties.

7. Conclusion

The architecture constrained sensor node for use in a different application domain demand an energy efficient routing protocol without losing service quality. From the compressive study of routing protocols, most of the location-based protocols are not much energy efficient, reliable, and sacrifice QoS requirements leading to a delay in transmission. Thus they are not appropriate for healthcare applications but can be useful for environmental monitoring or target tracking application if new protocols designed considering energy efficiency and reliability issues. Cluster-based and mobility-based routing protocols are best for healthcare and military applications. These applications require a reliable and continuously functioning environment that can be achieved by a mobile device equipped with a higher amount of battery power compared to conventional sensors like biosensors. Multipath-based protocols are suited for military and medical applications since they ensure reliable data delivery. Appropriate design of cluster and selection of CH minimize energy consumption during message communication and aggregation, which is one of the most significant design issues. To develop an efficient protocol, one must combine the best features of protocols of different routing categories.

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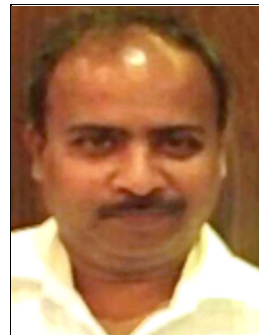
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Passive Node-Assisted Wireless Networks Forming Raptor Codes with Random Access

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Abstract—In this paper, wireless networks with passive nodes forming a pre-code of Raptor codes, also known as Raptor code-structured wireless networks (RCS-WN), are considered. In RCS-WN, the passive nodes utilize a decode-and-forward (DF) scheme and send the decoded data to the destination by using a scheduling scheme. We propose to deploy random access (RA) in passive nodes, to achieve a solution referred to as an RA-based RCS-WN (RAR-WN), and attempt to compare it with conventional wireless networks applying irregular repetition slotted ALOHA (IRSA). We investigate regular and irregular degree distributions for repetition codes to test the capacity of the proposed RAR-WN, using the extrinsic information transfer (EXIT) chart. This paper also examines the capabilities of the stopping set (both with regular and irregular degree distribution) in RAR-WN. Performance of the proposed system is evaluated in terms of EXIT chart, packet loss rate (PLR), and throughput. The results show that the proposed system is capable of achieving non-dropped throughput performance.

Keywords—EXIT chart, passive nodes, PLR, random access, Raptor codes, throughput, wireless networks.

1. Introduction

The qualities of 5G cellular communication systems have been determined in the International Telecommunication Union Recommendation (ITU-R), defining the novel approach as an enhanced mobile broadband (eMBB), massive machines-type communication (mMTC), ultra-reliable and low latency communication (URLLC) [1] solution. The development of communication techniques that involve a massive number of users supporting 5G applications mean that random access mechanisms turn out to be essential and crucial for wireless access [2]. Random access techniques are preferred, as conventional scheduling becomes difficult when the number of users connected to a given network is large.

In a random access technique, all users from whom no scheduling is required send data packets randomly, based on probability distribution, through selected time slots [3]. The problem of interference occurs when two or more users send data packets using the same time slot. These data packets, referred to as collided packets, are usually dis-

carded. In coded slotted ALOHA (CSA) [4] and irregular repetition slotted ALOHA (IRSA) [5] approaches, collided packets are resolved and stored for decoding using the successive interference cancellation (SIC) technique to improve the overall performance of the system. Both CSA and IRSA differ in terms of network encoding approaches applied. Maximum distance separable (MDS) codes are used in CSA, while repetition codes are used in IRSA. The SIC process depends on the selected time slots for sending the data packets. Practically, a pointer is used to indicate the position of each data packet in the selected time-slots. In CSA, the time slot with the packets sent in a frame is shown by a pointer. In IRSA, the pointer indicates the time slot in which a replica of the data packet is sent in the frame.

IRSA may asymptotically achieve the normalized throughput of 0.97 packets/slot [p/s] by optimizing the distribution degree of repetition codes. Normalized throughput is the probability of a packet being successfully decoded, multiplied by the traffic offered. In practice, the normalized throughput of IRSA is $T \approx 0.8$ p/s. In CSA, the maximum normalized throughput is $T = 1/e \approx 0.37$ p/s. Therefore, the fact that normalized throughput of IRSA is capable of approaching 1 p/s is very encouraging.

CSA and IRSA are interesting because random transmissions with the SIC process can be represented by bipartite graphs that resemble the codes-on-graph. Further, the authors in [6]–[8] propose the use of coded random access (CRA) in a massive multi-way relay, where repetition codes and MDS codes are exploited. Few literature examples discuss the CRA. In [9], a CRA with human priority is proposed to protect humans from machines, with repetition codes applied along with a set difference assigned to the access to each time slot per priority. On the other hand, the authors in [10] propose high-rate CRA using MDS codes with two priority groups for non-orthogonal multiple access (NOMA). In [11] and [12], the concept of CRA is further improved by applying multi-packet reception (MPR) to achieve higher throughput. The highest throughput is achieved when multi-user detection (MUD) is performed at each time slot. In this paper, no MUD or MPR are performed.

However, one of the unsolved problems in CRA is the dropped throughput performance after reaching a value that exceeds the threshold. A simple approach, known as Raptor code-structured super-dense wireless networks (RC-SWN), is proposed to achieve favorable non-dropped throughput by using passive nodes that function as pre-code Raptor codes [13]. In [14], sub-optimal degree distributions of repetition codes in Raptor code-structured wireless networks (RCS-WN) are proposed to achieve high performance in terms of non-dropped throughput. In RCS-WN, passive nodes decode and forward data to the destination with a scheduling scheme. In this paper, we propose the random access (RA) approach for accessing passive nodes in order to simplify RCS-WN, in connection with both user nodes and passive nodes.

The contributions of this paper are listed as follows:

- RA technique for passive node access in RCS-WN, known as RA-based RCS-WN (RAR-WN) is proposed, to simplify the access of RCS-WN – a solution that is crucial for analyzing the capability of serving more users or of maintaining a lower number of time slots;
- regular and irregular degree distributions for repetition codes in RAR-WN are investigated, which are not discussed in detail in [13]–[15]. Optimization of distribution degrees is important to evaluate the capacity of RAR-WN based on the extrinsic information transfer (EXIT) chart [16];
- six most important stopping sets for both regular and irregular degree distributions in RAR-WN are derived.

The rest of this paper is organized as follows. Section 2 presents structure and graphical representation of RAR-WN. Section 3 presents induced of degree distributions. The examples of the decoding strategy and EXIT chart analysis are presented in Section 4. Section 5 measures the performance of RAR-WN in terms of throughput and packet loss rate (PLR). Finally, Section 6 concludes this paper.

2. Structure and Graphical Representation

Let us consider an RAR-WN serving a massive number of users. Users $U_m = \{u_1, u_2, \dots, u_M\}$ transmit data packets to the destination (denoted as D). Figure 1 shows the structure of wireless networks without and with passive nodes $P_q = \{p_1, p_2, \dots, p_Q\}$ forming a pre-code of Raptor codes, respectively. With this condition taken into consideration, users and passive nodes are served by D in RAR-WN.

Then, RAR-WN is modeled in the manner shown in Fig. 2. The circles represent user nodes (UN), the dark circles represent passive nodes (PN), and the squares represent slot nodes (SN). RAR-WN is denoted as $\text{RAR}(\eta, Q)$, where η represents the number of users connected to each PN and Q

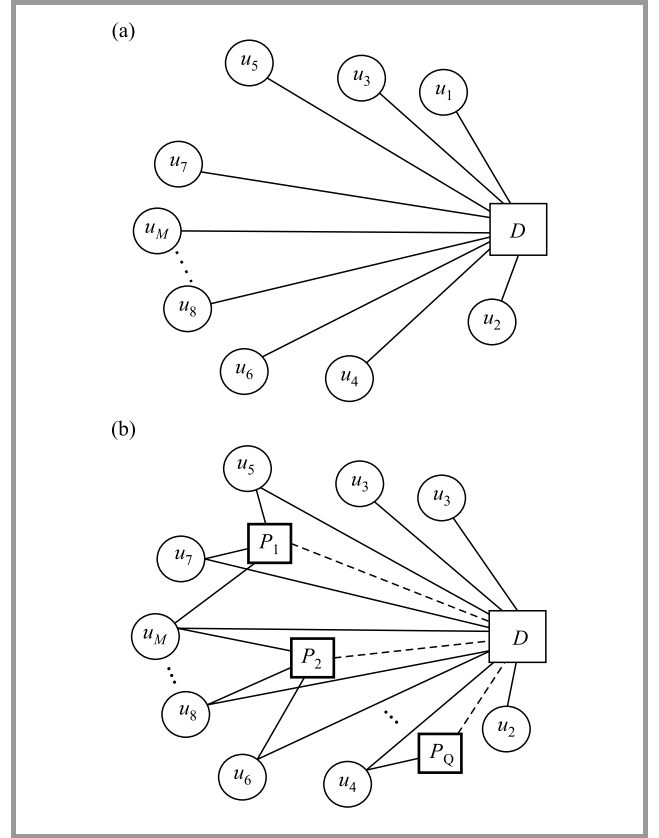


Fig. 1. The structure of wireless networks: (a) wireless networks without passive nodes, (b) wireless networks with passive nodes forming a pre-code of Raptor codes.

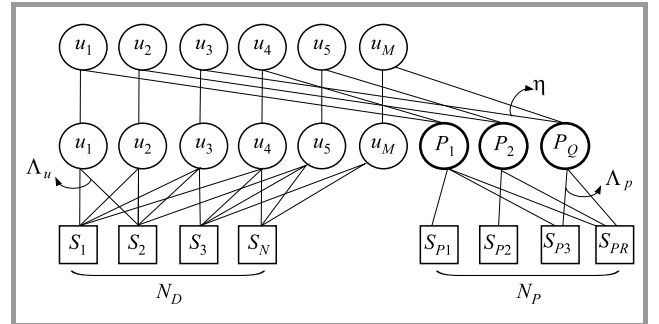


Fig. 2. Graphical representation of RAR-WN.

represents the total number of PN in RAR-WN. Each UN randomly selects a set of time slots s_n , $1 \leq n \leq N$, based on a degree distribution of repetition codes $\Lambda = \{\Lambda_u\}_{u=1}^U$, and transmits data packets via the selected time slots. D and PN receive the transmitted data packets with a different probability of collisions, because only some UNs which are connected to the PN as a pre-code of Raptor code. In PN, the data packets from connected users are decoded and forwarded to destination. The PN randomly transmits decoded data packets to D based on a degree distribution $\Lambda = \{\Lambda_p\}_{p=1}^P$ at selected time slots s_{pr} , $1 \leq r \leq R$. In this paper, we set:

$$\{\Lambda_u\}_{u=1}^U = \{\Lambda_p\}_{p=1}^P = \{\Lambda_h\}_{h=1}^H. \quad (1)$$

Degree distributions of UN and PN are defined in the set of H repetition codes denoted as $\{((n_h, k_h), \Lambda_h)\}_{h=1}^H$ and with the minimum distance of at least 2, where:

- n_h represents the number of output data packets of the h -th set of codes,
- k_h represents the number of input data packets of the h -th set of codes (for repetition codes $k_h = k = 1$),
- Λ_h represents the percentage of users that pick the h -th set of codes, where $\sum_{h=1}^H \Lambda_h = 1$ and $0 \leq \Lambda_h \leq 1$.

An example of degree distribution with $M = 4$ and $N = 4$ is illustrated as shown in Fig. 3. In this example, we assume that the first and second (u_1 and u_2) pick the $(2, 1)$ code, while the third and fourth user (u_3 and u_4) pick the code $(3, 1)$ code.

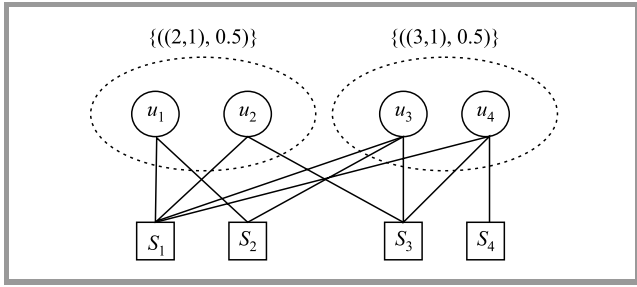


Fig. 3. An example of degree distribution with $M = 4$ and $N = 4$.

All transmissions are slot-synchronized. The channel time is divided into time slots of equal length. Channel gain of the destination, passive nodes and of all users is assumed to be perfectly known at all. Offered traffic G , as seen by each user, is defined as:

$$G = \frac{M}{(N_D + N_P)}. \quad (2)$$

3. Induced Degree Distributions

We define induced degree distributions as new distributions caused by dynamic Rayleigh fading channels, where a faded message below a threshold of P_T cannot be resolved or is considered for erasure. This phenomenon has been carefully investigated in [17] for $P_T = 0.25$, where it is found that in wireless networks, naturally the harmful degrees – degrees of $h = \{0, 1\}$, are unavoidable. However, in this paper, we assume a perfect induced degree distribution, without degrees, of $h = \{0, 1\}$, as in networks aiming to investigate the maximum performance of RAR-WN.

An example of an RAR-WN with eight users $\mathcal{U} = \{u_1, u_2, \dots, u_8\}$ and two PNs $\mathcal{P} = \{P_1, P_2\}$ is shown in Fig. 4a. The structure of RAR-WN may be seen as codes-on-the-graph, resembling the structure of Raptor codes shown in Fig. 4b. In addition, a factor graph is shown in Fig. 4c.

The degree distributions of UN, as shown in Fig. 4b and Fig. 4c, are represented by

$$\Lambda_M = \left\{ \left((1, 1), \frac{1}{8} \right) + \left((2, 1), \frac{6}{8} \right) + \left((3, 1), \frac{1}{8} \right) \right\}. \quad (3)$$

By resembling the structure of Raptor codes in [18], matrices of networks for main users ($\mathcal{G}_M$), first passive node ($\mathcal{G}_{P_1}$), and second passive node ($\mathcal{G}_{P_2}$) are shown as follows:

$$\mathcal{G}_M = \begin{pmatrix} \mathbf{1} & 1 & 1 & 0 & 0 & 0 & 1 & 0 \\ 1 & 0 & 1 & 0 & 1 & 0 & 0 & 0 \\ 0 & 1 & 1 & 1 & 0 & 1 & 0 & 1 \\ 0 & 0 & 0 & 1 & 1 & 1 & \mathbf{1} & 0 \end{pmatrix}, \quad (4)$$

$$\mathcal{G}_{P_1} = \begin{pmatrix} 1 & 1 & \mathbf{1} & 0 & \mathbf{0} & 0 & 1 & 0 \\ 1 & 0 & \mathbf{1} & 0 & \mathbf{1} & 0 & 0 & 0 \\ 0 & 1 & \mathbf{1} & 1 & \mathbf{1} & 1 & 0 & 1 \\ 0 & 0 & \mathbf{0} & 1 & \mathbf{1} & 1 & 1 & 0 \end{pmatrix}, \quad (5)$$

$$\mathcal{G}_{P_2} = \begin{pmatrix} 1 & 1 & 1 & 0 & 0 & \mathbf{0} & 1 & \mathbf{0} \\ 1 & 0 & 1 & 0 & 1 & \mathbf{0} & 0 & \mathbf{0} \\ 0 & 1 & 1 & 1 & 0 & \mathbf{0} & 0 & \mathbf{1} \\ 0 & 0 & 0 & 1 & 1 & \mathbf{0} & 1 & \mathbf{0} \end{pmatrix}. \quad (6)$$

The rows represent time slots $\mathcal{S} = \{T_1, T_2, \dots, T_4\}$, while the columns represent users $\mathcal{U} = \{u_1, u_2, \dots, u_8\}$. A loss message caused by a deep fading is represented with light grey cells in matrices $\mathcal{G}_M$, $\mathcal{G}_{P_1}$, $\mathcal{G}_{P_2}$ and the dashed lines in Fig. 4c. The connectivity of passive nodes is represented by the dark grey cells in matrices $\mathcal{G}_{P_1}$, $\mathcal{G}_{P_2}$, and the solid lines in Fig. 4c.

The number of edges connected to a node is called a node degree. The degree distribution of a user node (UN) from the node-perspective is expressed as:

$$\Lambda(x) = \sum_{h=1}^H \Lambda_h x^h. \quad (7)$$

Furthermore, the degree distribution of user node (UN) from the edge-perspective is expressed as:

$$\lambda(x) = \frac{\Lambda(x)'}{\Lambda(1)'} = \sum_{h=1}^H \lambda_h x^{h-1}, \quad (8)$$

where Λ_h represents the probability of UN having a degree h , and H represents the maximum degree of UN.

The degree distribution of user node (SN) from the node perspective is expressed in the polynomial as:

$$\Omega(x) = \sum_{d=1}^D \Omega_d x^d, \quad (9)$$

where Ω_d represents the probability of UN having a degree d and D represents the maximum degree of SN. However, since the transmission is random (and is out of control of the system designers), the degree distribution of SN in binomial distribution is expressed as:

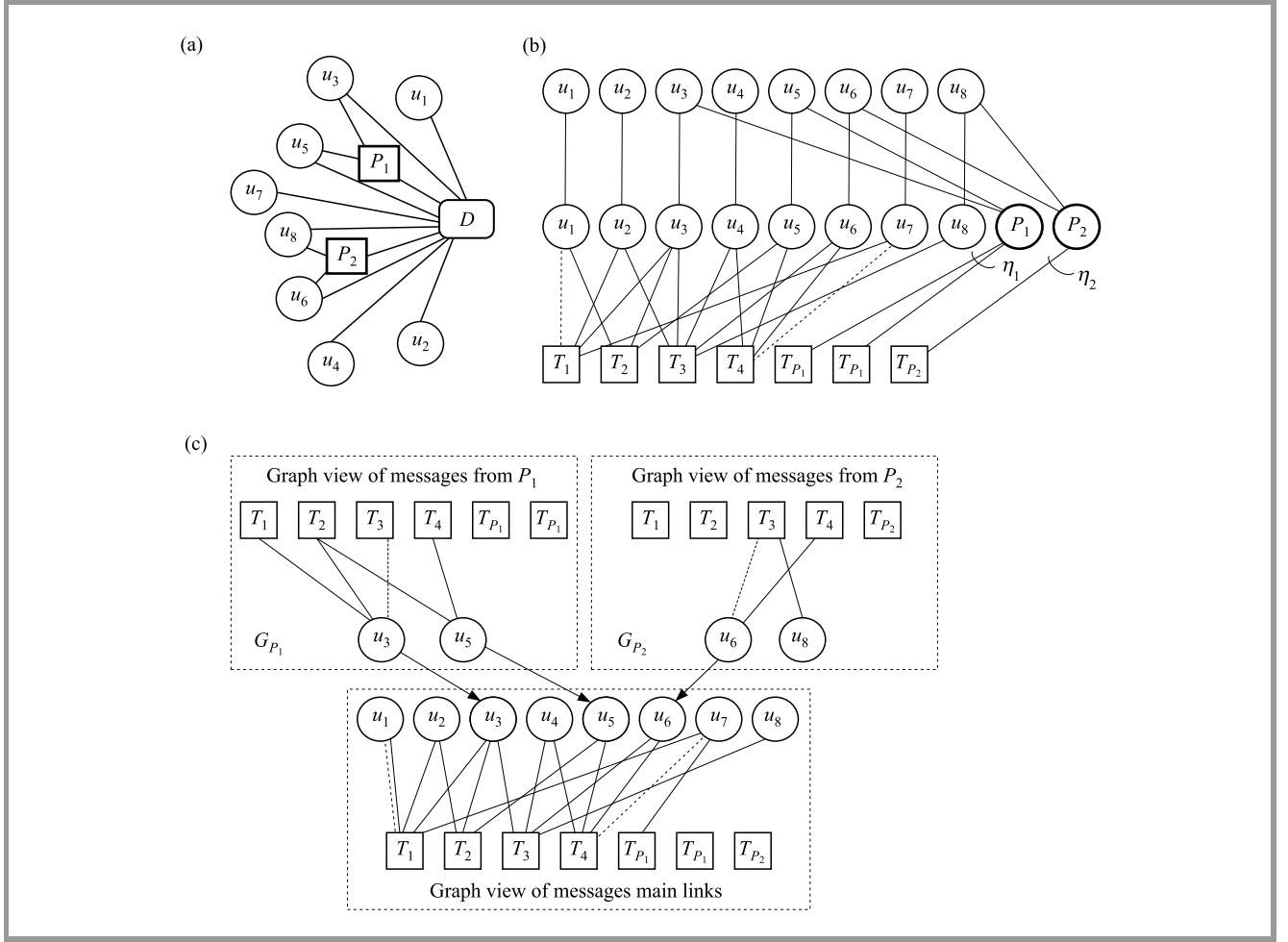


Fig. 4. Example of RAR-WN: (a) networks, (b) model, (c) factor graph.

$$\begin{aligned}\Omega(x) &= \sum_{d=1}^M \Omega_d x^d = \binom{M}{d} \left(\frac{G}{\bar{R}M} \right)^d \left(1 - \frac{G}{\bar{R}M} \right)^{M-d} x^d \\ &= \left(1 - \frac{G}{\bar{R}M} (1-x) \right)^M \approx e^{-\frac{G}{\bar{R}}(1-x)},\end{aligned}\quad (10)$$

where it is assumed that the number of users M becomes very large ($M \rightarrow \infty$) and that G remains constant. $\bar{R}$ represents the average rate of the networks:

$$\bar{R} = \frac{1}{\sum_{h=1}^H \Lambda_h x^h}. \quad (11)$$

4. Decoding Strategy and EXIT Chart

This paper considers the following degree distributions:

$$\Lambda_a = \{((2, 1), 1)\}, \quad (12)$$

$$\Lambda_b = \{((2, 1), 0.5) + ((3, 1), 0.28) + ((8, 1), 0.22)\}, \quad (13)$$

$\Lambda_a(x)$ and $\Lambda_b(x)$ are regular and irregular degree distributions, respectively. From Eq. (11), the network rate of

distribution $\Lambda_a(x)$ is $R_{\Lambda_a(x)} = \frac{1}{\Lambda_a(1)} = \frac{1}{2}$. The network rate of distribution $\Lambda_b(x)$ is $R_{\Lambda_b(x)} = \frac{1}{\Lambda_b(1)} = 0.278$. The highest asymptotic threshold $G^* = 0.932$ packets/slot is achieved even though the practical traffic is $G = 0.83$ packets/slot at PLR of 10^{-1} , which is quite high for a loss rate. The regular degree distribution is expected to be simple in practice.

This paper analyses the property of the degree distributions and their achievable offered traffic G using the EXIT chart. The probability of an edge coming out from SN is defined as p . The probability of an edge coming out from UN is defined as q . Since the number of passive nodes is denoted as Q and $(Q+1)$ graphs are involved, then internal and external erasure probabilities are defined as p^{int}, q^{int} , and p^{ext}, q^{ext} .

We consider the repetition codes as a type of practical coding schemes.

4.1. $(n_h, 1)$ Repetition Codes

A repetition code is the simplest coding scheme that is as simple as its analysis. For RAR-WN, the internal erasure probability is:

$$q^{int} = \frac{1}{\bar{n}} \sum_{h=2}^H \Lambda_h (p^{int})^{h-1} p^{ext} = \sum_{h=2}^H \lambda_h (p^{int})^{h-1} p^{ext}, \quad (14)$$

meaning that it is lower compared to that of wireless networks without PN case, because of the factor of p^{ext} . Here, λ_h is the degree of UNs from the edge perspective [19], [20]. For external q^{ext} , Fig. 4c shows that an additional edge $\mathcal{E}^e$ exists due to the connection of PN and, therefore, the external erasure probability is becoming:

$$q^{ext} = \sum_{h=2}^H \lambda_h (p^{int})^h, \quad (15)$$

which does not exist in the case of conventional wireless networks.

4.2. Decoding Algorithm

The decoding steps of PN and D are shown in Algorithm 1 and Algorithm 2, respectively.

Algorithm 1: Decoding of passive nodes

Data: Connected bipartite graphs $\mathcal{G}_{P_i}$

Result: Messages ηM are decoded.

```

for  $j=1$  until  $Q$  do
  Loop  $\mathcal{G}_{P_j}$  : Consider  $\mathcal{G}_{P_j}$  and find user  $u_i$ 
    connected to a slot node having degree  $d$ ;
  while  $\text{degree } d = 1$  is found do
    Decode message of  $u_i$ ;
    Perform SIC in all slot nodes connected to
    user  $u_i$  at graph  $\mathcal{G}_{P_j}$ ;
    Send message of  $u_i$  via  $s_{P_j}$  and subtract the
    received signals in all slot nodes connected to
    user  $u_i$  at graph  $\mathcal{G}_M$ ;
  end
end

```

Algorithm 2: Decoding of destination

Data: Connected bipartite graphs $\mathcal{G}_M$

Result: Messages M are decoded.

```

Loop  $\mathcal{G}_M$  : Consider  $\mathcal{G}_M$  and find user  $u_i$  connected to
  a slot node having degree  $d$ ;
while  $\text{degree } d = 1$  is found do
  Decode message of  $u_i$ ;
  Subtract the received signals in all slot nodes
  connected to user  $u_i$  with user  $u_i$ 's own message;
end

```

Several nodes failed during the contention period are resolved via passive nodes although the networks are overloaded.

4.3. EXIT Chart Analysis

In this paper, the equivalent parameter is defined for UN, instead of using the mutual information (MI) as used in

EXIT chart analysis. The priori and extrinsic information for the user can be expressed as Eqs. (16) and (17), respectively:

$$I_{\text{User}}^A = 1 - p^{int}, \quad (16)$$

$$I_{\text{User}}^E = 1 - q^{int}. \quad (17)$$

A similar definition is applied for SN as:

$$I_{\text{Slot}}^A = 1 - q^{int}, \quad (18)$$

$$I_{\text{Slot}}^E = 1 - p^{int}, \quad (19)$$

The decoding trajectory is used for two purposes: to confirm the accuracy of EXIT analysis, and to obtain the asymptotic maximum offered traffic G^* , where G^* is defined as the maximum G when EXIT curves of both SN and UN do not intersect each other. The EXIT curves and the corresponding decoding trajectories for distributions $\Lambda_a(x)$ and $\Lambda_b(x)$ are shown in Figs. 5 and 6, respectively.

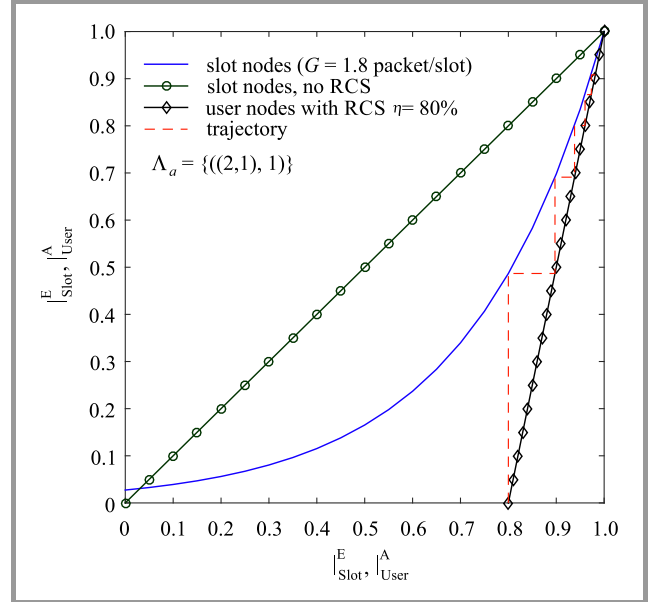


Fig. 5. EXIT projection and decoding trajectories for graphs with Λ_a at $G = 1.8$ packets/slot.

The projected EXIT curves of the UN are shown in dash-dotted lines below the EXIT curves of SN. Figure 4 shows an excellent agreement between the trajectory and EXIT curves of both SN and UN. Loss of distribution becomes higher if the gap between the UN and the SN curves is increased. A smaller gap with $\Lambda_a(x)$ is preferable to provide a higher throughput of G , as shown in Fig. 4a. Figure 4b shows a bigger gap with $\Lambda_b(x)$ resulting in smaller traffic $G = 1.8$ p/s. In addition to the indication of using EXIT curves, the trajectories also confirm a tunnel between the EXIT curves. It opens at $G = 1.8$ p/s and $G = 1.8$ p/s for $\Lambda_a(x)$ and $\Lambda_b(x)$, respectively, but the trajectories intersect for G exceeding those values.

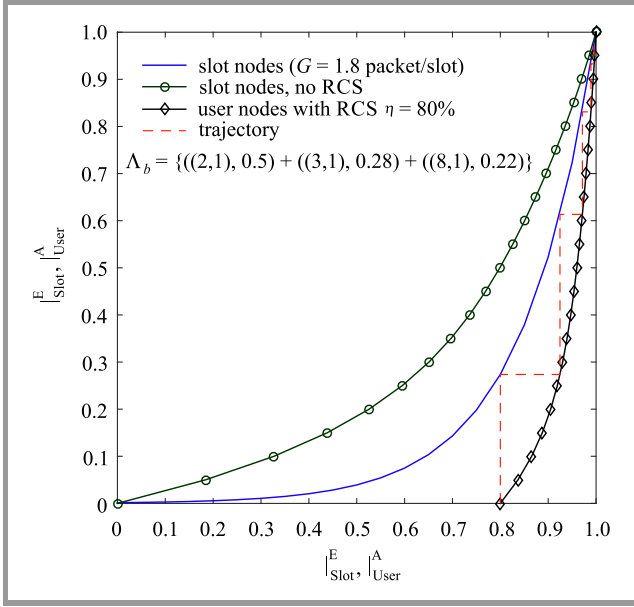


Fig. 6. EXIT projection and decoding trajectories for graphs with Λ_b at $G = 1.8$ packets/slot.

5. Performance Evaluations

This section evaluates PLR and throughput performance using a series of computer simulations with TS length N , having $N \rightarrow \infty$ ¹. The simulations are conducted under block Rayleigh fading channels to exploit the nature of broadcast messages received by destination D and PNs. PLR performance of the decoding scheme is evaluated based on a simulation.

5.1. PLR and Stopping Sets

A series of computer simulations with $N = 1000$ time slots was performed to evaluate practical PLR performance. The results are plotted in Fig. 7 for degree distributions $\Lambda_a(x)$ and $\Lambda_b(x)$. The figure shows that PLR of RAR-WN is always worse at $G < 1$, but is better at $G > 1$. The $\Lambda_a(x)$ has a faster waterfall (closer to the theoretical limit) than $\Lambda_b(x)$. Figure 7 shows PLR performance under dynamic block Rayleigh fading channels. It is found that regular degree distribution provides better PLR than irregular degree distribution, as indicated by RAR-WN Λ_a and RAR-WN Λ_b .

Among the possible stopping sets, we proved that not all stopping sets have a significant contribution to the error floor. In Table 1 six most significant stopping sets, based on their occurrence probability, for various frames N are shown, where $P_o(\cdot)$ is the occurrence probability of its argument in one graph [21]. These stopping sets limit PLR performance of RAR-WN which should be significantly reduced by the use of PNs. It is important to note that the stopping sets in RAR-WN occur only if PNs are experiencing the same stopping sets expressed by the factor $\mathcal{X}(\cdot)$ as shown in Table 1.

¹As a consequence of $M \rightarrow \infty$ and G constant is that $N \rightarrow \infty$.

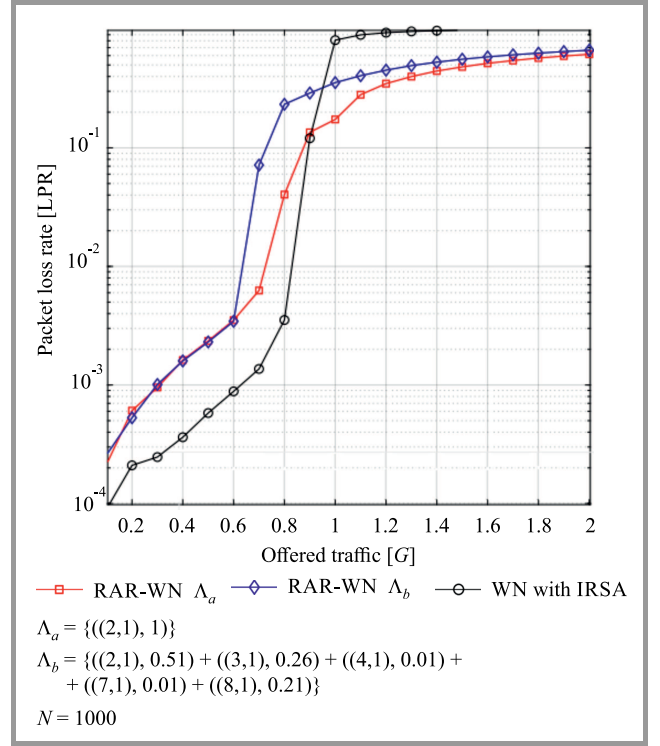


Fig. 7. PLR performances for RAR-WN (10%, 8).

Table 1
Six most important stopping sets

Set	Graph	$P_o(\cdot)$	$\mathcal{X}(\cdot)$
$\mathcal{S}_1$		$\frac{M!}{(M-2)!} \frac{\Lambda_2^2}{2!} \frac{2}{(N-1)N}$	$\binom{M}{2} \Lambda_2^2$
$\mathcal{S}_2$		$\frac{M!}{(M-3)!} \frac{\Lambda_3^3}{3!} \frac{8(N-2)}{(N-1)^2 N^2}$	$\binom{M}{3} \Lambda_2^3$
$\mathcal{S}_3$		$\frac{M!}{(M-4)!} \frac{\Lambda_4^4}{4!} \frac{288(N-3)(N-2)}{3(N-1)^3 N^3}$	$\binom{M}{4} \Lambda_2^4$
$\mathcal{S}_4$		$\frac{M!}{(M-3)!} \frac{\Lambda_2^2 \Lambda_3}{2!} \frac{24}{(N-1)^2 N^2}$	$\binom{M}{3} \frac{\Lambda_2^2 \Lambda_3}{2}$
$\mathcal{S}_5$		$\frac{M!}{(M-3)!} \frac{\Lambda_2 \Lambda_3^2}{2!} \frac{36(N-3)}{(N-2)(N-1)^2 N^2}$	$\binom{M}{3} \frac{\Lambda_2 \Lambda_3^2}{2}$
$\mathcal{S}_6$		$\frac{M!}{(M-4)!} \frac{\Lambda_2^3 \Lambda_3}{3!} \frac{288(N-3)}{(N-1)^3 N^3}$	$\binom{M}{4} \frac{\Lambda_2^3 \Lambda_3}{6}$

The occurrence probability of stopping sets for conventional networks is:

$$\dot{p}(\mathcal{S}_i) = P_o(\mathcal{S}_i), \quad (20)$$

$i \in \{1, 2, \dots, 8\}$, while for RAR-WN:

$$\ddot{p}(\mathcal{S}_i) = P_o(\mathcal{S}_i)^2 \cdot \frac{1}{\mathcal{X}(\mathcal{S}_i)}, \quad (21)$$

$i \in \{1, 2, \dots, 8\}$, where the terms $\frac{1}{\mathcal{X}(\mathcal{S}_i)}$ expresses the probability that both graphs experience the same stopping sets with $\mathcal{X}(\mathcal{S}_i)$ is obtained from the following multinomial probability:

$$\mathcal{X}(\mathcal{S}_i) = \binom{M}{v(\mathcal{S}_i)} v(\mathcal{S}_i)! \prod_{h=1}^H \frac{\Lambda_h^{v_h(\mathcal{S}_i)}}{v_h(\mathcal{S}_i)!}. \quad (22)$$

However, with regular distribution, $\mathcal{X}(\mathcal{S}_i)$ may be simplified into $\mathcal{X}(\mathcal{S}_i) = \binom{M}{v(\mathcal{S}_i)}$. PLR for conventional networks may be approximated by:

$$\dot{P}_L \approx \frac{\dot{\rho} \{2(\mathcal{S}_1)+3(\mathcal{S}_2)+4(\mathcal{S}_3)+2(\mathcal{S}_4)+(\mathcal{S}_5)+3(\mathcal{S}_6)\}}{M\Lambda_2} + \frac{\dot{\rho} \{(\mathcal{S}_4)+2(\mathcal{S}_5)+(\mathcal{S}_6)\}}{M\Lambda_3}. \quad (23)$$

Similarly, PLR for RAR-WN may be approximated by:

$$\dot{P}_L \approx \frac{\dot{\rho} \{2(\mathcal{S}_1)+3(\mathcal{S}_2)+4(\mathcal{S}_3)+2(\mathcal{S}_4)+(\mathcal{S}_5)+3(\mathcal{S}_6)\}}{M\Lambda_2} + \frac{\dot{\rho} \{(\mathcal{S}_4)+2(\mathcal{S}_5)+(\mathcal{S}_6)\}}{M\Lambda_3}, \quad (24)$$

which is definitely lower than $\dot{P}_L$ of conventional networks due to the terms $\mathcal{X}(\mathcal{S}_i)$ in Eq. (21).

The results of simulations is plotted in Fig. 7. It indicates that the stopping sets shown in Table 1 is good enough to predict the error-floor of the networks, while the traffic G indicated by the waterfall curve can be predicted using EXIT chart as the upper bound, since EXIT chart analysis assume that the frame-length N is very large.

5.2. Throughput Analysis

The considered throughput of the networks is shown in Fig. 8. Surprisingly, throughput performance of RAR-WN

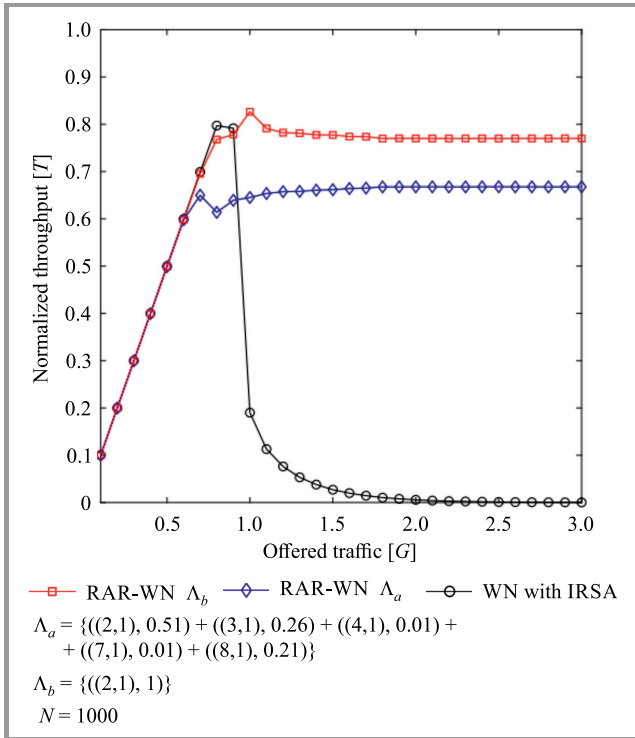


Fig. 8. Throughput performances for RAR(10%, 8).

is more stable, although the offered traffic is beyond 1 p/s. IRSA may improve throughput with the help of passive nodes in the Raptor code-structured parameter with RAR (10%, 8). RAR-WN with regular degree distribution has better throughput than RAR-WN with irregular degree distribution.

6. Conclusions

In this paper, random access-based passive nodes with regular and irregular degree distributions are developed in RAR-WN. The results show that the destination is capable of decoding more information from users by using RAR-WN. It is proved with the cross-line between SN and UN lines at I_{slot}^A and I_{user}^E close to one. Regular degree is outperform irregular degree distribution close to PLR when traffic is overloaded. RAR-WN maintains achievable throughput when wireless networks are overloaded.

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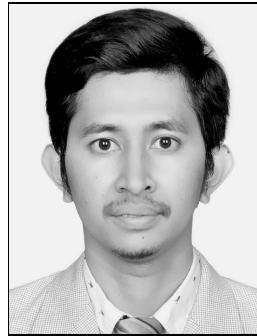
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Simulations of the MAC Layer in the LoRaWAN Networks

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Abstract—The Internet of Things is changing the approach to data transmission, protocol design and network services. The challenge faced by designers of IoT solutions is to determine the scalability of a given technology, with a particular emphasis placed on unlicensed frequency bandwidth (ISM) transmission in highly urbanized areas. Because the design and implementation of a wireless network for the Internet of Things, relying on each of the presented technologies, is expensive and time consuming, it must be preceded by a performance assessment based on computer simulations. The literature contains various approaches to modeling the mechanisms of the MAC layer of LoRa technology and to its implementation in LoRaWAN networks. The article provides an overview of major LoRa MAC network simulators. It presents and comments on the most important research results obtained by the authors of the aforementioned software.

Keywords—*Internet of Things, LoRa, LoRaWAN, LPWAN, simulations.*

1. Introduction

The fundamental requirements applicable to Internet of Things (IoT) data transmission technologies include low power consumption, low cost and low complexity of end nodes capable of transmitting over long distances. With such assumptions kept in mind, the end devices may be battery operated or may operate autonomously, with the support of energy harvesting. The communication range covers distances from hundreds of meters to several kilometers, and the network is based on the star topology. Thus, routing-related problems in this type of network are not significant and are not considered here.

Among the solutions dedicated to the IoT, three competing categories are mentioned, differing in range, cost and throughput, that will dominate these networks. They include the following:

- Low rate wireless personal area network (LR-WPAN) technologies that are relied upon to create short range networks interconnecting devices owned by an individual or constituting a building network. These networks provide low data rates and short range communications in order to focus on low energy consumption and saving battery capacity. As far as channel access is concerned, IEEE 802.15.4

uses the carrier sense multiple access with collision avoidance (CSMA/CA) technology. This multiplexing approach allows multiple nodes access the same channel at different times, without any interference. Most transmissions consist of short packets and occur with a very low duty cycle (less than 1%), minimizing power consumption. The defined minimum TX power level is -3 dBm or 0.5 mW. Most modules use 0 dBm (1 mW).

- NB-IoT and LTE-M standards operate in licensed bands and leverage the already existing cellular network coverage to provide access to IoT nodes. New standards will play a key role in the development of 5G networks which serve as a foundation stone for mass-scale implementation of IoT solutions [1].
- Low power wide area networks (LPWAN) which provide wireless connections using the star topology and long distance transmission in the unlicensed frequency bands below 1 GHz. Sigfox uses the ultra narrow band (UNB) radio technology at 868 MHz in Europe and in the Middle East, and 902 MHz in North America. Thanks to its bandwidth of 192 kHz, it allows to transmit more signals simultaneously. It is possible to communicate over long distances while displaying high noise immunity levels. The potential of losing a packet is minimized by sending three copies of the data packet at random frequencies. A Sigfox frame carries 12 bytes of data, and together with the data needed to control the transmission, it is no more than 26 bytes in size. The LoRaWAN technology, in turn, is becoming a leader of LPWAN range of solutions due to ultra-low power consumption, operation in the unlicensed band (similarly to Sigfox) and exploits a new spread spectrum physical layer design that enables higher receiver sensitivity and better coverage. The past three years have seen the research community placing an increasing emphasis on LoRaWAN. Based on data presented at The Things Conference, the LoRaWAN IoT ecosystem is the largest, with over 500 companies and over 3,000 members deploying that particular solution.

It turns out that the LoRaWAN technology enjoys – due to its low radio emissions, long range and, most importantly,

wide support from the global community – the greatest popularity among both IoT enthusiasts and scientific communities [2].

The design and implementation of a wireless network for IoT, relying on each of the aforementioned technologies, is expensive and time consuming. Therefore, it must be preceded by a performance assessment using computer simulations. Mathematical models are required to assess the impact of specific network parameters, to assess the performance of the solution and, consequently, its usability.

The article is divided into five sections. Section 1 discusses the IoT solutions, while Section 2 describes basics of the LoRaWAN technology. Section 3 covers performance and restrictions of the MAC layer. Section 4 reviews application of the LoRaWAN network simulation at the MAC layer, and presents the simulation results. Section 5 summarizes the work performed and presents the conclusions drawn.

2. LoRaWAN Technology Overview

LoRaWAN is a network standard proposed by the LoRa Alliance [3], using proprietary LoRa modulation (based on the chirp spread spectrum modulation technique) – developed and owned by Semtech Company [4]. The main modulation parameter is the *spreading factor* (SF) that affects the data rate and the range of radio transmission (the value of this parameter varies from 7 to 12). Signals modulated with different SF parameters are orthogonal, so they may be decoded when they are transmitted at the same time and on the same frequency [5]. Moreover, it is possible to decode signals using the same SF when the difference in their power levels is greater than 6 dB (4 mW) [4].

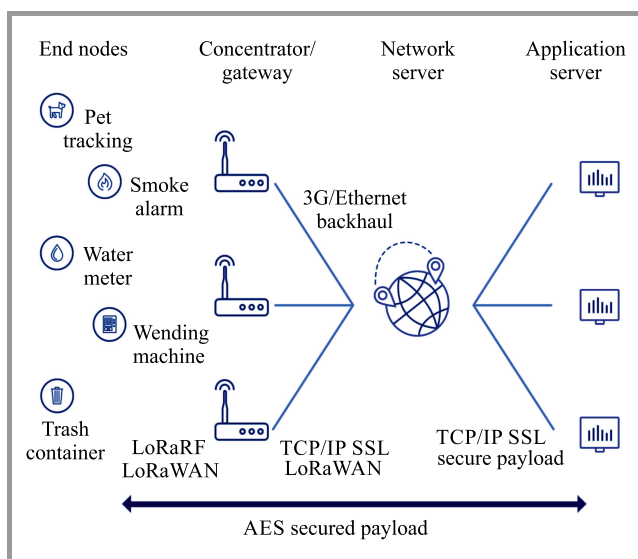


Fig. 1. Architecture of LoRaWAN system.

At the network access layer (MAC), LoRaWAN architecture defines an open protocol standardized by the LoRa

Alliance. The LoRaWAN network architecture relies on a star topology, in which the end nodes (e.g. sensors, systems measuring physical values) communicate only with a finite number of LoRaWAN gateways in a given area and do not communicate with each other (Fig. 1). Therefore, the gateways perform the function of so-called packet relays in communicating with the network server, by encapsulating raw data into IP packets using TCP or UDP protocols. The network server also allows to send downlink packets towards end nodes. The details of this transmission are based on the class of the end device – the standard defines three classes of end devices: A, B and C.

In order to save energy, Class A devices remain in the sleep mode for most of the time, opening two reception windows (RX) only 1 and 2 seconds after the end of the packet transmission (from the end device to the gateway). This mechanism allows the end devices to send acknowledgment packets (the ACK message is sent in one of the two windows after the network server has received the message correctly). For transmitting in the first window, the same frequency channel is used. The transmission relying on the second window takes place via the 869.525 MHz channel, with a spreading factor of 12 and transmission power increased to 24 dBm (250 mW).

To increase the system's transmission capabilities toward the end nodes, Class B devices open reception windows at specific, scheduled times. The gateway transmits a beacon downlink to the Class B end devices in order to synchronize and let the web server know when a specific end device will listen for downlink traffic. Class C devices open reception windows continuously, being available for downlink traffic all the time (except for the time during which they are transmitting).

The LoRaWAN protocol relies on mechanisms that ensure reliable and secure communication. The adaptive data rate (ADR) mechanism allows to dynamically manage link parameters to increase the packet delivery rate. Transmission parameters may be managed at both end device and network server sides. According to the standard's documentation [6], the end device first tries to tune connectivity increasing its transmit power. If this turns out to be insufficient, it will continue to decrease the data rate.

3. LoRa MAC Performance

LoRa uses the ALOHA technique to access the transmission channel. This is one of the simplest and least effective access methods. In a pure ALOHA, the probability of collision is proportional to the number of users of the common transmission channel, but LoRa is characterized by a variable packet length – this increases the collision rate [7]. Therefore, many manufacturers implement the listen-before-talk (LBT) mechanism in addition to the standard solution. LBT means that the end nodes enter the receiving mode and check the interference signal level before starting the transmission (Murata modem firmware enables LBT while selecting Japan and Korean LoRa frequency

ranges) [8]. The impact of the LBT mechanism is also the subject of simulation studies [9].

The important limitations affecting scalability of LoRaWAN networks include interference caused by other wireless technologies operating in the same ISM band, and self-interference. In Europe, LoRaWAN operates on 868 MHz band channels. Similar technologies dedicated to the IoT, such as Sigfox, IEEE 802.15.4g or Z-Wave, also operate in this band. Therefore, they affect the MAC layer of LoRa. Such cross-technology interference (CTI) is the subject of research as well [10], [11].

LoRa transmission at the MAC layer level is also exposed to interference from end nodes and downlink transmission from LoRaWAN gateways, caused by:

- packet collisions that occurs on the same frequency channel and due to the same spreading factor SF,
- collisions of packets sent downlink to the end device in the first transmission window (RX1) and packets sent to the gateway by other devices on the same channel,
- half-duplex communication between the LoRaWAN gateway and the end node (collision between the message transmitted from the end node during the transmission of the message from the gateway to the node) when the gateway is transmitting data or is just starting the transmission,
- unavailability of RX1 and RX2 reception windows due to other transmissions or duty cycle restrictions.

In addition, the following standard-related restrictions should be taken into account:

- ETSI regulations for the 863–870 MHz ISM band define the limits of the maximum duty cycle value at 0.1% or 1%, depending on the selected sub-band, which limits the bandwidth of devices and the total network capacity – the number of devices in a given area may be large, but they are limited to a few bytes of data per day,
- in computer modeling, in order to determine the radio range of wireless network nodes, it is important to choose the propagation model and to select radio transmission parameters; it was shown in [12], [13] that the packet loss rate does not exceed 10% when the range is less than 2 km and when SF is 9–12, and that it increases to above 60% when the range exceeds 3.4 km and SF is 12.

4. Network Models and Simulators

Network performance is difficult to evaluate based on measurements performed in real networks. The number of parameters and potential scenarios precludes the construction of a real-world, large network. Therefore, models of

self-interference, as well as of techniques used for accessing the common transmission channel and for controlling data transmission (e.g. ACK, JOIN) enable researchers to evaluate the performance of LoRa- and LoRaWAN-based operations.

A review of the literature shows that most articles presenting LoRa network simulations analyze access to a common channel only for traffic generated by the end nodes. In [7], the authors included specific assumptions of a model simulating the MAC sublayer of a LoRa network in the *LoRaSim* application, written in Python, using the *SimPy* framework. However, some simplifications were made, e.g. the effect of imperfect orthogonality of messages generated in the same channel with different SF was omitted. The main metric used in the simulation to evaluate the quality of the system is called data extraction rate (DER), and is determined as the ratio between the amount of received messages and transmitted messages over a specific period of time.

Figure 2 (presented in [7]) shows the dependence of the abovementioned parameter on three sets of transmitting node parameters for different SF values. For the SN3 set, all nodes share the same SF. For SN⁴, the parameters are selected to minimize the airtime at a constant transmission power of 14 dBm (25 mW), while for SN⁵ airtime is minimized, followed by a transmission power that still allows the end node to successfully reach the gateway. Based on this configuration, it may be observed that the DER parameter drops below 50% for a gateway serving more than 900 devices. In turn, assuming that the ratio of correctly received packets (DER) equals > 90%, the number of serviceable end nodes is 1600.

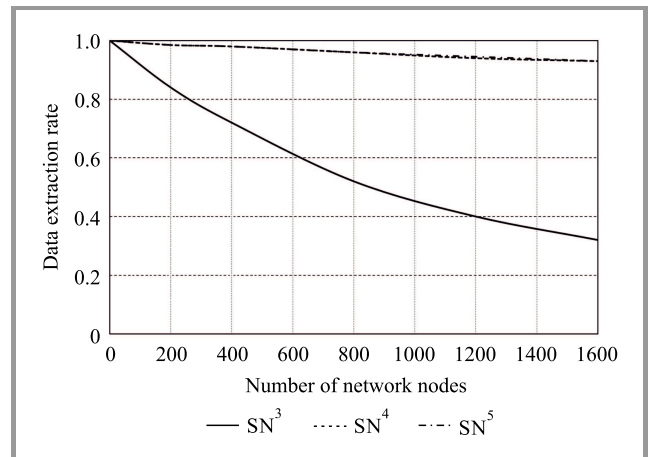


Fig. 2. Data extraction rate (DER) versus number of nodes for three transmitter parameters sets [7].

LoRaWANSim [14] extends the LoRaSim simulator by supporting MAC layer mechanisms of the LoRaWAN protocol which supports bidirectional communication. It implements a more complicated LoRaWAN MAC protocol instead of the features of the ALOHA protocol used in the LoRaSim simulator. Downlink transmission to the end node is crucial in IoT applications due to handshakes, acknowledgement traffic (i.e. ACK, LinkCheck, Duty Cycle,

RXParam-Setup), and key exchange in cryptographic algorithms, and has been taken into account.

LoRaWANSim provides a 1% duty cycle for most European sub-bands. It also implements a realistic collision model in which collisions for uplink and downlink traffic do not occur even when the transmission is carried out at the same time and on the same channel and the same SF. The end device will only receive the packet sent by the gateway, even if it experiences a signal with a higher received signal strength indicator (RSSI) from its neighboring nodes, if it is part of a group of end nodes far away from the gateway. This simulator considers the retransmission strategy as well. Retransmission occurs when the reception of a packet is not confirmed (e.g. due to a collision or duty cycle limitations). In that case, the LoRaWAN specification recommends to transmit it up to 8 times. If 8 consecutive attempts fail, the application should be notified. Moreover, the specification recommends that the end nodes limit the transmission data rate every two unsuccessful transmission attempts, in order to increase the robustness of uplink connectivity.

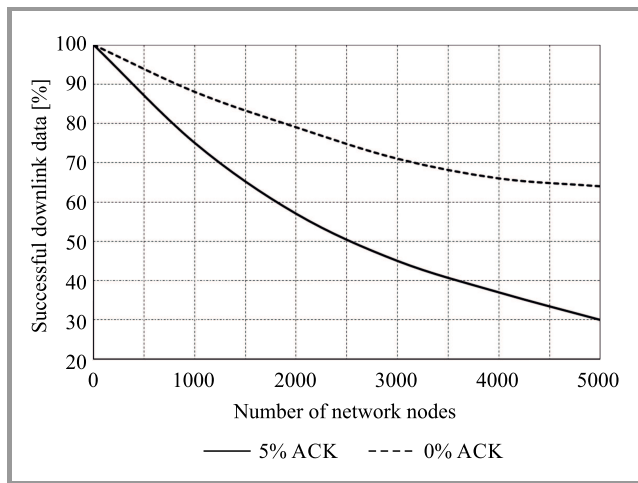


Fig. 3. Impact of network size on successful downlink data transmission [14].

The main conclusion of the research conducted in [14] is an unreliable downlink transmission caused by gateway duty cycle limitations and collisions. Figure 3 (presented in [14]) shows the percentage of successful downstream transmissions with and without confirmation (ACK). These conclusions may be of particular importance when considering the implementation of LoRaWAN protocols in upper layers that require two-way communications (e.g. IPv6 stack).

Data rate reduction experienced when no further ACK messages have been received is an important problem that surfaces in this study. Such a strategy is correct when losses at higher data rates are associated with poor link quality and are not the result of exhaustion of the duty cycle limit at the gateway. Otherwise, reduction in the data rate is unreasonable, because by increasing the air time of the ACK message, the probability of collision is increased as well. Therefore, the recommended number of retransmission attempts in the LoRaWAN specification may not be appro-

priate for all scenarios and largely depends on the network size. For small networks with less than 600 nodes, 90% of messages are confirmed immediately, while for networks of up to 1000 nodes, 95% of messages are confirmed after two retransmissions.

The increase in energy consumption is another important phenomenon associated with retransmissions. Figure 4 (presented in [14]) shows the increase in battery drainage for data transmission with retransmissions, for networks with 1000 and 5000 nodes, respectively.

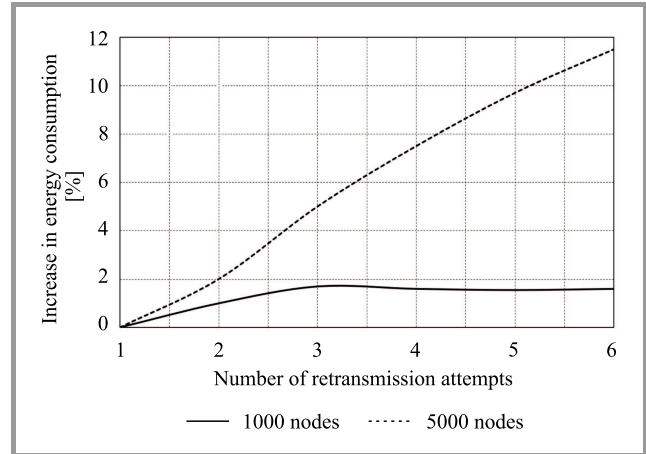


Fig. 4. Energy consumption versus the number of transmission attempts [14].

Indicating the high collision rate and the adverse impact of retransmission in the simulators presented earlier, the *LoRaFREE* simulator which uses synchronized bulk data transmission was proposed [15]. It also uses *Simpy* and the long-distance path loss model of *LoRaSim* [7]. This solution assumes that the data is buffered before transmission and sent in scheduled time slots. It includes, in LoRaWAN packets sent toward the end devices, basic transmission parameters, such as frequency channels, transmitter powers, spreading factors and time slots. In addition, longer packages were used, which reduced the overhead of MAC headers. Hence, scalability problems were overcome – the number of collisions and group notifications was minimized. The authors showed that, due to such an approach, the operating time of the battery-powered devices was increased more than five-fold, with the data delivery ratio exceeding 99%.

The data collection process takes place periodically at intervals that are known to the gateway and end devices. In order to schedule each data collection period, the gateway must know the number of end devices that store the data to be sent, the amount of buffered data, and estimated path loss. Then, the gateway broadcasts the schedule and synchronizes all end devices with the same time reference.

In *LoRaFREE*, retransmissions are scheduled in such a way that devices with the same spreading factor transmit sequentially, and devices with different spreading factors transmit simultaneously. The transmission was organized into six

parallel frames that correspond to six SFs (7–12). Each frame consists of a specific number of uplink slots and one downlink slot at the end of the frame. The end nodes send join-request messages to the gateway, containing the size of the data to be sent and the RSSI parameter of the message. Based on this data, the gateway determines the SF that the device should use in the scheduled transmission that takes place in the slot of the above-mentioned frame.

Due to the fact that transmissions with different SFs are not completely orthogonal and cause mutual interferences, it is necessary to determine the threshold difference between the strength of the reference transmission and the interfering transmissions at which both signals may be correctly received. For this purpose, the results of experimental studies presented in [5] were used, and the interference thresholds among the spreading factors are presented in Table 1.

Table 1
Interference thresholds [dBm]

SF_{ref}	SF_{int}					
	7	8	9	10	11	12
7	1	-8	-9	-9	-9	-9
8	-11	1	-11	-12	-13	-13
9	-15	-13	1	-13	-14	-15
10	-19	-18	-17	1	-17	-18
11	-22	-22	-21	-20	1	-20
12	-25	-25	-25	-24	-23	1

The authors consider two scenarios for scheduling data transmissions in the proposed simulator (when the algorithm allocates the spreading factors so as to minimize data collection time and when the energy minimization objective is considered) and compare the results obtained with two other LoRaWAN implementations, named *Legacy LoRaWAN* and *Delayed LoRaWAN*, respectively. The *Legacy LoRaWAN* uses the standard LoRaWAN MAC, where devices comply with the Class A specification (ALOHA-type

MAC, two reception windows after each uplink and immediate transmission of the data). In the *Delayed LoRaWAN*, end devices buffer the data and send it in bulk, at scheduled times. The devices also comply with Class A specification and do not perform any synchronization before the transmission. A time offset is only introduced to synchronize the devices in order to minimize collisions. The end devices send long packets to reduce the overhead of MAC headers. Then, traffic was simulated where the transmissions to the gateway are not acknowledged. The application generates 20-byte packets with a Poisson distribution at the rate of 5 minutes.

Figure 5 (taken from [15]) shows that achievement of low power consumption and reduction in the duration of the transmission takes place at the expense of the data delivery ratio (DDR) which decreases significantly along with the increase in the number of network nodes for the Delay LoRaWAN model (DDR represents the ratio between correctly received packets and the initial buffer sizes of all end devices). It would seem that the pure ALOHA mechanism (Legacy LoRaWAN) generates more collisions and a higher packet loss, but the large packet size in the Delayed LoRaWAN model results in lower DDR.

5. Conclusions and Future Works

A literature review and research using existing simulators shows that it is possible to model the majority of MAC layer mechanisms in a LoRaWAN network using free programming environments. The simulators presented in this paper introduce further extensions of the MAC sublayer compared to the basic implementation [7]. They are written in Python and their source codes have been published. Therefore, there is no need to create complex real-world networks and advanced testbeds. Research conducted using these tools provides answers to questions concerning the error rate and network scalability limits. New mechanisms that will be implemented in subsequent versions of the specification (e.g. firmware update over the air, FUOTA) [16] basically increase the number of downlink packets and degrade the quality of transmission even further. The firmware update over the air is not part of the MAC layer but is performed at the application layer.

From the point of view of the development of new public and private LoRaWAN networks, restrictions on the maximum number of nodes and collision limitation mechanisms are important. These problems are modeled by the simulators described in this article. However, LoRaWAN network topology modeling methods are equally important. Algorithms supporting the placement of gateways in combination with radio range modeling in urban environments are main challenges faced by computer simulation environments.

The authors' participation in the research agenda aimed at sending data from electricity meters will allow to compare the simulation test outcomes with results obtained in real measurement networks using LoRaWAN technology.

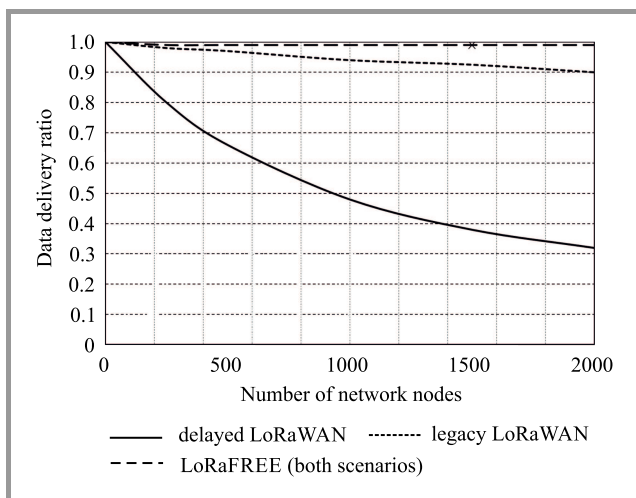


Fig. 5. Impact of network size on data delivery ratio [15].

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Estimation of UPMC Fading Channels Using H_∞ Filter

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Abstract—Universal filtered multi-carrier (UPMC) modulation is a very powerful candidate to be employed for future 5G mobile systems. It overcomes the limitations and restrictions in current modulation techniques employed in 4G mobile systems and supports future applications, such as machine-to-machine (M2M), device-to-device (D2D), and vehicle-to-vehicle (V2V) communications. In this paper, we address the estimation of UPMC fading channels based on the comb-type pilot arrangement in the frequency domain. The basic solution is to estimate the fading channel based on the mean square error (MSE) or least square (LS) criteria with adaptive implementation using least mean square (LMS) or recursive least square (RLS) algorithms. However, these adaptive filters seem not to be effective, as they cannot fully exploit fading channel statistics, particularly at high Doppler rates. To take advantage of these statistics, time-variations of the fading channel are modeled by an autoregressive process (AR), and are tracked by an H_∞ filter. This, however, requires that AR model parameters be known, which are estimated by solving the Yule-Walker equation (YWE), based on the Bessel autocorrelation function (ACF) of the fading channel with a known Doppler rate. Results of Matlab simulations show that the proposed H_∞ filter-based channel estimator is more effective when compared with existing estimators.

Keywords—5G, autoregressive model, channel estimation, fading channel, H_∞ filter, Kalman filter, LMS filter, RLS filter, UPMC.

1. Introduction

Orthogonal frequency division multiplexing (OFDM) is the current multiple access technique employed in modern communication systems, such as wireless local area networks (WLAN), power line communication (PLC), asymmetric digital subscriber lines (ADSL), digital audio broadcasting (DAB), and digital video broadcasting (DVB) [1], [2]. Simple transceiver design is a significant advantage of OFDM. Indeed, it uses the inverse fast Fourier transform (IFFT) at the transmitter and the fast Fourier transform (FFT) at the receiver [3], [4]. Nevertheless, it suffers from numerous shortcomings and limitations, such as the overhead of inserting the cyclic prefix (CP), high sensitivity to carrier frequency offset (CFO), and high peak average power ratio (PAPR) [5], [6]. To overcome these drawbacks, filter bank multi-carrier (FBMC) modulation is suggested and compared with OFDM in [3]. The design

of the FBMC system is similar to that of OFDM, as it is based on IFFT/FFT pair's process. In contrast, it uses groups of digital filters to increase spectral efficiency as compared with OFDM, particularly when it is integrated with offset quadrature amplitude modulation (OQAM). The digital filters are capable of overcoming interference between the neighboring subcarriers, reducing sensitivity to CFO, and increasing resistance to estimation error due to time/frequency shift by the elimination of side-lobes.

The key design features of future multicarrier modulation techniques include flexibility, versatility, scalability, and efficiency, as these qualities mean that a wide range of future applications may be served [7]–[10]. Several multiple access schemes have been recently suggested as candidates for 5G [11]–[13]. Among them are FBMC, generalized frequency division multiplexing (GFDM) [14], and universal filtered multi-carrier (UPMC) [15]. It is found that UPMC has better spectral efficiency than FBMC [16], [17]. In addition, unlike FBMC, UPMC offers simple implementation and backward compatibility, as well as integrity with OFDM. It is also easier to update a CP-OFDM system to UPMC, than it is the case with FBMC. Furthermore, its implementation is based on IFFT/FFT pairs combined with a prototype filter known as the Chebyshev window with Nyquist pulse shaping, which may minimize spectral leakage limitations of OFDM and FBMC. Therefore, UPMC combines the benefits of OFDM and FBMC systems [13] and avoids their shortcomings. Its operation is based on filtering successive sub-bands (a group of subcarriers). The existence of orthogonal carriers is not important, which means that better spectral efficiency and low latency are achieved. Thus, UPMC may support future 5G applications, such as IoT and D2D, which require a strict synchronous transition between the transmitter and the receiver [18], while V2V requires a signal that is less sensitive to CFO in order to achieve reliable communication between mobile nodes.

Fading channel estimation and equalization techniques are critical in wireless communication systems in order to recover transmitted symbols at the receiver side. Several UPMC system studies have examined the issue of channel estimation and equalization so far [19]–[21]. In contrast, numerous studies were adopted in regard to OFDM [22]–[25] and FBMC systems [26]–[29] and may be found in the literature. Channel estimation procedures in the

UPMC and FBMC/OQAM systems are not similar to the regular techniques used in the OFDM system, since channel frequency response values are real in OFDM and complex in FBMC and UPMC. In addition, the use of the sub-bands filtering process to reduce out-of-band emissions could yield different filter gains on different subcarriers for one UPMC symbol [30]. In [30], [31], the authors have proposed to estimate UPMC channels and show that the conventional pilot-aided channel estimation used in OFDM is efficient in UPMC. In [32], the authors investigated channel estimation in an UPMC system based on the comb-type pilot pattern, using various algorithms, such as LSLI, DFT and RMMSE.

Recently, in [33] the authors introduced the estimation of UPMC time-varying fading channels based on comb-type pilots using adaptive filters. Evolution of the channel fading process is modeled by an autoregressive model and tracked by the Kalman filter. Results of the simulation show that performance of the Kalman filter is better than that of least mean square (LMS) and recursive least square (RLS) filters. Nevertheless, the Kalman filter requires strict Gaussian assumptions on the state-space representation of the fading channel system. Indeed, the driving process and measurement noise must be independent white and Gaussian. To avoid these assumptions and any model uncertainties, the H_∞ filter may be employed [34]–[38].

In this paper, we readdress the estimation of UPMC time-varying fading channels based on comb-type pilot symbols. In particular, UPMC fading channels are modeled by an autoregressive process and tracked by an H_∞ filter, while the autoregressive parameters are estimated using the so-called Yule-Walker equations based on the Bessel autocorrelation function of the fading channel with a known Doppler rate. The structure of the paper is as follows. The UPMC transceiver is introduced in Section 2. Autoregressive modeling and estimation of the UPMC fading channels by using the H_∞ filter are described in Section 3. Simulation results are presented in Section 4. Finally, the conclusions are drawn in Section 5.

2. UPMC System Model

The UPMC system is built on filtering a set of sub-bands, instead of filtering the whole spectrum, as it is the case in OFDM, or a single carrier, as in FBMC. Therefore, the spectrum range in the UPMC system will be formed from successive sub-bands, as shown in Fig. 1 [39].

Table 1 shows a comparative study between three multicarrier modulation systems: OFDM, FBMC, and UPMC. It

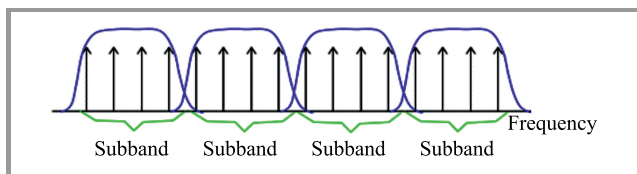


Fig. 1. UPMC signal in frequency domain.

is clear that the UPMC system is more attractive than the other two systems.

Table 1

Comparison between OFDM, FBMC, and UPMC systems

Item	OFDM	FBMC	UPMC
PAPR	High	High	High
Overhead	High	Low	Low
Receiver complexity	Low	High	High
MIMO support	Yes	No	Yes
Short bursts	No	No	Yes
Fragmented spectrum	No	Yes	Yes

Figures 2 and 3 show UPMC transmitter and receiver models, respectively. The operation of UPMC is based on filtering a group of sub-bands, instead of filtering a single carrier or the entire band. In the UPMC system, the input has the form of a set of complex QAM symbols. These symbols are then serial-to-parallel converted and divided into B sub-bands, containing the K symbol each. Frequency domain symbols $\mathbf{S} = [\mathbf{S}_{1K} \mathbf{S}_{2K} \dots \mathbf{S}_{BK}]$ are then converted to time domain symbols $\mathbf{X} = [\mathbf{x}_{1K} \mathbf{x}_{2K} \dots \mathbf{x}_{BK}]$ by the IFFT blocks. Since the number of complex symbols in each sub-band is lower than that of IFFT points N , the rest of the

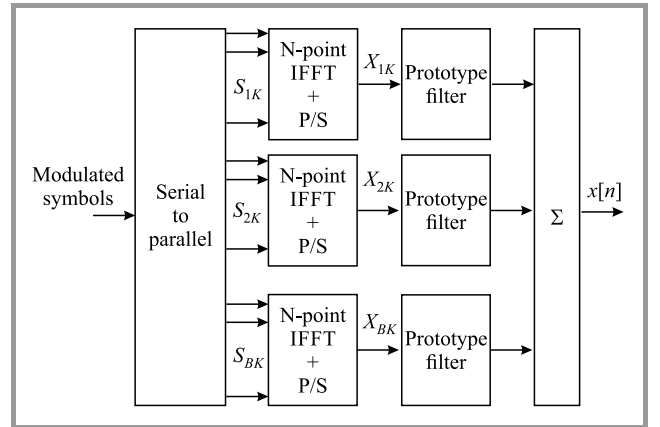


Fig. 2. UPMC transmitter block diagram.

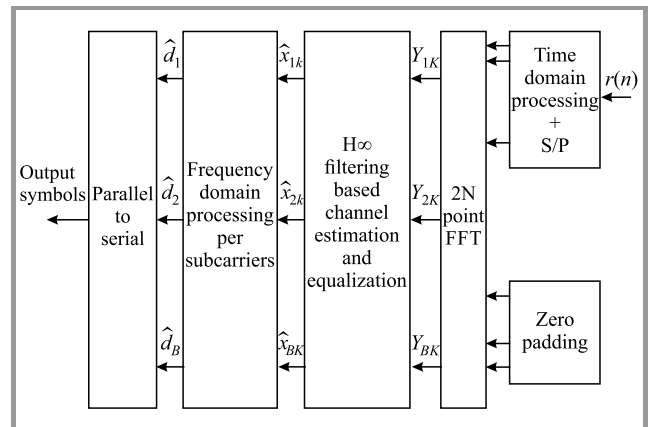


Fig. 3. UPMC receiver block diagram.

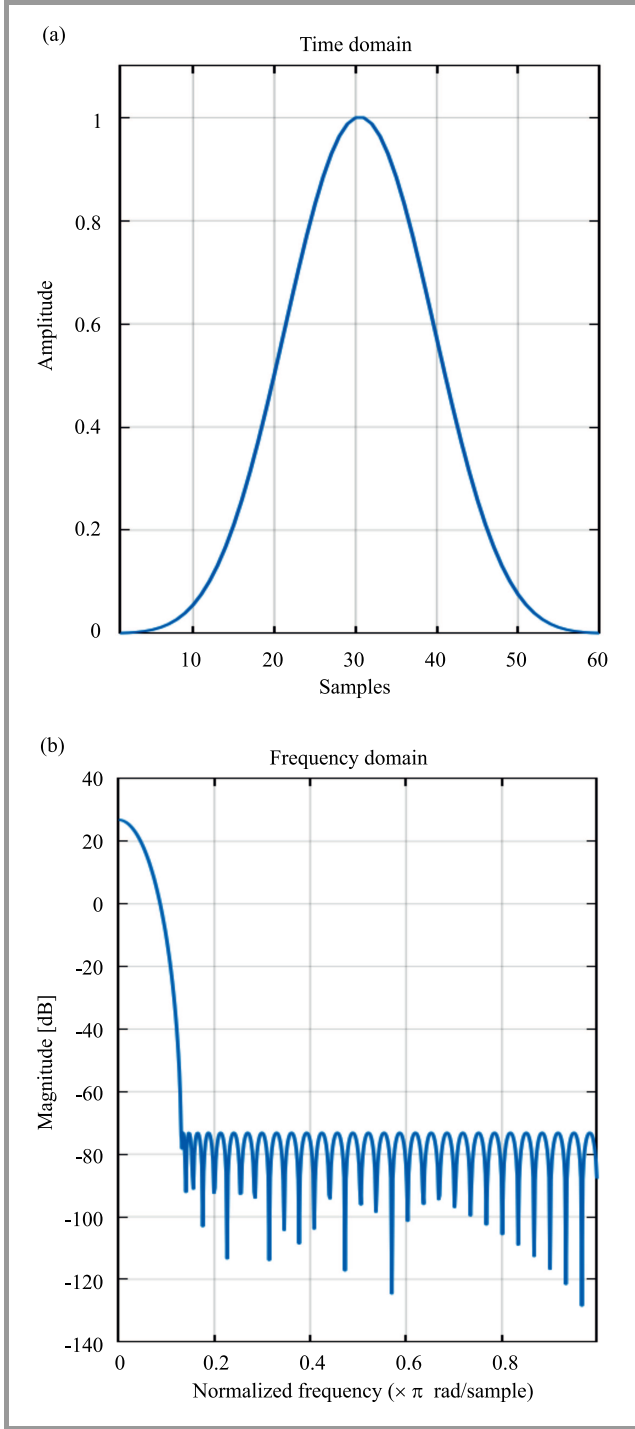


Fig. 4. Dolph-Chebyshev window in time and frequency domains for filter length $L = 60$.

block is padded with zeros. The output of IFFT is converted from parallel to serial. Each padded zero symbol is filtered with the Chebyshev finite impulse response (FIR) prototype filter with the length L , and a specific side lobe attenuation level. The Dolph-Chebyshev window had low side-lobe levels when compared with fixed window functions, such as the Hanning window. The Dolph-Chebyshev window in time and frequency domain for filter length of $L = 60$ is shown in Fig. 4. This will effectively decrease

the out-of-band (OOB) emission. The output signals from each prototype filter are then added, and transmitted into the wireless channel. The transmitted UPMC signal may be expressed as [34]:

$$x(n) = \frac{1}{N} \sum_{i=1}^B \sum_{l=0}^{N+L-1} \sum_{k=1}^K s_{ik} e^{j \frac{2\pi k l}{N}} f(n-l), \quad (1)$$

where $f(n-l)$ denotes the Dolph-Chebyshev filter whose length is L .

UPMC has more elasticity for filtering every sub-band by its spectrum. It may allow the scheme to adapt service types by regulating sub-band and filter coefficients only. Indeed, UPMC may provide simultaneous access for two different applications, such as M2M and D2D, in two dissimilar sub-bands, without creating inter-symbol band interference (ISBI) due to the reduction of OOB [30].

3. H_∞ Filter-based Channel Estimation

The signal received from the time-varying frequency flat fading channel with additive white Gaussian noise (AWGN) over the k -th sub-carriers may be written as follows:

$$r_k(n) = x_k(n)h_k(n) + w_k(n), \quad (2)$$

where $x_k(n)$ is the transmitted UPMC signal over the k -th subcarriers, $w_k(n)$ denotes the AWGN process over the k -th subcarriers with the noise processes over all subcarriers being mutually independent and identically distributed (i.i.d.), each with zero-mean and variance σ_w^2 . The time-varying flat fading process over the k -th subcarriers $h_k(n)$ is a complex Gaussian process $h_k(n) = |h_k(n)|e^{-j\phi_k(n)}$ with a Rayleigh distributed envelope and a uniformly distributed phase.

The statistical characteristics of the fading channel $h_k(n)$ are described by its power spectrum density (PSD) and autocorrelation function (ACF). The PSD of the fading process is usually represented by the well-known U -shaped band limited Jakes spectrum with the maximum Doppler frequency f_d [40]:

$$\Psi_{hh}(f) = \begin{cases} \frac{1}{\pi f_d \sqrt{1 - (\frac{f}{f_d})^2}}, & \text{for } f \leq |f_d| \\ 0, & \text{otherwise} \end{cases}, \quad (3)$$

where $f_d = \frac{v}{\lambda}$ with v denotes the mobile speed and λ denotes the signal wavelength.

The corresponding discrete-time ACF is given by:

$$R_{hh}(m) = J_0(2\pi f_d T_s |m|), \quad (4)$$

where $J_0(\cdot)$ denotes the zero-ordered Bessel function, T_s denotes the symbol period, and $f_d T_s$ is the Doppler rate.

To exploit the statistical properties of the fading channel given by its PSD – Eq. (3) and ACF – Eq. (4), the fading process over the k -th carrier is often approximated by

a p -th order AR process, denoted by AR(p) and defined as follows [41]:

$$h_k(n) = -\sum_{i=1}^p a_i h_k(n-i) + v_k(n), \quad (5)$$

where $v_k(n)$ denotes the zero-mean complex white Gaussian driving process with equal variance σ_v^2 over all carriers and $\{a_i\}$ denotes the AR parameters.

To estimate the fading process $h_k(n)$ over the k -th carrier, and for the sake of simplicity and clarity of presentation, the carrier subscript is dropped. Let us define the state-vector as:

$$\mathbf{h}(n) = [h(n) \ h(n-1) \ \dots \ h(n-p+1)]^T. \quad (6)$$

Equations (5)–(6) may be written in the following state-space form:

$$\mathbf{h}(n) = \Phi \mathbf{h}(n-1) + \mathbf{g}v(n), \quad (7)$$

where Φ denotes the state transition model matrix and may be expressed as follows:

$$\Phi = \begin{bmatrix} -a_1 & -a_2 & \dots & -a_p \\ 1 & 0 & \dots & 0 \\ \vdots & \vdots & \ddots & \vdots \\ 0 & \dots & 1 & 0 \end{bmatrix}, \text{ and } \mathbf{g} = [1 \ 0 \ \dots \ 0]^T.$$

In addition, after dropping the carrier subscript, it follows from Eqs. (2) and (6) that:

$$r(n) = \mathbf{x}^T(n) \mathbf{h}(n) + w(n), \quad (8)$$

where:

$$\mathbf{x}^T(n) = [x_{k,pilot} \ 0 \ \dots \ 0], \quad (9)$$

with $x_{k,pilot}$ denoting the pilot symbols. Unlike Kalman filtering, H_∞ filtering not only deals with the estimation of the state vector $\mathbf{h}(n)$, but also makes it possible to focus on the estimation of a specific linear combination of the state vector components as:

$$z(n) = \mathbf{l} \mathbf{h}(n), \quad (10)$$

where $\mathbf{l}$ denotes a linear transformation operator whose size is $1 \times p$. In order to estimate the fading process $\mathbf{h}(n)$, the linear operator is set to be $\mathbf{l} = \mathbf{g}^T = [1 \ 0 \ \dots \ 0]^T$.

Given the state space representation of the fading channel system by Eqs. (7), (8), and (10), H_∞ filtering may provide an estimation of the fading process $\hat{\mathbf{h}}(n) = \mathbf{l} \hat{\mathbf{h}}(n)$ by minimizing the H_∞ norm of the transfer operator that maps the noises $w(n)$, $v(n)$, and the initial state error $\mathbf{e}_0 = \mathbf{h}(0) - \hat{\mathbf{h}}(0)$ to the estimation error $e(n) = h(n) - \hat{h}(n)$, as follows:

$$J_\infty = \sup_{w(n), v(n), \mathbf{h}(n_0)} J, \quad (11)$$

where:

$$J = \frac{\sum_{n=0}^{N-1} |e(n)|^2}{e_0^H \mathbf{P}_0^{-1} e_0 + \sum_{n=0}^{N-1} (Q_w^{-1} |w(n)|^2 + R_v^{-1} |v(n)|^2)}, \quad (12)$$

with $\mathbf{P}_0$, $Q_w > 0$ and $R_v > 0$ being the weighting parameters adjusted by the designer to achieve the best performance needs and N denotes the number of data samples.

However, as a closed-form solution to the above optimal H_∞ estimation problem does not always exist, the following suboptimal design strategy is usually considered [37]:

$$J_\infty < \gamma^2, \quad (13)$$

where $\gamma > 0$ is a prescribed threshold of disturbance attenuation. Taking into account the approach introduced in [37], there exists an H_∞ channel estimator $\hat{\mathbf{h}}(n)$ for a given $\gamma > 0$ if there exists a stabilizing symmetric positive definite solution $\mathbf{P}(n)$ to the following Riccati-type equation:

$$\mathbf{P}(n+1) = \Phi \mathbf{P}(n) \Phi^H + \mathbf{g} Q_w \mathbf{g}^T, \quad \mathbf{P}(0) = \mathbf{P}_0, \quad (14)$$

where:

$$\mathbf{C}(n) = \mathbf{I}_p - \gamma \mathbf{l}^T \mathbf{I} \mathbf{P}(n) + \mathbf{x}(n) R_v^{-1} \mathbf{x}^T(n) \mathbf{P}(n). \quad (15)$$

This leads to the following constraint:

$$\mathbf{P}(n) \mathbf{C}^{-1}(n) > 0. \quad (16)$$

If the condition (16) is satisfied, then the H_∞ channel estimator may be written as:

$$\hat{\mathbf{h}}(n) = \mathbf{l} \hat{\mathbf{h}}(n), \quad (17)$$

$$\hat{\mathbf{h}}(n) = \Phi \hat{\mathbf{h}}(n-1) + \mathbf{K}(n) \alpha(n), \quad \hat{\mathbf{h}}(0) = \mathbf{0}, \quad (18)$$

where the so-called innovation process $\alpha(n)$ and the H_∞ estimator gain $\mathbf{K}(n)$, are given, respectively, by:

$$\alpha(n) = r(n) - \mathbf{x}^T(n) \Phi \hat{\mathbf{h}}(n-1), \quad (19)$$

$$\mathbf{K}(n) = \mathbf{P}(n) \mathbf{C}^{-1}(n) \mathbf{x}(n) R_v^{-1}. \quad (20)$$

It should be noted that the H_∞ channel estimator (14)–(20) has a similar observer structure as its Kalman counterpart. However, due to (15), the H_∞ channel estimator has a computational cost that is slightly higher than that of the Kalman solution. Indeed, if the weighting parameters $\mathbf{P}_0$, Q_w and R_v are chosen to be σ_w^2 , σ_v^2 , respectively, and the initial error covariance matrix is $\mathbf{h}(0)$, then as $\gamma \rightarrow \infty$ the H_∞ estimator is reduced to the Kalman estimator.

According to Fig. 2, the received signal $r_k(n)$ is first passed into the $2N$ point FFT block that transforms the time domain signal to the frequency domain. This is necessary for recovering symbols at N subcarriers from $N+L-1$ received signal samples. The result of the FFT block transformer is:

$$Y(k) = \frac{1}{\sqrt{N}} \sum_{l=0}^{N+L-2} r_k(n) e^{-\frac{j2\pi lk}{2N}}, \quad k = 0, 1, \dots, 2N-1. \quad (21)$$

Once the fading channel $h_k(n)$ at pilot symbol location is estimated using the proposed H_∞ based channel estimator, channel equalization may be performed by multi-

plying Eq. (24) with a normalized version of the complex conjugate of the fading channel estimate $\hat{h}_k(n)$ as:

$$\hat{x}_k(n) = Y_k(n) \left(\frac{\hat{h}_k^*(n)}{|\hat{h}_k(n)|^2} \right). \quad (22)$$

4. Simulation Results

In this section, a comparative simulation study concerned with estimation of the UPMC fading channels is carried in connection with the proposed H_∞ based channel estimator and other channel estimators based on LMS, RLS and Kalman filters. Several interpolation techniques, such as linear interpolation, spline interpolation, and low-pass interpolation are investigated for estimating the fading channel at data symbol location. The fading channel is generated according to the AR model with the order of $p = 2$ [41]. The simulation parameters are summarized in Table 2.

Table 2
Simulation parameters

Modulation type	QAM modulation
Filter type	Dolph-Chebyshev with length $L = 60$
AR parameters	$a_1 = 1.96, a_2 = -0.99$
LMS step size	$\mu = 0.03$
RLS forgetting factor	$\lambda = 0.4$
Pilot ratio	1/8
Number of pilot symbols N_p	1024

Figure 5 shows the bit error rate performance of UPMC versus SNR with QAM. It is clear that the QAM modulation with bits-per-subcarrier of $M = 4$ results in the best BER performance.

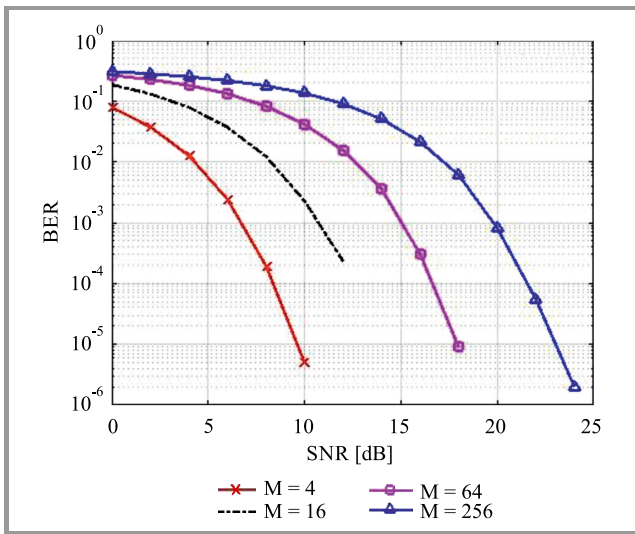


Fig. 5. BER versus SNR for UPMC with QAM modulation. (For color pictures see the electronic version of the paper.)

Figures 6–8 illustrate power spectrum density (PSD) with the same number of subcarriers (200 subcarriers) for UPMC, OFDM, and FBMC systems, respectively. It is clear that PSD of the UPMC system is more attractive than that of OFDM and FBMC.

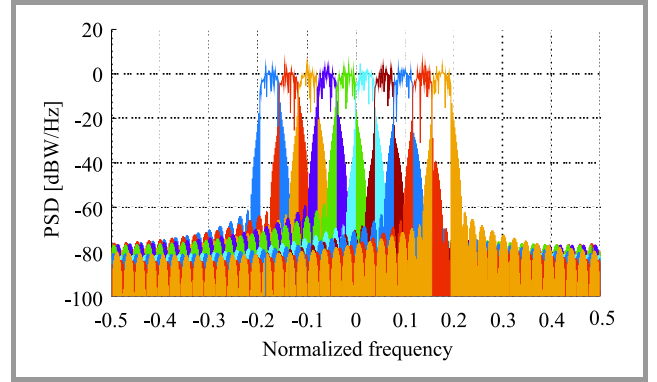


Fig. 6. PSD versus normalized frequency for UPMC system.

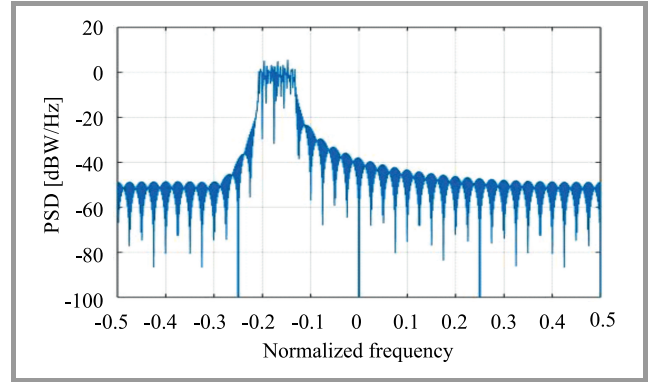


Fig. 7. PSD versus normalized frequency for OFDM system.

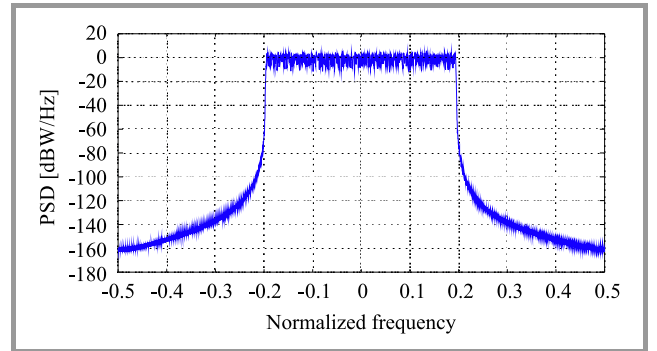


Fig. 8. PSD versus normalized frequency for FBMC system.

Figures 9 and 10 show the BER performance versus SNR for the UPMC system with two different values of Doppler rates $f_d T_s = 0.1111$ and $f_d T_s = 0.0741$, respectively. One may notice that the proposed H_∞ channel estimator yields the best BER performance, particularly at high Doppler rates. In addition, the LMS and RLS channels estimators fail to track the fading channel at high Doppler rates, which results in worse BER performance.

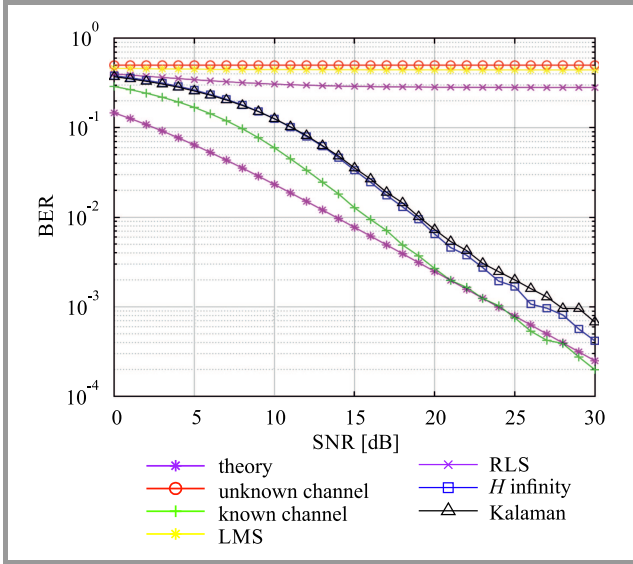


Fig. 9. BER vs. SNR for $f_d T_s = 0.1111$.

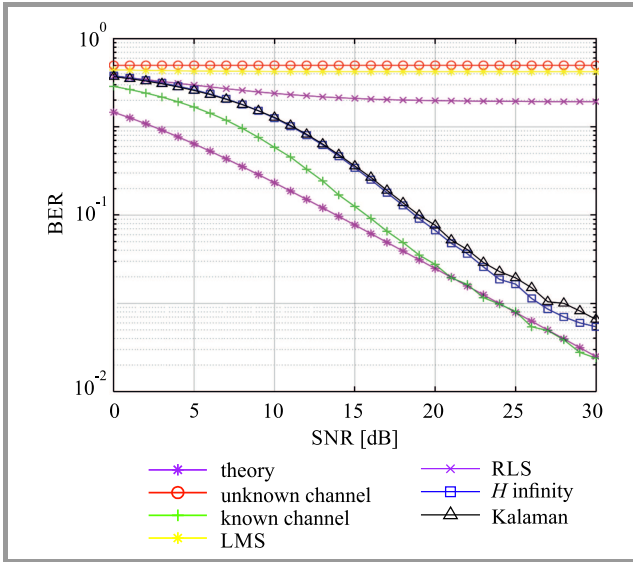


Fig. 10. BER vs. SNR for $f_d T_s = 0.0741$.

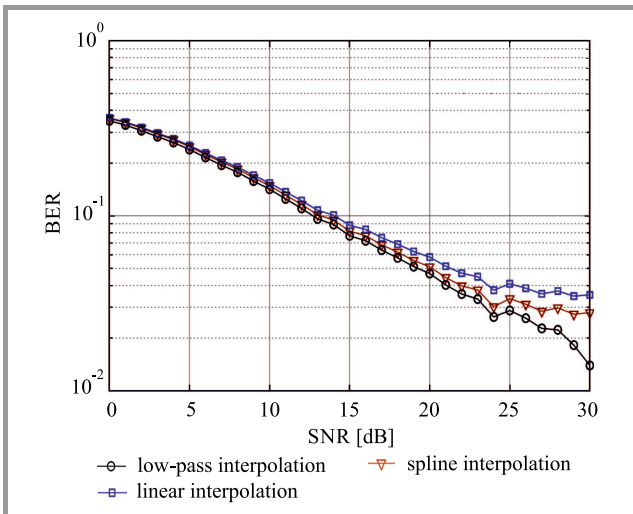


Fig. 11. BER vs. SNR with various interpolation techniques.

Figure 11 illustrates the BER performance of the UPMC system when employing the proposed H_∞ channel estimator with various interpolation methods for the Doppler rate of $f_d T_s = 0.0167$. It is confirmed that low-pass interpolation yields the best BER performance.

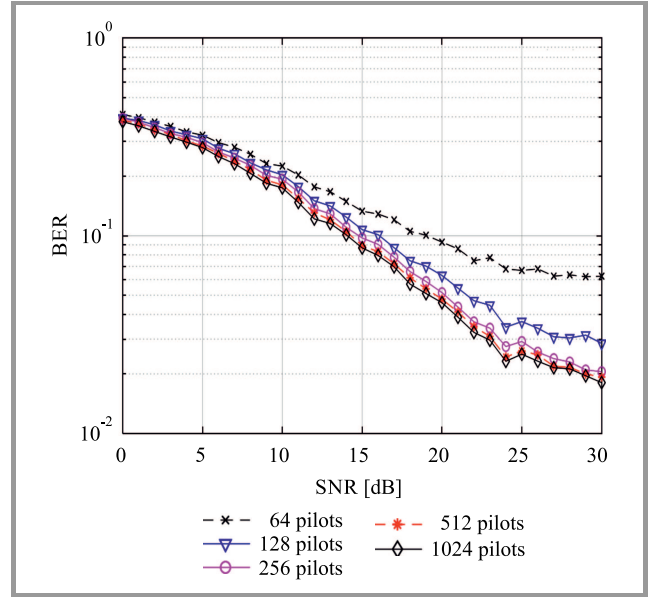


Fig. 12. BER vs. SNR with various pilot symbol values.

Figure 12 shows the effect that changing the number of pilot symbols N_p has on the BER performance of the UPMC system when using the proposed channel estimator with low-pass interpolation. Indeed, increasing the number of pilot symbols N_p will improve the BER performance of the system.

5. Conclusion

The UPMC system is a very strong candidate for the future generation of wireless applications. Indeed, it combines the advantages of both OFDM and FBMC and avoids their drawbacks. It achieves a higher data rate and higher spectral efficiency than other modulation systems. Furthermore, it solves the fixed synchronization problem experienced by current modulation systems and supports new applications, such as IoT, D2D, V2V, and M2M.

Estimation and equalization of UPMC fading channels based on the comb-type pilot symbol arrangement is addressed in this paper. The estimation process is carried-out in two stages. In the first stage, the fading channel at pilot carrier is estimated by using the H_∞ filter, the Kalman filter, the RLS algorithm, and the LMS algorithm. In the second stage, estimation of the fading channel at data symbol position is performed by using various interpolation techniques (linear, low-pass, and spline). Simulation results show that the proposed H_∞ filter-based channel estimator outperforms LMS, RLS, and Kalman estimators. In addition, the low-pass interpolation is confirmed to outperform both spline and linear interpolation.

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First and Second Order Characteristics of a Wireless Three-hop Relay Channel with the Presence of Rayleigh Fading

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Abstract—In this paper, a wireless three-hop relay communication system operating in a Rayleigh multipath fading environment is analyzed. The output signal from such a system is a product of signal envelopes from all sections, meaning that it is actually a product of three random variables (RVs) with Rayleigh distribution. We considered here the first-order characteristics: probability density function (PDF), cumulative distribution function and outage probability (OP). Then, the moments and amount of fading (AoF) were derived in the closed forms. The second order characteristics we present include the following: level crossing rate (LCR) and average fade duration (AFD). A few graphs are given to show the impact of the specific parameters of the wireless three-hop relay system.

Keywords—first order performance, Rayleigh fading, second order performance, three-hop relay telecommunication systems.

1. Introduction

The output signal of a wireless relay system is a product of signal envelopes from each section. Due to this fact, the performance of products of a higher number of random variables (RVs) has become an important topic over the past decade. RV products are applied in wireless channel modeling, multi-hop relay systems, multiple input multiple output (MIMO) keyhole systems, cascaded channels with fading, but also in other natural sciences, such as biology and physics (especially quantum physics), and also in social sciences, namely econometrics [1]–[3].

The first studies in this field were carried out in the previous century [4]. Donahue *et al.* were the first to study distributions of the products of two RVs [5]–[8]. In the new millennium, research in that field commenced again [9]–[12], and newer papers greatly impact practical applications in wireless relay communication systems [13]–[16].

Previously published papers focused primarily on products of two RVs. By using the level crossing rate (LCR) of a product of two Nakagami- m RVs from [3], the aver-

age fade duration (AFD) of a wireless relay system which works in a Nakagami- m fading channel is obtained. Further, probability density function (PDF) of the product of Rayleigh, exponentially, Nakagami- m and gamma RVs is calculated based on the Mellin transform in [17]. The authors in [18] present double fading channels with a line-of-sight (LOS) channel (double Rician fading), typical of keyhole MIMO models.

An analysis of the features of the product of two independent κ - μ distributed RVs is given in [19]. The analytical formulas for PDF and cumulative distribution function (CDF) are derived. In [20], new formulas for PDF, CDF, and moment generating function (MGF) are given, based on their expressions for higher order moments, amount of fading (AoF) and the channel quality estimation index in closed-forms. The formulas for outage probability (OP), average channel capacity (aCC), average symbol error probability (aSEP) and average bit error probability (aBEP) are derived as well. Some properties of the product of independent RVs with a general α - μ distribution, such as: PDF, CDF, moments, OP and AoF, are presented in [13] in closed-form expressions. This type of fading impacts also the dual-hop wireless communication system with a non-regenerative relay that is considered in [21]. The authors derived closed-form expressions for MGF, CDF, and PDF of the harmonic mean of the end-to-end signal-to-noise ratio (SNR), using the α - μ fading model. They also determined closed-form expressions for the end-to-end capacity and outage capacity of that system.

Dual-hop communication systems are known as tools enabling to extend coverage and reduce transmit power in scenarios where one-hop communication is not possible. An example of a dual-hop communication system is presented in Fig. 1 and in many papers such as [21]–[23]. PDF, CDF and MGF of the total received SNR of a dual-hop amplify-and-forward relay (AFR) transmission over non-identical Rician fading channels are obtained in the form of rapidly

converging infinite sums in [23]. Then, OP and aBER are calculated.

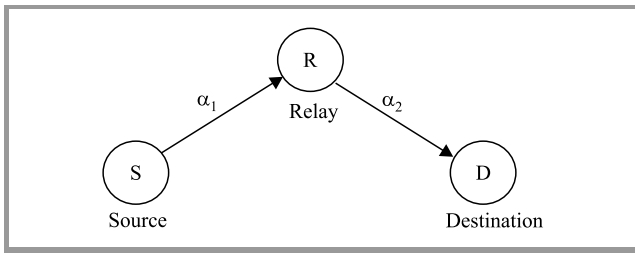


Fig. 1. Example of a dual-hop wireless communication system [22].

One group of authors has observed in [24] such a scenario for general α - η - μ fading distribution, performed an analysis of SNR for dual-hop amplify-and-forward (AF) cooperative communication systems, and derived ergodic capacity. The first-order outage statistics (CDF, OP and aBER) of an asymmetrical radio-frequency (RF) optical wireless (OW) dual-hop AFR communication system have been processed in [25]. Their system is modeled as a product of independent Nakagami- m and double squared Nakagami- m (or gamma-gamma) random processes. A similar scenario was also analyzed in [26].

By dint of spatial diversity and dual-hop relaying widely adopted in cooperative wireless systems, communication coverage may be effectively extended and the potentially severe channel fading may be mitigated [27]. The AF relaying protocol has been applied, since it generates an effective tradeoff between low implementation costs and a sufficient performance level. So, satellite communication represents a direct application of dual-hop AF relaying configurations. New calculations for measuring important performance parameters of a satellite relay have been performed in the presence of shadowed Rician fading for OP, ASEP and ergodic capacity of the end-to-end SNR.

One of the newest papers focusing on this area is [28]. Its authors analyzed the performance of a dual-hop cooperative decode-and-forward (DF) relaying system with beamforming relying on different adaptive transmission techniques over κ - μ shadowed fading channels.

Although three-hop relay systems have been discussed before [29]–[31], our group of authors is still among the first dealing with this telecommunication system configuration. So, we started observing the products of three RVs with different distributions and deriving system performance in closed forms [32]–[36]. The product of three RVs in relay telecommunication channels when multipath fading is present, is given in [32]. The LCR of the product of Nakagami- m RV, Rician RV and Rayleigh RV is obtained in a closed form in [33]. The product of three Nakagami- m RVs is analyzed in [34] and [35], and the statistics of the second order of the product of three Rician random variables is analyzed in [36].

Before our investigations were conducted, a survey on packet retrieval for a three-hop routing protocol in a hybrid

wireless network was presented in [29]. On the other hand, paper [30] presents a distributed three-hop routing protocol (DTR) for hybrid wireless networks, combining the advantages of mobile ad-hoc networks and infrastructure wireless networks, and achieving ultra-high performance.

Fixed cellular relays used in modern wireless communication systems are powerful technique allowing to improve reliability and to increase coverage under shadowing propagation conditions and limited transmit power conditions [31]. There, the lower and upper bounds of the sum capacity of different relaying strategies are analyzed. Figure 2 presents an example of a spectral efficiency sharing model deployed in two adjacent base stations, as described in this article [Fig. 1, 31]. In Fig. 3, a three-hop relay with two channels is shown [37].

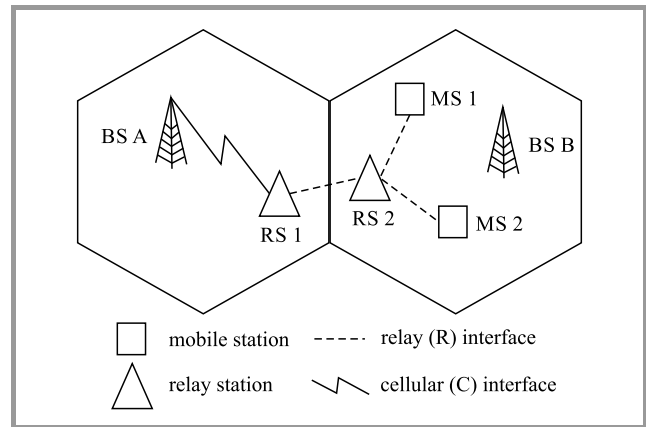


Fig. 2. Example of a dual-hop communication system [31].

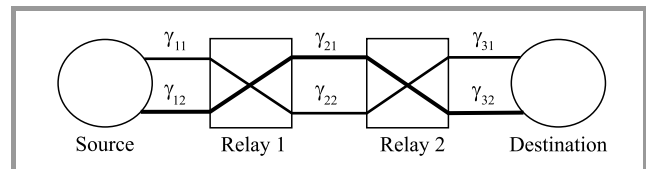


Fig. 3. A three-hop relay with two channels [37].

In a mobile ad-hoc network, data is routed to its destination through the intermediate nodes in a multi-hop manner. One of multi-hop system models, using antenna selection per hop, is plotted in Fig. 4 [38].

In the aforementioned work, OP of multi-hop MIMO relaying with a transmit antenna selection and an ideal relay gain over the Rayleigh fading environment is observed. Further, in [39]–[41], multiple RVs with different distributions are studied. So, PDF of the product of n -Rayleigh distribution is obtained in [39]. Distribution is derived by dint of inverse Mellin transform, and is presented using the Meijer G-function. An analysis of the second order statistics of the AF multi-hop Rayleigh fading environment is presented in [40]. The output signal from this cascaded channel is a product of N Rayleigh fading signal envelopes, written as N *Rayleigh RV. The formulas for LCR and AFD are derived for such an RV. Then, PDF, CDF, and mo-

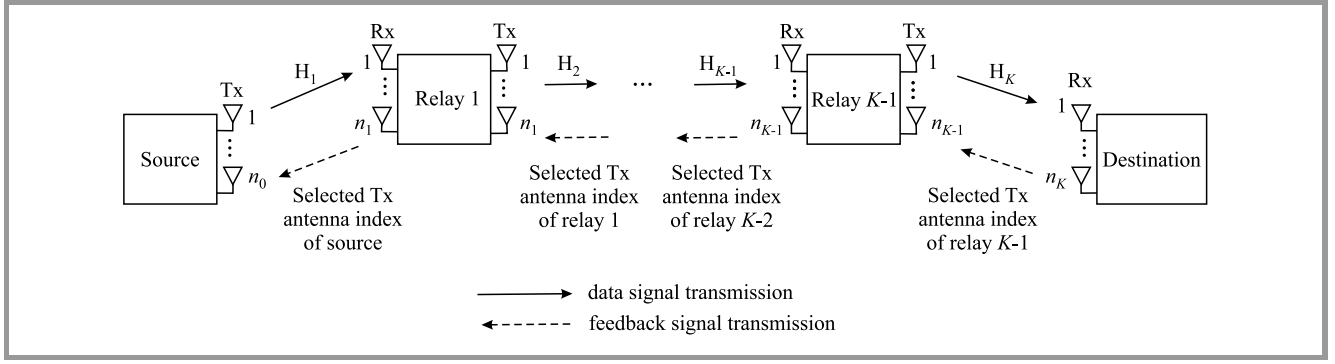


Fig. 4. A multi-hop system using antenna selection per hop [38].

ments of the N^* Nakagami distribution are worked out in closed forms across the Meijer's G-function in [41]. The expressions are derived for OP, AOF, and aBEP, for a few modulation formats.

Unlike previously published works, we derived some second-order performance parameters in their closed forms, because time variations of the fading environments are characterized by second order statistics. This applies, in particular, to LCR and AFD of wireless links. These two important statistics are used for Markov modeling of wireless channels [42] and for designing diversity combiners which will be used in wireless communication systems to mitigate the impact of fading. Consequently, they are important topics of many available articles.

This paper is concerned primarily with a three-hop wireless relay system functioning in the presence of Rayleigh fading. The multi-hop relay system's output signal is in fact a product of output signals from all sections (hops) [43]. For such a system, we obtained in [44]: probability density function, cumulative distribution function, outage probability, and the following second order system performance: level crossing rate and average fade duration. To complete the set of first order system performance, moments and amount of fading will be calculated in closed forms in this article.

This paper consists of four sections. In introduction, the topic is presented and the literature is analyzed. In Section 2, first order system characteristics of the product of three RVs are analyzed. Second order characteristics for triple Rayleigh RV are given in Section 3. All performance parameters are validated by graphical expressions. Section 4 presents the conclusion and proposals concerning future work.

2. First Order Characteristics of the Product of Three Rayleigh Random Variables

The multipath transmission in wireless systems is shown in Fig. 5 [45]. The path from the transmitter to the receiver consists of more distinct paths. In this figure, three such paths between the transmitter and the receiver are pre-

sented. In this scenario, the line-of-sight path does not exist. Only multiple indirect paths are available, and this is a Rayleigh channel.

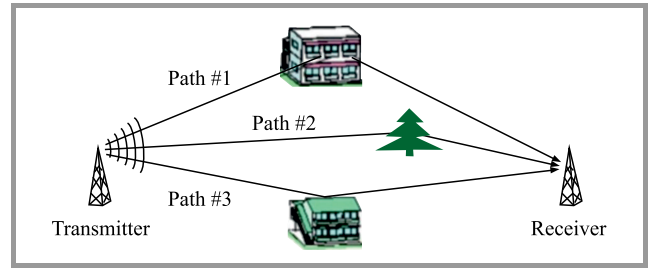


Fig. 5. Multipath transmission in wireless systems without a direct path between the transmitter and the receiver [45].

The signal envelopes represent random variables x_i , $i = 1, 2, 3$ which are Rayleigh distributed [46]:

$$p_{x_i}(x_i) = \frac{2x_i}{\Omega_i} e^{-\frac{x_i^2}{\Omega_i}}, \quad x_i \geq 0, \quad (1)$$

where $\Omega_i = E\{x_i^2(t)\}$, $1 \leq i \leq 3$, are the mean powers of the i -th RV [Eq. (1), 40].

We assume that random variable x is a product of three Rayleigh RVs [Eq. (2), 44]:

$$x = \prod_{i=1}^3 x_i. \quad (2)$$

Based on Eq. (2), it is valid that: $x_1 = \frac{x}{(x_2 x_3)}$.

2.1. PDF of the Product of Three Rayleigh RVs

The probability density function of the product of three Rayleigh RVs x is given by [Eq. (5), 44]:

$$\begin{aligned} p_x(x) &= \int_0^\infty \int_0^\infty \frac{1}{x_2 x_3} \frac{x}{x_2 x_3} \frac{x_2}{\Omega_2} \frac{x_3}{\Omega_3} \frac{1}{\Omega_1} e^{-\frac{1}{\Omega_1} \left(\frac{x}{x_2 x_3}\right)^2 - \frac{x_2^2}{\Omega_2} - \frac{x_3^2}{\Omega_3}} dx_2 dx_3 = \\ &= \frac{8}{\Omega_1 \Omega_2 \Omega_3} x \int_0^\infty x_2^{-1} K_0 \left(2 \sqrt{\frac{1}{\Omega_1 \Omega_3} \frac{x^2}{x_2^2}} \right) dx_2, \quad (3) \end{aligned}$$

where $K_n(x)$ is the modified Bessel function of the second kind [Eq. (1.1), 47].

In Fig. 6 [Fig. 1, 44], some curves are presented showing PDF of the product of three Rayleigh RVs depending on that product for more values of the signal's average power Ω_i at all sections. One may notice that PDF increases at low x values for all of Rayleigh signal powers values, then reaches its maximum and decreases at higher signal envelope values.

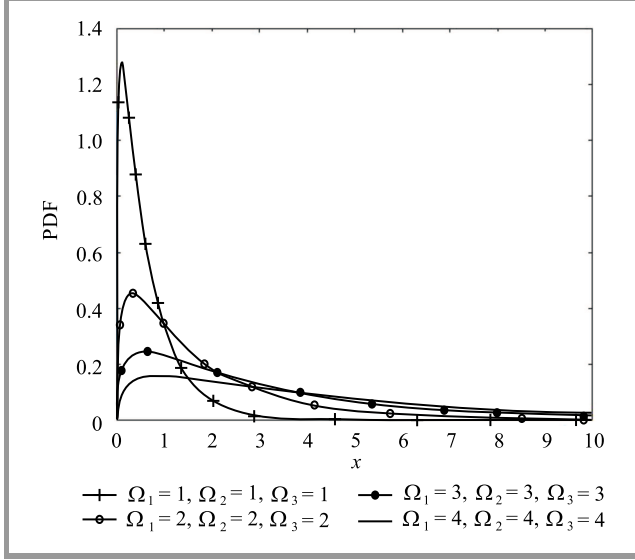


Fig. 6. PDF of the product of three Rayleigh RVs for different signal powers values.

We can notice from this figure that small amplitudes affect the PDF curves more distinctly.

2.2. CDF of the Product of Three Rayleigh RVs

The cumulative distribution function of x (product of three Rayleigh RVs) was derived in [Eq. (6), 44] as:

$$F_x(x) = \int_0^x p_x(t) dt = \frac{8}{\Omega_1 \Omega_2 \Omega_3} x^2 \sum_{j_1=0}^{\infty} \frac{1}{2(j_1)} \frac{x^{2j_1}}{\Omega_1^{j_1}} \times \int_0^{\infty} dx_2 x_2^{-1} e^{-\frac{1}{\Omega_2} x_2^2} \left(\frac{\Omega_3}{\Omega_1} x^2 \right)^{-j_1} K_{-2j_1} \left(2 \sqrt{\frac{x^2}{\Omega_1 \Omega_3 x_2^2}} \right) \cdot \quad (4)$$

2.3. OP of the Product of Three Rayleigh RVs

Outage probability is defined as the probability of the information rate (signal envelope, SNR or signal-to-interference ratio (SINR)) being lower than the information rate threshold value.

It is the probability of an outage occurring within a specified time period and determined by the following formula [Eq. (2.23), 47]:

$$OP = P(X \leq x) = \int_0^x p_x(t) dt = F_x(x) \cdot \quad (5)$$

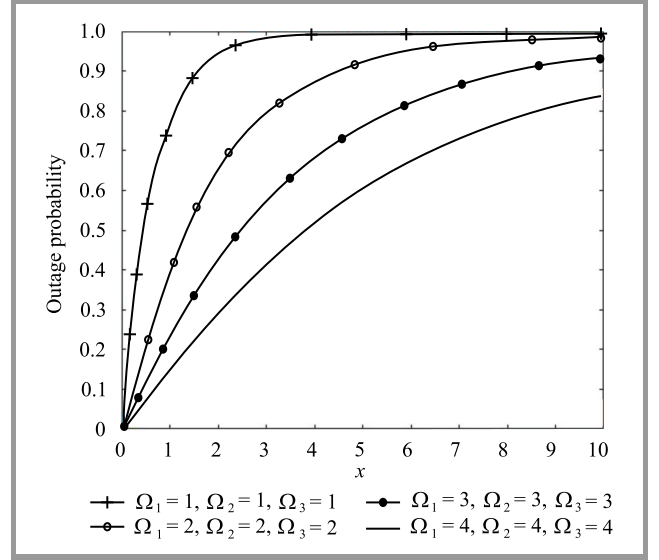


Fig. 7. OP of the product of three Rayleigh random variables for different average signal envelope powers Ω_i .

It is mathematically the CDF from Eq. (4).

OP is presented in Fig. 7 [Fig. 2, 44] for a few values of the average signal envelopes' powers from each of the sections. It was concluded that OP grows along with an increase in signal envelope x and achieves its maximum value. Likewise, it is evident that OP drops with the increase in Ω_i .

2.4. Moments of the Product of Three Rayleigh RVs

Moment is a quantitative measure that describes the shape of a function in mathematics. It is used both in mechanics and statistics. If the function concerned is a probability distribution function, then moments have the following meanings:

- zero-th moment is the total probability,
- first moment is the mean of the signal, i.e. the expected value,
- second moment is the variance or the signal's average power,
- third moment is the skewness,
- fourth moment is the kurtosis.

For the product of three Rayleigh RVs, the first moment m_1 , or the mean of the signal, is defined by:

$$m_1 = \bar{x} = \int_0^{\infty} x p_x(x) dx = \int_0^{\infty} x^2 \frac{8}{\Omega_1 \Omega_2 \Omega_3} \int_0^{\infty} x_2^{-1} K_0 \left(2 \sqrt{\frac{1}{\Omega_1 \Omega_3} \frac{x^2}{x_2^2}} \right) dx_2 dx \cdot \quad (6)$$

The n -th moment of an independent Rayleigh distributed RV is given by Eq. (3) in [39]:

$$E[x_i^n] = \Omega_i^n \left[\Gamma\left(\frac{n}{2} + 1\right) \right].$$

If we use Eq. (8) derived in [48], and enter Nakagami- m parameters as $m_1 = m_2 = m_3 = 1$, the first moment of the product of three Rayleigh RVs may be expressed as:

$$m_1 = \sqrt{\Omega_1 \Omega_2 \Omega_3} [\Gamma(3/2)]^3. \quad (7)$$

The second moment, m_2 , is:

$$\begin{aligned} m_2 = \overline{x^2} &= \int_0^\infty x^2 p_x(x) dx = \\ &= \frac{8}{\Omega_1 \Omega_2 \Omega_3} \int_0^\infty x^3 \int_0^\infty x_2^{-1} K_0 \left(2 \sqrt{\frac{1}{\Omega_1 \Omega_3} \frac{x^2}{x_2^2}} \right) dx_2 dx = \\ &= (\Omega_1 \Omega_2 \Omega_3) [\Gamma(2)]^3. \end{aligned} \quad (8)$$

The second moment is the variance. The standard deviation is a square root of the variance. In telecommunications, this is the average power of the signal.

The normalized third central moment is the skewness. In the probability theory and statistics, skewness is the size of the limitedness of distribution. It is a measure of symmetry or, more precisely, of a lack of symmetry. Distribution of a data set is symmetrical if it looks the same on the left- and right-hand side of the center value. In this case, the third moment is equal to zero. Also, skewness may assume positive or negative values, or even may be undefined.

A distribution that is skewed to the left, i.e. the tail of the distribution is longer on the left-hand side of the curve, has negative skewness and vice versa, a distribution skewed to the right has positive skewness. So, the third moment, m_3 , or the skewness, is:

$$\begin{aligned} m_3 = \overline{x^3} &= \int_0^\infty x^3 p_x(x) dx = \\ &= \frac{8}{\Omega_1 \Omega_2 \Omega_3} \int_0^\infty x^4 \int_0^\infty x_2^{-1} K_0 \left(2 \sqrt{\frac{1}{\Omega_1 \Omega_3} \frac{x^2}{x_2^2}} \right) dx_2 dx. \end{aligned} \quad (9)$$

As for previous moments, the final formula is obtained in a similar manner, from [Eq. (10), 48]:

$$m_3 = (\Omega_1 \Omega_2 \Omega_3)^{3/2} [\Gamma(5/2)]^3. \quad (10)$$

In Fig. 8, the moments of the product of three Rayleigh RVs are shown. One may see from this figure that all moments decrease with increasing Ω , in line with theoretical expectations.

The fourth moment, m_4 , the kurtosis, is a measure of whether the data are sharper or flatter distributed relative to normal distribution. This means that a kurtosis dataset

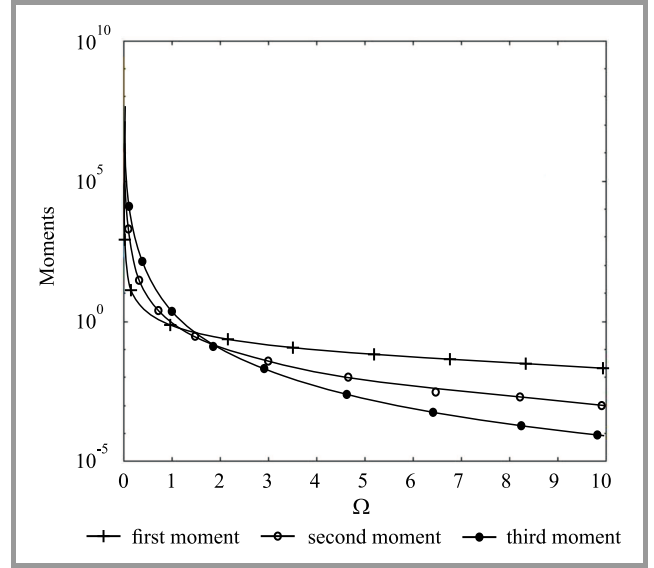


Fig. 8. Moments of the product of three Rayleigh RVs depending on signal power $\Omega = \Omega_1 = \Omega_2 = \Omega_3$.

has a large, significant peak near the mean value, suddenly decreasing and having long tails.

A dataset with a low kurtosis has a flat tip near the middle. Uniform distribution is an extreme case and standard normal distribution has a kurtosis equal to zero. In this case, positive kurtosis indicates distribution with a peak, and negative kurtosis indicates that distribution is more even.

Finally, the moment of n -th order, m_n is:

$$\begin{aligned} m_n = \overline{x^n} &= \int_0^\infty x^n p_x(x) dx = \\ &= \frac{8}{\Omega_1 \Omega_2 \Omega_3} \int_0^\infty x^{n+1} \int_0^\infty x_2^{-1} K_0 \left(2 \sqrt{\frac{1}{\Omega_1 \Omega_3} \frac{x^2}{x_2^2}} \right) dx_2 dx = \\ &= (\Omega_1 \Omega_2 \Omega_3)^{n/2} [\Gamma(1 + n/2)]^3. \end{aligned} \quad (11)$$

2.5. AoF of the Product of Three Rayleigh RVs

The severity of fading is defined by a measure known as the amount of fading [49]. This measure is used to describe different models which define the fading condition in radio channels. Here, we present comprehensive insights illustrating how faded a given channel is, based on fading severity in radio channels. This parameter does not depend on the mean power. It depends on the current signal value, SNR or SINR.

The amount of fading, connected with PDF of fading, is defined by the ratio of the variance of the received energy and the square of the mean of the received energy [Eq. (2), 49], [Eq. (2.5), 50]:

$$\text{AoF} = \frac{\text{var}(x^2)}{(\text{E}[x^2])^2},$$

with $\text{var}(\cdot)$ denoting variance and $\text{E}[\cdot]$ denoting the statistical average value or the mathematical expectation.

In [49, p. 29; 6], Charash gave this formula for AoF as a unified measure of the severity of the fading. AoF is usually independent of the average fading power. Amount of fading is obtained by developing the formula below; so, it is specified by [Eq. (9), 51]:

$$\begin{aligned} \text{AoF} &= \frac{\text{var}(x^2)}{(\mathbb{E}[x^2])^2} = \frac{\mathbb{E}[(x^2 - \Omega)^2]}{\Omega^2} = \\ &= \frac{\mathbb{E}[x^2] - (\mathbb{E}[x])^2}{(\mathbb{E}[x])^2} = \frac{\mathbb{E}[x^2]}{(\mathbb{E}[x])^2} - 1. \end{aligned}$$

AoF may be expressed directly from the moments as:

$$\text{AoF} = \frac{m_2}{m_1^2} - 1. \quad (12)$$

In Fig. 9, the amount of fading of the product of three Rayleigh RVs, i.e. of a three-hop Rayleigh channel, is presented versus average power envelopes Ω . It is worth noting that AoF is independent of Ω , which is in compliance with the theoretical assumptions that AoF typically does not depend on average power. As stated earlier, AoF provides a quantitative measure of fading that exists in wireless channels. Since the definition of AoF depends on two moments only (squared first and second moments of the power), it is possible to obtain the same value of AoF with more different PDFs.

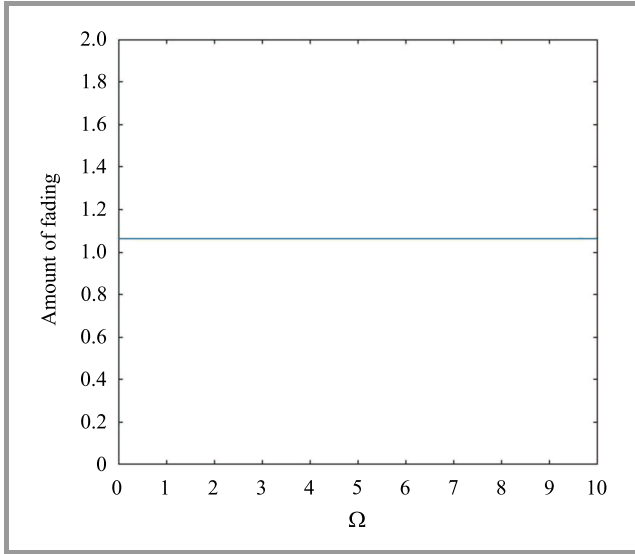


Fig. 9. Amount of fading of the product of three Rayleigh RVs versus signal power Ω .

In other words, quantification of fading that is defined in Eq. (12) does not fully quantify the measured statistics of the fading channel, or PDF of the received power [51]. The reasons for the signal power going below a predefined threshold are: data rate, coding, modulation, demodulation. Because of that, outage probabilities are used as well for quantification of the performance of the fading channels, as presented in Subsection 2.3 for this scenario.

3. Second Order Characteristics of the Product of Three Rayleigh Random Variables

3.1. LCR of the Product of Three Rayleigh RVs

LCR of X at threshold x is defined as the rate at which the random process crosses a given level x in the negative (or positive) direction [52]. To derive the LCR, we need to determine the joint probability density function (JPDF) between x and $\dot{x}$, $p_{x\dot{x}}$, and apply the Rice's formula [Eq. (2.106), 53]. So, LCR of the product x of three Rayleigh RVs is:

$$N_X(x) = \int_0^\infty \dot{x} p_{x\dot{x}}(x\dot{x}) d\dot{x}. \quad (13)$$

After adequate replacements, LCR of x is [Eq. (15), 44]:

$$\begin{aligned} N_X &= \frac{1}{\sqrt{2\pi}} \pi f_m \Omega_1^{\frac{1}{2}} x \frac{8}{\Omega_1 \Omega_2 \Omega_3} \times \\ &\int_0^\infty \int_0^\infty e^{-\frac{1}{\Omega_1} \frac{x^2}{x_2^2 x_3^2} - \frac{1}{\Omega_2} x_2^2 - \frac{1}{\Omega_3} x_3^2} \sqrt{1 + \frac{x^2}{x_2^4 x_3^2} \frac{\Omega_2}{\Omega_1} + \frac{x^2}{x_2^2 x_3^4} \frac{\Omega_3}{\Omega_1}} dx_2 dx_3. \end{aligned} \quad (14)$$

It is solved in [44] by using Laplace approximation theorem [54] and presented here in Fig. 10 [Fig. 3, 44]. In this figure, LCR, normalized by f_m , is shown versus the signal envelopes for different signal powers Ω_i . It is visible that LCR is increasing for small values of the signal envelope, reaches the maximum and then decreases for higher x values. Small signal envelopes have a greater impact on LCR. LCR is higher for bigger Ω_i . The system has better performance for lower LCR values.

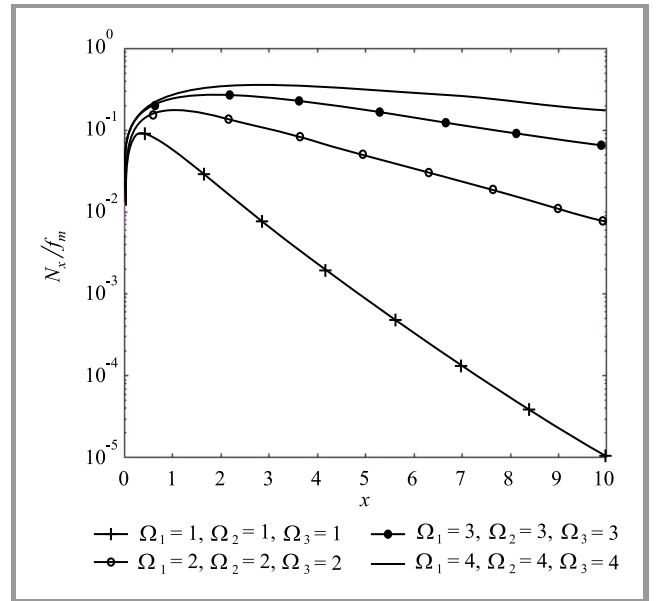


Fig. 10. LCR normalized by f_m versus signal envelope x for different values of signal power Ω_i

3.2. AFD of the Product of Three Rayleigh RVs

AFD of RV at threshold x defined as the average time over which the product of three Rayleigh RVs remains below x after passing the defined level in the downward (or upward) direction. It is the ratio of the OP and the LCR given by [55]:

$$T_x(x) = \frac{F_x(x)}{N_x(x)}, \quad (15)$$

with $F_x(x)$ from Eq. (4) denoting the CDF of x evaluated at X . The LCR $N_x(x)$ is given in Eq. (14).

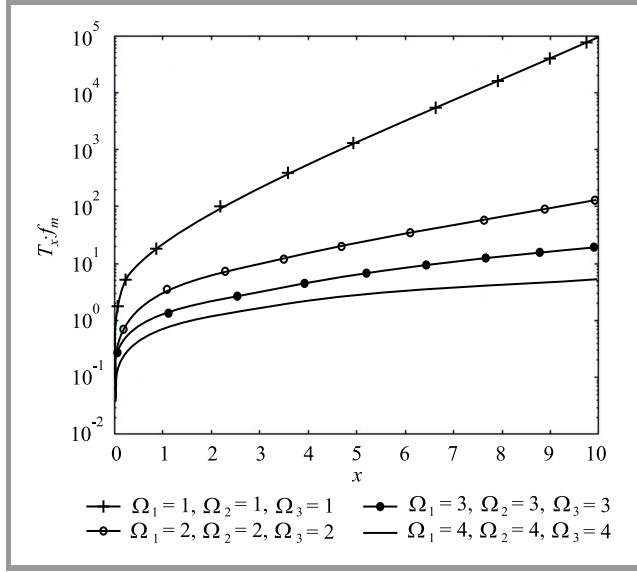


Fig. 11. AFD normalized by f_m depending on the signal envelope for some values of signal powers Ω_i .

In Fig. 11, [Fig. 4, 44], normalized AFD ($T_x f_m$) is shown depending on signal envelope x . It is visible that AFD is increasing for all signal envelopes and also that AFD has smaller values for bigger power values.

4. Conclusion

Owing to transmit power limitations, multi-hop communication in relay systems offers a very contemporary approach to improving transmission in cellular and ad-hoc networks. The advantages of multi-hop relaying are particularly pronounced for rural areas with small populations and low traffic densities [56].

In this article, we presented formulas for calculating PDF and CDF, and derived exact and simple closed-form formulas for moments of the product of three Rayleigh RVs. Based on these formulas, the outage probability and amount of fading for cascaded fading channels were worked out. LCR and AFD values for the same scenario, as obtained in [30], are also shown to complement the performance of a system of cascaded three-hop Rayleigh fading channels.

Our future work will be related to determining the performance for the product of multiple RVs with other general fading distributions κ - μ , η - μ , α - μ , α - η - μ ... used in multi-hop wireless relay communication systems.

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Extrapolation of Electromagnetic Response from Linear Antennas in Time Domain without Late-Time Instabilities in Numerical Solution of EFI Equation

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Abstract—The paper presents a new hybrid method relied upon to solve integral equations of the electric field in time domain and to model linear antennas with pulse excitation. The method consists in a mixed, numerical-analytical description of the process that helps maintain the stability of calculations during the late time phase. As described in the analytical part, the modified spherical Bessel function of the first kind allows for extrapolation with a high degree of accuracy and low-order expressions. The modified spherical Bessel functions of the first kind are of the oscillating character, and their combination with the exponential factor makes them convenient for extrapolation of the answer of the antenna with pulse excitation. New functions are introduced to computational practice.

Keywords—*electric field integral equation (EFIE), linear antennas, solution of time domain EFIE – late time instabilities.*

1. Introduction

The problem of obtaining a transient response from an arbitrarily shaped conducting body excited either as an antenna or a scatterer, raises much interest in the electromagnetic research community. Traditionally, in order to determine a wideband response of an antenna or a conducting structure, Maxwell equations are solved either in the time or frequency domain, using one of the many available computational electromagnetic methods.

The method of moments is one of the most popular ones. It allows to perform an electromagnetic analysis in the frequency domain, but in the case of wideband analysis, this approach may be rather complex. Therefore, for wideband analysis, the time domain approach is preferred. Moreover, a time domain formulation includes easier modeling of non-linear and time-varying media.

Using time domain method, wideband information may be obtained by exciting the structure with a narrow pulse

(Gaussian pulse), and by time-stepping the transient response to zero. However, for resonant structures, energy dissipation may be exceedingly slow. So, obtaining the complete response might require thousands of time steps, and the increased probability of numerical dispersion errors needs to be dealt with.

In order to solve the problems referred to above affecting a properly defined integral equation in the time domain, the marching-on in time (MOT) method is usually employed. Unfortunately, this algorithm suffers from a disadvantage consisting in the occurrence of late-time instabilities that take the form of exponentially increasing oscillations whose sign alternates at each time step due to the finite precision of the computation [1]. In order to overcome these instabilities, a specific technique is introduced. It is shown that the time and frequency response caused by a wideband source may be generated based on early-time and low-frequency data, by fitting the response with a summation of orthogonal polynomials (Hermite, Laguerre [2], Bessel-Chebyshev polynomials). But the use of polynomials only in order to represent typical responses (which decay slowly in the time domain) of resonant structures is computationally inefficient and numerically difficult in practice. Furthermore, an extrapolation in either the time or frequency domain is sometimes a numerically unstable process, and its accuracy often cannot be guaranteed [3]–[6].

Therefore, the aim to perform accurate extrapolation in time domain leads to the idea of extend the extrapolation technique by using damped modified spherical Bessel functions of the first kind. Using these functions enable efficiently and accurately modeling the resonance. Thus, the number of required functions is reduced, computation time decreased and the extrapolation accuracy is improved.

In this paper, a technique relying on stable, initial, late-time information is deployed to generate information for the

entire late time phase. Hence, using a time domain code generates the early-time and initial late-time information. The generation of this information is not only quite simple, but also not very computationally intensive. Next, the new modified spherical Bessel functions of the first kind are used to extrapolate the information in the time domain. In other words, by using early-time and initial late-time data and the extrapolation technique with damped modified spherical Bessel functions of the first kind, stable late-time information is obtained for the remaining portion of the process.

2. Formulation of Time Domain Electric Field Integral Equation

Let S denotes a perfectly conducting surface that may be closed or open, with a transient electromagnetic wave incident upon it. This incident wave induces a surface current density $\mathbf{J}(\mathbf{r}, t)$ on S , so the induced currents and charges become the sources of a secondary wave. The scattered electric field $\mathbf{E}^s(\mathbf{r}, t)$ computed from the surface current $\mathbf{J}$ is given by:

$$\mathbf{E}^s(\mathbf{r}, t) = -\frac{\partial \mathbf{A}(\mathbf{r}, t)}{\partial t} - \nabla \Phi(\mathbf{r}, t), \quad (1)$$

where $\mathbf{A}(\mathbf{r}, t)$ and $\Phi(\mathbf{r}, t)$ are the magnetic vector and the electric scalar potentials, defined as:

$$\mathbf{A}(\mathbf{r}, t) = \frac{\mu}{4\pi} \int_S \frac{\mathbf{J}(\mathbf{r}', \tau)}{R} dS', \quad (2)$$

$$\Phi(\mathbf{r}, t) = \frac{1}{4\pi\epsilon} \int_S \frac{q(\mathbf{r}', \tau)}{R} dS'. \quad (3)$$

In addition, the influence of the current density along surface S may be described by the vector potential, and the influence of charge density may be described by the scalar potential.

The permittivity and permeability of the surrounding medium are μ ($\mu = \mu_r \mu_0$) and ϵ ($\epsilon = \epsilon_r \epsilon_0$) respectively. $R = |\mathbf{r} - \mathbf{r}'|$ is the distance between observation point $\mathbf{r}$ and source point $\mathbf{r}'$ on S , and $\tau = t - R/c$ is the retarded time. R/c is the time for the wave to travel the distance between the observation and the source points. The velocity of propagation in the surrounding medium is $c = (\mu\epsilon)^{-1/2}$.

The surface charge density $q(\mathbf{r}, t)$ is related to the surface divergence of $\mathbf{J}(\mathbf{r}, t)$, based on the equation of continuity:

$$q(\mathbf{r}, t) = - \int_0^t \nabla \cdot \mathbf{J}(\mathbf{r}, t') dt'. \quad (4)$$

Using (4), Eq. (3) may be rewritten in the following form:

$$\Phi(\mathbf{r}, t) = \frac{1}{4\pi\epsilon} \int_S \int_0^t \frac{\nabla' \cdot \mathbf{J}(\mathbf{r}', t')}{R} dt' dS'. \quad (5)$$

On the surface of the perfectly conducting structure, the electromagnetic boundary condition for the intensity of the electric field requires that:

$$[\mathbf{E}^i(\mathbf{r}, t) + \mathbf{E}^s(\mathbf{r}, t)]_{\tan} = 0, \quad \mathbf{r} \in S. \quad (6)$$

The total tangential electric field is zero on the conducting surface for all times, and:

$$\left[\frac{\partial \mathbf{A}(\mathbf{r}, t)}{\partial t} + \nabla \Phi(\mathbf{r}, t) \right]_{\tan} = [\mathbf{E}^i(\mathbf{r}, t)]_{\tan}, \quad (7)$$

where $\mathbf{E}^i(\mathbf{r}, t)$ is the incident electric field (this field is parallel to the surface S) on the scatterer, and the subscript “tan” denotes the tangential component.

Equations (7), (2) and (5) represent a time domain electric field integral dependence from which the unknown current $\mathbf{J}(\mathbf{r}, t)$ may be determined.

Here, the primary objective is to obtain current distribution on a thin conducting wire (linear antenna), as a function of time when the wire is excited by a narrow electric Gaussian pulse. Furthermore, a “thin wire” approximation is assumed, so the current’s components exist along the z axis of the wire only. They are only a function of the z variable and the current flows only on the wire surface. Thus, there is neither current nor conductivity, across the wire. Moreover, the axially directed current may only change its value with distance along the wire, and the wire radius a is much lower than the wavelength and the antenna length.

Under the “thin wire” approximation, we have only the z component for the potentials given by Eqs. (2) and (3), so when the incident electric field impinges on a thin wire, current distribution satisfies the following equation:

$$\left[\frac{\partial A(z, t)}{\partial t} + \nabla \Phi(z, t) \right]_{\tan} = [E^i(z, t)]_{\tan}. \quad (8)$$

3. Numerical Solution of TD-EFIE

The time domain electric field integral equation (TD-EFIE) has been solved by applying the marching-on in time (MOT) procedure that is very efficient and simple to code. For the numerical solution of TD-EFIE, the time axis is divided into equal intervals Δt , and $t_k = k \cdot \Delta t$, ($k = 0, 1, 2, \dots, \infty$) is defined. A linear antenna of length L and radius a is divided into N sections with the length of Δz each.

The time derivative in Eq. (8) may be approximated by the forward, central or backward finite difference and an explicit or implicit scheme may be obtained [8]. All three cases could be combined (forward, central, backward) into a unified scenario in order to use the same code for implementing the various time – differentiating schemes, by:

$$\frac{A(z, t_k) - A(z, t_{k-1})}{\Delta t} + (1-p) \cdot \nabla \Phi(z, t_k) + p \cdot \nabla \Phi(z, t_{k-1}) = E^i(z, t_{k-p}), \quad (9)$$

where $p = 1$ for the forward, $p = 0$ for the backward and $p = 1/2$ for the central finite difference approximation of the derivative for the magnetic vector potential.

Equation (9) is solved numerically in two steps. The first step is connected with the space discretization problem, while the other one concerns the application of time algorithms that use data obtained in the first step. We use the method of moments for the Eq. (8) transformation to a set of the retarded differential equations (the space discretization problem).

For the second step, it is possible to use either the explicit or the implicit algorithm. In the explicit one, the length of time step Δt has to satisfy the following Courant condition [9] i.e. $\Delta t \leq R_{\min}/c$, where $R_{\min}$ is the smallest distance between the center of the discretization elements and c is the speed of light in the surrounding medium. In the implicit algorithm, the length of the time step may be chosen arbitrarily. Using a larger length of the time step increases the speed of the computation but lowers its accuracy. These approaches are the varieties of the MOT method.

If we focus on time instant $t = t_k$, Eq. (9) may be rewritten as [10]:

$$2A(z, t_k) + \Delta t \cdot \nabla \Phi(z, t_k) = 2\Delta t \cdot E^i(z, t_{k-1/2}) + 2A(z, t_{k-1}) - \Delta t \cdot \nabla \Phi(z, t_{k-1}) . \quad (10)$$

This is the formula that needs to be solved for the unknown current density $J(z, t)$. It is apparent that the left-hand side of Eq. (10) contains only the unknown currents at time instant $t = t_k$, while the right-hand side contains the known currents at $t = t_{k-1}$. Therefore, the algorithm may commence with the assumption that $J(t_0) = 0$ to find $J(t_1)$. When current $J(t_1)$ is computed, then it is possible to compute $J(t_2)$, and so on.

4. Extrapolation Process

The purpose of extrapolation is to use stable, initial late-time data to determine a representation of the complete response which accurately approximates late-time data even for the very late time phase.

In order to accurately represent the response from the antenna, which is usually slowly decaying over time, hundreds or thousands of orthogonal polynomials are potentially required [2]. The total number of polynomials N constitutes the number of equations to be solved in the extrapolation process. Therefore, it is wise to minimize N while maintaining accuracy. Furthermore, oscillations which may appear in higher-order polynomials may lead to computation errors and numerical instabilities during computations.

Therefore, we chose, as the basis, a new kind of functions, namely – special spherical Bessel functions of the first kind [11]. These functions are incorporated into the summations to accurately and efficiently represent the response.

Let the function $\bar{J}(t)$ denote the approximation of $J(t)$ at $z = 0$. In order to determine $\bar{J}(t)$, $J(t)$ is fitted with N functions $j_n(t)$:

$$J(t) \cong \bar{J}(t) = e^{-bt} \sum_{n=0}^{N-1} a_n j_n^s(\omega t) , \quad (11)$$

where b is a response damping coefficient, ω is an angular frequency, a_n are the expansion coefficients, and $j_n^s(\omega t)$ is the n -th order modified spherical Bessel functions of the first kind.

Using Eq. (11), accuracy of the extrapolation in the time domain is improved, and the computation time is reduced because fewer terms are needed.

$j_n^s(\omega t)$ functions may be defined, for $n \geq 1$, as:

$$j_n^s(\omega t) = (-1)^n \cdot \frac{t^{n+1}}{k^{n-1}} \cdot \frac{\partial^n}{(t \cdot \Delta t)^n} \left(\frac{j_0^s(\omega t)}{\omega t} \right) , \quad (12)$$

and

$$j_0^s(\omega t) = \sin(\omega t), \quad \text{for } n > 1, \quad (13)$$

where:

$$\frac{\partial^n}{(t \cdot \Delta t)^n} = \underbrace{\frac{1}{t} \cdot \frac{\partial}{\partial t} \left[\frac{1}{t} \cdot \frac{\partial}{\partial t} \left(\frac{1}{t} \cdot \frac{\partial}{\partial t} \dots \right) \right]}_n \quad \text{for } n > 1 . \quad (14)$$

Alternatively, assuming, that $j_0^s(\omega t) = \sin(\omega t)$ and

$$j_1^s(\omega t) = \frac{\sin(\omega t)}{\omega t} - \cos(\omega t) , \quad (15)$$

a recursive formula may be used:

$$j_{n+1}^s(\omega t) = \frac{2n+1}{\omega t} \cdot j_n^s(\omega t) - j_{n-1}^s(\omega t) , \quad (16)$$

to generate the $j_n^s(\omega t)$ functions for $n > 1$.

Where $n = 2$, we have:

$$j_2^s(\omega t) = \frac{3 \sin(\omega t)}{(\omega t)^2} - \frac{3 \cos(\omega t)}{\omega t} + \sin(\omega t) . \quad (17)$$

The new spherical modified Bessel functions of the first type are shown in Fig. 1. With a fixed frequency ω , for large values of the argument, all functions tend to assume the same sinusoidal shape, but have a different phase shift. The major differences between the functions $j_n^1(\omega t)$ are observed with low argument values (Fig. 1). The difference is the greatest for the first period, and then decreases. It may be said that non-stationary functions whose periods differ initially, tend to become steady.

The functions were designed for the purpose of studying transient phenomena in antennas. However, these functions may effectively model other transient processes that are initially not stationary.

Let M be the number of initial late-time domain samples that are given for $J(t)$. The samples of the original data are $J_k = J(t_k) = J(k \cdot \Delta t)$ and a simple set of N unknown weighting coefficients a_n is used for time representation.

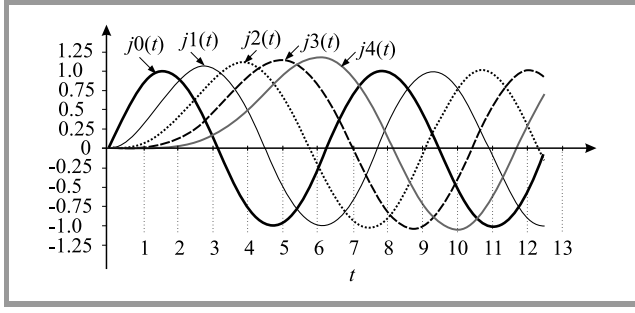


Fig. 1. New modified spherical Bessel functions of the first kind.

These coefficients may be determined by solving a set of linear equations using M samples only.

In a matrix form, the set of equations may be written as:

$$[\Psi_{kj}] \cdot [a_n] = [J_k], \quad (18)$$

where the coefficient matrix components are:

$$\Psi_{kj} = \sum_{i=0}^m j_k^s(\omega t_i) \cdot j_j^s(\omega t_i) \cdot e^{-2bt_i}, \quad (19)$$

$$J_k = \sum_{i=0}^m j_k^s(\omega t_i) \cdot e^{-bt_i} \cdot J(t_i). \quad (20)$$

Matrix $[\Psi_{kj}]$ is symmetric and positively definite, assuring a single-valued solution of the matrix from Eq. (18).

The least squares solution for a_n may be found. By solving Eq. (18), we may obtain the value of the coefficient vector $[a_0, a_1, \dots, a_n]^T$. The reconstruction of signal $\bar{J}(t)$ in late-time may be performed by multiplying the coefficient vector with the respective $j_n^s(kt)$ functions. If the extrapolation is successful, $\bar{J}(t)$ closely approximates $J(t)$, including in the late-time range.

5. Numerical Example

In this section, we present an example in order to verify the technique presented in the previous section. Software has been relied upon to evaluate current distribution on a linear antenna using the time domain electric field integral equation. The antenna is assumed to be perfectly conducting and it was excited with a plane wave with a Gaussian profile in time. The excitation has the following form:

$$E^i(t) = \frac{4E_0}{cT\sqrt{\pi}} e^{-[\frac{t}{T}(t-t_0)]^2}, \quad (21)$$

where E_0 is the amplitude of the incoming wave, t_0 is a delay and T controls the pulse width.

In the presented example, we assumed that: $E_0 = 120 \pi$ V/m, $t_0 = 8$ ns, $T = 6$ ns. The length of the antenna $L = 1$ m, the radius of the antenna $a = 0.002$ m and the number of segments is chosen to be $N_s = 20$. The remaining parameters of Eq. (11) are obtained from the MOT algorithm: damping coefficient $9b = 0.067714$ and angular frequency $\omega = 0.89264$.

The extrapolated time domain response is compared with the time domain response obtained with the use of MOT software. The time step used in MOT is:

$$\Delta t_1 = \frac{\Delta z}{c} = 0.1667 \text{ ns},$$

$$\Delta t_2 = 2 \cdot \frac{\Delta z}{c} = 0.3334 \text{ ns},$$

$$\Delta t_3 = 2.25 \cdot \frac{\Delta z}{c} = 0.3751 \text{ ns},$$

where Δz denotes the length of the segment and $\Delta z = L/N_s$. Moreover, the time domain responses calculated by using MOT are presented in the Figs. 2, 3 and 4.

One may notice that for $\Delta t > \Delta z/c$ the time domain response contains late-time instabilities.

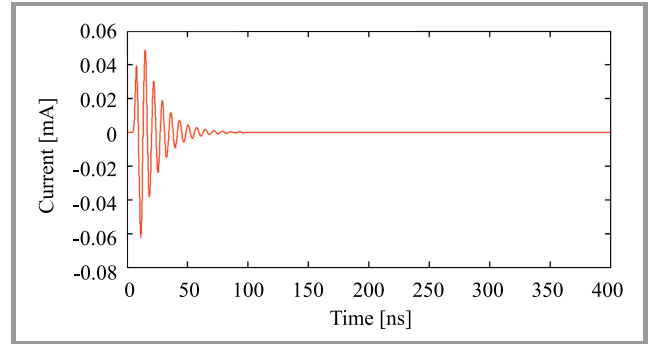


Fig. 2. Time domain response calculated by using MOT and $\Delta t_1 = 0.1667$ ns.

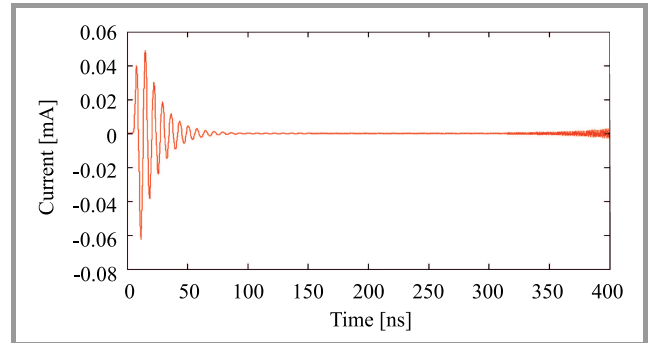


Fig. 3. Time domain response calculated by using MOT and $\Delta t_2 = 0.3334$ ns.

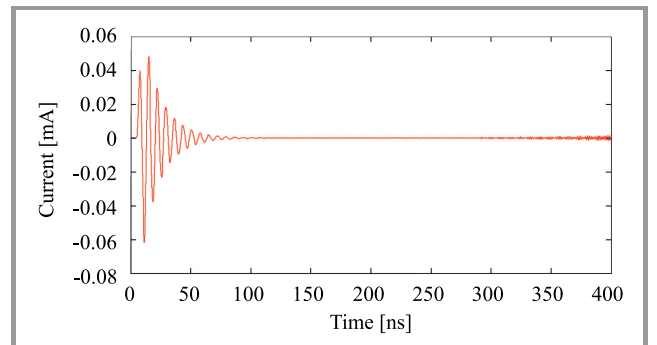


Fig. 4. Time domain response calculated by using MOT and $\Delta t_3 = 0.3751$ ns.

Using the MOT algorithm, the time domain data are obtained from $t = 8$ ns to 112 ns. In this time interval, the time domain responses from the MOT algorithm are identical for time steps $\Delta t > \Delta z/c$.

Assuming that only 8 time-data points from $t = 98.669$ ns to $t = 111.06$ ns with $\Delta t = 1.77$ ns are available and that the order of the polynomial is chosen to be two, only three weighting coefficients a_n (a_0 , a_1 and a_2) are needed.

By solving Eq. (18) using the available data, and then using coefficients a_0 , a_1 and a_2 , the time domain response is extrapolated over the late-time of $t = 98.669$ ns to $t = 400$ ns. The approximation is also effective for a period over $t = 400$ ns.

The result of the computation is:

$$\begin{aligned} \bar{J}(t) = e^{-0.067714 \cdot t} \cdot & \left[-0.244 \cdot \frac{\sin(\omega t)}{t^2} + \right. \\ & + 0.0226 \cdot \frac{\sin(\omega t)}{t} + 0.2175 \cdot \frac{\cos(\omega t)}{t} + \\ & \left. + 0.144 \cdot \sin(\omega t) - 0.02014 \cdot \cos(\omega t) \right], \end{aligned} \quad (22)$$

for $t \in (98.669, \infty)$ and $\omega = 0.89264$.

Using Eq. (22), for the time from $t = 0$ to 98.669 ns, the stable current waveform is calculated numerically by the MOT method, and for times greater than $t = 98.669$ ns – the current waveform is described by Eq. (22). As long as the numerical solution is stable, the numerical solution is used, then approximation by Eq. (22) is performed. In this way, stability is obtained throughout the process considered. Stability of the approximation is illustrated in Fig. 5.

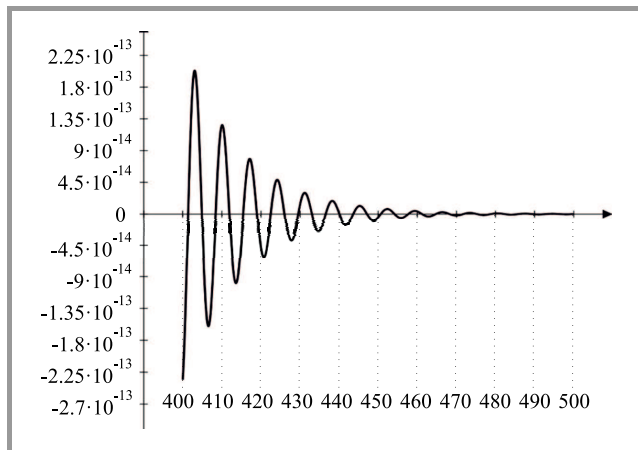


Fig. 5. Illustration of the absolute stability of the Eq. (18).

The extrapolated time domain response is agreeable with the MOT data. The results are satisfactory, as they compare well with the actual solution. The relative errors of the approximation are in the range from 0.0012% to 0.0637%. In this way, by using early-time data, we obtained stable late-time information.

6. Conclusions

In this paper, a technique using stable initial late-time information is relied upon to generate stable information for the entire late-time phase.

The proposed solution offers a considerable advantage in the form of its ability to determine approximating functions of small orders, which approximate and extrapolate data obtained from the stable antenna response with a high degree of accuracy.

For a simple but effective approximation, new special spherical Bessel functions have been introduced. Although the purpose of constructing these functions was to study transient states in antennas, they may be useful for describing other initially non-stationary processes.

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Performance Evaluation of the Threshold Call Admission Policy in Multi-rate Loss Systems

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Abstract—In this paper we consider a link, characterized by specific capacity, that services multi-rate random or quasi-random traffic. Random traffic is generated by an infinite number of traffic sources, while quasi-random traffic is generated by a finite population of traffic sources. The link is modeled as a multi-rate loss system. Handover and new calls are distinguished. New calls compete for the available bandwidth under a threshold call admission policy. In that policy, a new call of a particular service-class is not allowed to enter the system if the in-service handover and new calls of the same service-class plus the new call, exceed a predefined threshold (which can be different for each service-class). On the other hand, handover calls compete for the available bandwidth based on the complete sharing policy. We show that the steady state probabilities in the proposed models have a product form solution (PFS). The PFS leads to a convolution algorithm for accurate calculation of congestion probabilities and link utilization.

Keywords—blocking, congestion, convolution, product form, quasi-random, random.

1. Introduction

A call admission policy is a necessary quality of service (QoS) mechanism in a link that services multi-rate traffic, since it provides access to the resources needed by arriving calls. The complete sharing (CS) policy, also known as a “full accessibility” policy, is considered in the literature as the simplest call admission policy [1], [2]. However, it is important to mention that the CS policy is not designed with the aim of providing a certain QoS to calls of a particular service-class. This is because the only criterion for the acceptance of a new call in a system that adopts the CS policy is the availability of bandwidth required by that call. In that sense, the CS policy is unfair with regard to calls characterized by high bandwidth requirements, since it results in high call blocking probabilities (CBP), e.g. [3]–[12]. Focusing on wireless networks, the CS policy cannot treat handover and new calls of the same service-class differently,

though it is obvious that handover calls should have higher priority compared to new calls. Based on these factors, various other call admission policies have been analyzed in the literature, including the bandwidth reservation (BR) policy (e.g. [13]–[22]), the multiple fractional channel reservation (MFCR) policy (e.g. [23]–[25]) and the threshold (TH) policy, all attempting to prioritize certain types of calls. The BR policy may introduce a simple prioritization mechanism while dealing with calls of different service-classes, by reserving bandwidth units (b.u.) favoring high-speed calls or handover calls. The MFCR policy generalizes the BR policy by allowing the reservation of a real (not an integer) number of b.u. We consider the TH policy, in which in-service calls of a particular service-class cannot exceed a specific threshold, even if resources are available in the system. This policy cannot be described via a particular selection of BR or MFCR parameters and has been applied not only in wired but also in wireless and satellite networks (e.g. [26]–[40]).

To prioritize handover calls over new calls, a variant of the TH policy may be considered, known as the threshold call admission (TCA) policy. In the TCA policy, handover calls are assigned with higher priority than new calls. More specifically, a new call (of a particular service-class) cannot enter the system if the number of in-service handover and new calls of the same service-class, plus the new call, exceeds a certain threshold (which is predefined and which differs for each particular service-class). On the other hand, the TCA policy is not applied to handover calls, i.e., these calls are blocked only if their required bandwidth is not available at the time of their arrival (CS policy). As a possible application of the TCA policy, consider the case of a cloud radio access network which consists of a large number of distributed base stations where the remote radio heads (RRHs), including radio frequency components and antennas, are separated from the baseband signal processing servers, named baseband units [41]. A possible movement of users from the serving RRH to a new RRH may create handover traffic [42]. In that case, the TCA policy may

be applied in the new RRH in order to prioritize handover traffic over traffic created by new users.

In this paper, we consider a link under the TCA policy, which services random or quasi-random multi-rate traffic. Random traffic is generated by an infinite number of calls (handover and new) of different service-classes, while quasi-random traffic is generated by a finite number of calls of different service-classes. We model the link as a multi-rate loss system and show that this loss system may be analytically described via a continuous time and reversible Markov chain, both in the case of random and quasi-random traffic. In both cases, the steady state distribution has a product form solution (PFS). Based on the PFS, a convolution algorithm is proposed for the calculation of congestion probabilities and link utilization. In general, convolution algorithms can be efficiently applied only in PFS models. Their main advantage is the fact that they capture information about the number of in-service calls belonging to each service-class. That information is significant when the TH and TCA policies are considered. On the other hand, in non-PFS models (such as models adopting the BR policy), the application of convolution algorithms for CBP determination may be quite complicated [43]–[46].

This paper is organized as follows. In Section 2, we review the model of [47] which considers a link under the TCA policy that accommodates multi-rate Poisson traffic (TCA model). In Section 3, the corresponding model for finite sources (f-TCA model) is proposed, the PFS is shown and a convolution algorithm for the calculation of congestion probabilities and link utilization is provided. In Section 4, analytical CBP and link utilization results for the TCA model, the CS model of [48] and the TH model of [26] assuming multi-rate Poisson traffic, are presented. Additionally, the analytical congestion probabilities results for the TCA and the f-TCA models are shown. The paper concludes in Section 5. In the Appendix, a tutorial example is presented in order to clarify the necessary calculations for the determination of congestion probabilities, when the convolution algorithm is applied in the f-TCA model.

2. Review of the TCA Model

2.1. Analytical Model

In the model of [47], we consider a link with a fixed capacity of C b.u. and let j express the b.u. occupied in the link ($j = 0, 1, \dots, C$). The link services Poisson traffic generated via K different service-classes. Let the arrival rate of handover and new service-class k ($k = 1, \dots, K$) calls be λ_{kh} and λ_{kn} , respectively. The total arrival rate of service-class k calls is $\lambda_k = \lambda_{kh} + \lambda_{kn}$. Each service-class k (new or handover) call requires b_k b.u. in order to be accepted in the link and has a generally distributed service time with a mean μ_k^{-1} . New and handover calls are accepted in the link based on the TCA and the CS policy, respectively.

In the TCA policy, each service-class k has a predefined threshold $n_{k,\max}$ whose role is to allow the TCA policy to prioritize handover calls against new calls. More precisely, a new call of service-class k will be accepted in the system if:

- 1) $j + b_k \leq C$ (i.e. if there exist available b.u. in the link)
- 2) the number of in-service handover and new calls of service-class k , n_k , plus the new call is at most equal to $n_{k,\max}$, i.e. $n_k + 1 \leq n_{k,\max}$.

The second constraint expresses the fact that a new call may not become accepted for service even if the b.u. it requires are available. On the other hand, a handover service-class k call needs to meet the first constraint only in order to be accepted in the link.

By assuming that the system is in a steady state, let $\mathbf{n} = (n_1, \dots, n_k, \dots, n_K)$ be the steady state vector and $\pi(\mathbf{n})$ be the corresponding probability distribution, which can be expressed by the following PFS [47]:

$$\pi(\mathbf{n}) = G^{-1} \left(\prod_{k=1}^K \frac{p_k^{n_k}}{n_k!} \right), \quad (1)$$

where:

$$\frac{p_k^{n_k}}{n_k!} = \begin{cases} \frac{\alpha_k^{n_k}}{n_k!} & \text{for } n_k \leq n_{k,\max} \\ \frac{\alpha_k^{n_{k,\max}} \alpha_{kh}^{(n_k - n_{k,\max})}}{n_k!} & \text{for } n_k > n_{k,\max} \end{cases},$$

G refers to a constant used for normalization and is expressed with the following formula:

$$G = \sum_{\mathbf{n} \in \Omega} \left(\frac{\prod_{k=1}^K p_k^{n_k}}{n_k!} \right),$$

$\alpha_k = \frac{\lambda_k}{\mu_k}$, $\alpha_{kh} = \frac{\lambda_{kh}}{\mu_k}$ (in Erl) while Ω refers to the state space, $\Omega = \{\mathbf{n} : 0 \leq \mathbf{n} \leq C, k = 1, \dots, K\}$ with $\mathbf{n} = \sum_{k=1}^K n_k b_k$ and $\mathbf{b} = (b_1, \dots, b_K)^T$.

Based on Eq. (1), we see that the term $\frac{\alpha_k^{n_{k,\max}} \alpha_{kh}^{(n_k - n_{k,\max})}}{n_k!}$ expresses the application of the TCA policy in new service-class k calls when $n_k > n_{k,\max}$.

2.2. Convolution Algorithm

According to Eq. (1), the following convolution algorithm can be adopted in the TCA model for the determination of the various performance measures including CBP.

Step 1. Assuming that the link of capacity C services only service-class k ($k = 1, \dots, K$) calls, determine the occupancy distribution $q_k(j)$, where $j = n_k b_k$, as follows:

$$q_k(j) = \begin{cases} q_k(0) \frac{\alpha_k^{n_k}}{n_k!} & \text{for } n_k \leq n_{k,\max} \\ q_k(0) \frac{\alpha_k^{n_{k,\max}} \alpha_{kh}^{(n_k - n_{k,\max})}}{n_k!} & \text{for } n_k > n_{k,\max} \end{cases}. \quad (2)$$

Step 2. Based on a successive convolution of all service-classes excluding service-class k , determine the occupancy distribution $Q_{(-k)}$:

$$Q_{(-k)} = q_1 \cdot \dots \cdot q_{k-1} \cdot \dots \cdot q_{k+1} \cdot \dots \cdot q_K. \quad (3)$$

More precisely, the corresponding convolution operation of service-classes k and r , is given by:

$$q_k \cdot q_r = \left\{ \begin{array}{l} q_k(0)q_r(0), \sum_{x=0}^1 q_k(x)q_r(1-x), \\ \dots, \sum_{x=0}^C q_k(x)q_r(C-x) \end{array} \right\}. \quad (4)$$

Step 3. Compute the convolution of $Q_{(-k)}$ and q_k , from:

$$Q_{(-k)} \cdot q_k = \left\{ \begin{array}{l} Q_{(-k)}q_k(0), \sum_{x=0}^1 Q_{(-k)}(x)q_k(1-x), \\ \dots, \sum_{x=0}^C Q_{(-k)}(x)q_k(C-x) \end{array} \right\}. \quad (5)$$

Via Eq. (5), we can also determine the link occupancy distribution $q(j)$, according to the following formulas:

$$q(0) = \frac{Q_{(-k)}(0)q_k(0)}{G},$$

$$q(j) = \frac{\sum_{x=0}^j Q_{(-k)}(x)q_k(j-x)}{G}, \quad j = 1 \dots, C. \quad (6)$$

Based on Eq. (6), we can determine the CBP of service-class k calls (new or handover), as well as the link utilization (in b.u.) via Eqs. (7), (8) and (9), respectively:

$$B_{kn} = \sum_{j=C-b_k+1}^C q(j) + G^{-1} \sum_{t=n_{k,\max}b_k}^{C-b_k} q_k(t) \sum_{y=t}^{C-b_k} Q_{(-k)}(C-b_k-y), \quad (7)$$

$$B_{kh} = \sum_{j=C-b_k+1}^C q(j), \quad (8)$$

$$U = \sum_{j=1}^C jq(j). \quad (9)$$

On the right hand side of Eq. (7), the first term refers to call blocking due to the unavailability of b.u., while the second term refers to the TCA policy. Indeed, in states $t = n_{k,\max}b_k, \dots, C-b_k$, call blocking occurs (due to the threshold $n_{k,\max}$) although there exist available b.u. for a new service-class k call.

3. f-TCA Model

3.1. Analytical Model

In the proposed f-TCA model, service-class k calls arrive in the link according to a quasi-random process with mean arrival rate $\lambda_{k,\text{fin}} = (N_k - n_k)v_k$, where N_k refers to the finite number of service-class k sources that are capable of

generating traffic, n_k is the number of in-service calls (new and handover), while v_k is the mean call arrival rate per idle service-class k source. Similarly, let v_{kh} and v_{kn} be the mean call arrival rate per idle source of handover and new service-class k , respectively. Then, $v_k = v_{kh} + v_{kn}$. The traffic-load per idle source offered is $\alpha_{k,\text{idle}} = \frac{v_k}{\mu_k}$. Assuming that $N_k \rightarrow \infty$, for $k = 1, \dots, K$, and that the total traffic load offered is constant, then the call arrival process becomes Poisson and we have the TCA model of [47]. Each service-class k call requires b_k b.u. in order to be accepted in the link and has a generally distributed service time with mean μ_k^{-1} . New and handover calls are accepted in the link based on the TCA and the CS policy, respectively.

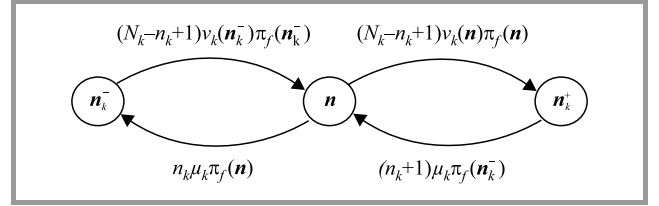


Fig. 1. State transition diagram, for service-class k calls, between states n_k^- , n and n_k^+ .

Similarly to the TCA model, we assume that the system is in a steady state and use $\pi_f(n)$ to denote the probability distribution of $n = (n_1, \dots, n_k, \dots, n_K)$. Figure 1 presents the state transition diagram of the proposed model for service-class k calls. Regarding the global balance equation for state n , it can be expressed as *rate into state n = rate out of state n* :

$$\sum_{k=1}^K (N_k - n_k + 1)v_k(n_k^-)\pi_f(n_k^-) + \sum_{k=1}^K (n_k + 1)\mu_k\pi_f(n_k^+) = \sum_{k=1}^K (N_k - n_k)v_k(n)\pi_f(n) + \sum_{k=1}^K n_k\mu_k\pi_f(n), \quad (10)$$

where: $v_k(n) = \begin{cases} v_k & \text{for } n_k \leq n_{k,\max} \\ v_{kh} & \text{for } n_k > n_{k,\max} \end{cases}$, Ω is the state space

$\Omega = \{n : 0 \leq n_b \leq C, k = 1, \dots, K\}$, $n_k^- = \{n_1, \dots, n_k - 1, \dots, n_K\}$ and $n_k^+ = \{n_1, \dots, n_k + 1, \dots, n_K\}$.

The Markov chain of the proposed loss model retains reversibility and therefore local balance exists between the adjacent states: (1) n_k^- and n and (2) n and n_k^+ . To this end, we can write the following local balance equations (*rate up = rate down*), for $k = 1, \dots, K$ and $n \in \Omega$:

$$(N_k - n_k + 1)v_k(n_k^-)\pi_f(n_k^-) = n_k\mu_k\pi_f(n), \quad (11)$$

$$(N_k - n_k)v_k(n)\pi_f(n) = (n_k + 1)\mu_k\pi_f(n_k^+). \quad (12)$$

The PFS that satisfies Eqs. (10)–(12) is:

$$\pi_f(n) = G^{-1} \left[\prod_{k=1}^K \binom{N_k}{n_k} p_{k,\text{idle}}^{n_k} \right], \quad (13)$$

where:

$$p_{k,\text{idle}}^{n_k} = \begin{cases} \alpha_{k,\text{idle}}^{n_k} & \text{for } n_k \leq n_{k,\max} \\ \alpha_{k,\text{idle}}^{n_{k,\max}} \alpha_{kh,\text{idle}}^{(n_k - n_{k,\max})} & \text{for } n_k > n_{k,\max} \end{cases},$$

$$\alpha_{k,\text{idle}} = \frac{v_k}{\mu_k}, \quad \alpha_{kh,\text{idle}} = \frac{v_{kh}}{\mu_k} \text{ (in Erl)},$$

$$G = \sum_{n \in \Omega} \left(\prod_{k=1}^K \binom{N_k}{n_k} p_{k,\text{idle}}^{n_k} \right).$$

Based on Eq. (13), we see that the term $\alpha_{k,\text{idle}}^{n_{k,\text{max}}} \alpha_{kh,\text{idle}}^{(n_k - n_{k,\text{max}})}$ expresses the application of the TCA policy in new service-class k calls when $n_k > n_{k,\text{max}}$.

3.2. Convolution Algorithm

Based on (13), we can modify the convolution algorithm from Section 2 as follows:

Step 1. Assuming that the link of capacity C services only service-class k ($k = 1, \dots, K$) calls, determine the occupancy distribution $q_{f,k}(j)$, where $j = n_k b_k$, as:

$$q_{f,k}(j) = \begin{cases} q_{f,k}(0) \binom{N_k}{n_k} \alpha_{k,\text{idle}}^{n_k} & \text{for } n_k \leq n_{k,\text{max}} \\ q_{f,k}(0) \binom{N_k}{n_k} \alpha_{k,\text{idle}}^{n_{k,\text{max}}} \alpha_{kh,\text{idle}}^{(n_k - n_{k,\text{max}})} & \text{for } n_k > n_{k,\text{max}} \end{cases} \quad (14)$$

Step 2. Based on a successive convolution of all service classes excluding service-class k , determine the occupancy distribution $Q_{f,(-k)}$:

$$Q_{f,(-k)} = q_{f,1} \cdot \dots \cdot q_{f,k-1} \cdot \dots \cdot q_{f,k+1} \cdot \dots \cdot q_{f,K}. \quad (15)$$

Step 3. Compute the convolution operation of $Q_{f,(-k)}$ and $q_{f,k}$ as:

$$Q_{f,(-k)} \cdot q_{f,k} = \left\{ \begin{array}{l} Q_{f,(-k)} q_{f,k}(0), \sum_{x=0}^1 Q_{f,(-k)}(x) q_{f,k}(1-x), \\ \dots, \sum_{x=0}^C Q_{f,(-k)}(x) q_{f,k}(C-x) \end{array} \right\}. \quad (16)$$

Via Eq. (16), we can determine the link occupancy distribution $q_f(j)$, according to:

$$q_f(0) = \frac{Q_{f,(-k)}(0) q_{f,k}(0)}{G},$$

$$q_f(j) = \frac{\sum_{x=0}^j Q_{f,(-k)}(x) q_{f,k}(j-x)}{G}, \quad j = 1, \dots, C. \quad (17)$$

Based on Eq. (17), we can determine the time congestion probabilities of service-class k calls (new or handover) as well as the link utilization (in b.u.) via Eqs. (7), (8) and (9), respectively, where $q(j)$ and $Q_{(-k)}$ should be replaced by $q_f(j)$ and $Q_{f,(-k)}$. To calculate call congestion probabilities for service-class k calls, we should run the convolution algorithm for $N_k - 1$ sources.

4. Evaluation

In this section, we present analytical CBP results for an application example. Due to the fact that the TCA and the

f-TCA loss models have a PFS, we do not present simulation results.

Consider a link of a fixed capacity of $C = 80$ b.u. that accommodates Poisson arriving calls of $K = 3$ service-classes, with the traffic characteristics shown in Table 1. Based on Table 1, the average traffic-load offered per b.u. equals $\frac{\alpha_1 + \alpha_2 + \alpha_3}{C} = 0.16875$.

Table 1

Traffic description parameters of the three service-classes

Service class	Traffic load (Erl)	Bandwidth per call (b.u.)	Threshold (no. of calls)
1st	$a_1 = 9.0$	$b_1 = 1$	$n_{1,\text{max}} = 48$
2nd	$a_2 = 3.0$	$b_2 = 5$	$n_{2,\text{max}} = 13$
3rd	$a_3 = 1.5$	$b_3 = 12$	$n_{3,\text{max}} = 4$

The provided analytical CBP and link utilization results for the TCA model and the TH model of [26], are based on two different values of $n_{3,\text{max}} = 4$ and 3 calls. As a reference, the analytical CBP and link utilization results for the CS model of [48] also are given. Regarding the f-TCA model, we provide analytical time congestion probabilities results when $n_{3,\text{max}} = 4$ calls and consider two different cases for the traffic sources cases: (1) $N_1 = N_2 = N_3 = 100$ sources and (2) $N_1 = N_2 = N_3 = 200$ sources.

In the x -axis of Figs. 2–5, the offered traffic-load of the 1st, 2nd and 3rd service-class increases in steps of 1.0, 0.5 and 0.25 Erl, respectively. So, point 1 refers to: $(\alpha_1, \alpha_2, \alpha_3) = (9.0, 3.0, 1.5)$, while point 7 to: $(\alpha_1, \alpha_2, \alpha_3) = (15.0, 6.0, 3.0)$. The values of the offered traffic-load of service-class k handover calls, required in the case of the TCA model, are given by the expression $\alpha_{kh} = \frac{\alpha_k}{2}$. In the x -axis of Figs. 6–8, which refer to the

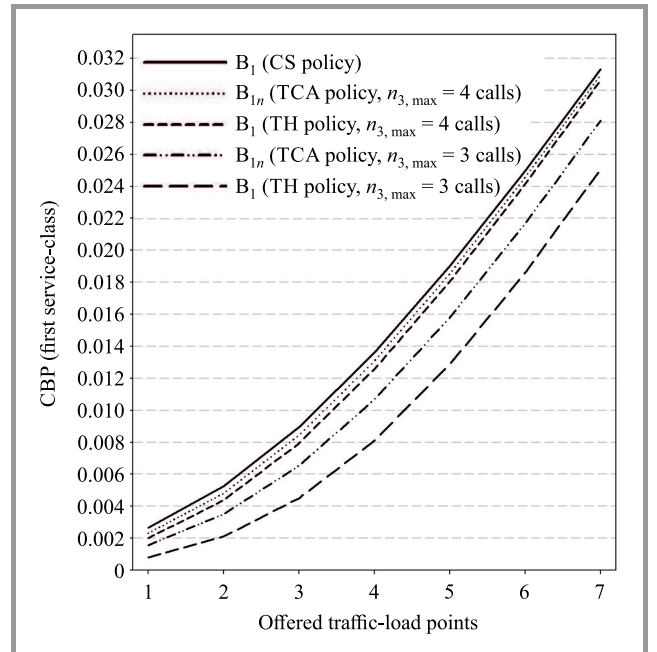


Fig. 2. CBP – first service-class.

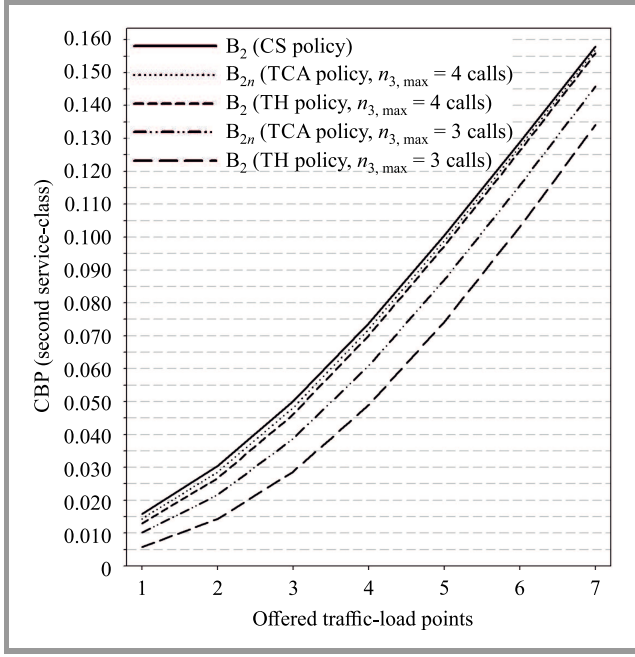


Fig. 3. CBP – second service-class.

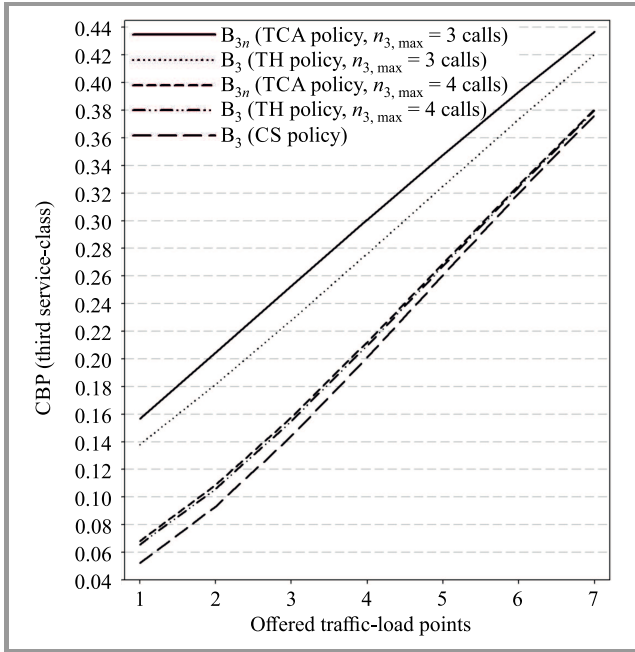


Fig. 4. CBP – third service-class.

f-TCA model, the values of the offered traffic-load per idle source of service-class k handover calls, are given by the expression:

$$\alpha_{kh, \text{idle}} = \frac{\alpha_{k, \text{idle}}}{2},$$

where $\alpha_{k, \text{idle}} = \frac{\alpha_k}{N_k}$.

Based on Figs. 2–4, we observe that:

- The TCA model affects the CBP of all service-classes. Decreasing the value of $n_{3, \text{max}}$ from 4 to 3 calls,

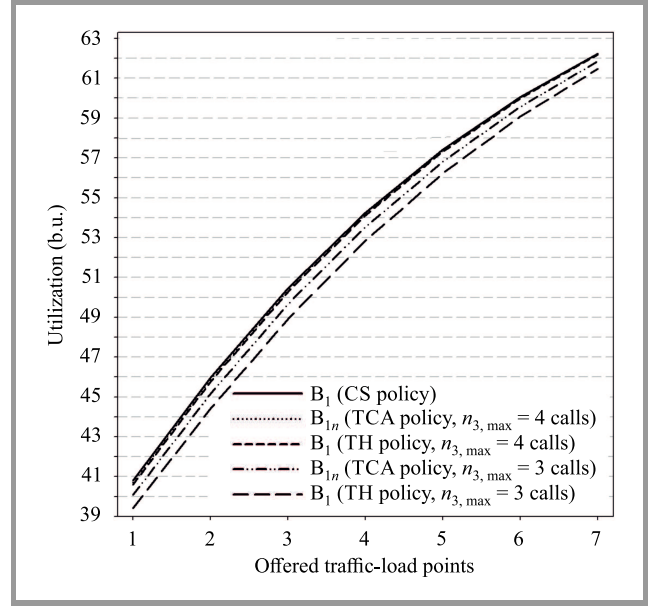


Fig. 5. Link utilization.

results in a decrease in the CBP of 1st and 2nd service-class calls, and in an increase in the CBP of 3rd service-class calls. This result is expected, since fewer in-service 3rd service-class calls are permitted in the system and, therefore, the available link bandwidth may be occupied by more calls of the other service-classes.

- The existing policies (CS and TH) cannot approximate the CBP results obtained via the TCA model, unless $n_{k, \text{max}}$ becomes quite high. In that case, the TCA policy does not actually affect CBP and behaves similarly to the CS policy.

Based on Fig. 5, we observe that the CS policy achieves a higher link utilization compared to the TCA and the TH policies. The TCA policy achieves a slightly better link utilization compared to the corresponding TH policy due to the fact that it is applied only to new calls of a given service-class.

In Figs. 6–8, we present analytical time congestion probabilities for the proposed models (TCA and f-TCA) and for each service-class, respectively. We observe that the f-TCA model results in lower time congestion probabilities compared to the TCA model, especially when the number of traffic sources is low ($N_1 = N_2 = N_3 = 100$ sources). As the number of traffic sources increases, the results of the f-TCA model become similar to those of the TCA model. To summarize the results, the following facts need to be emphasized:

- The TCA policy is applied only to new calls of a given service-class, while the TH policy is applied to both new and handover calls. On the other hand, the CS policy does not introduce restrictions to calls (apart from the link capacity). In that sense, the TCA policy cannot be approximated by the TH or the CS

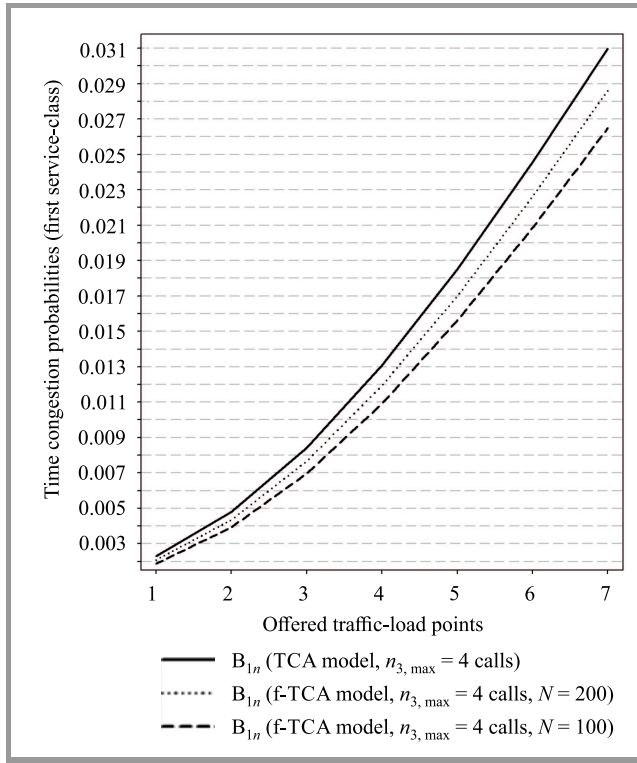


Fig. 6. Time congestion probabilities— first service-class.

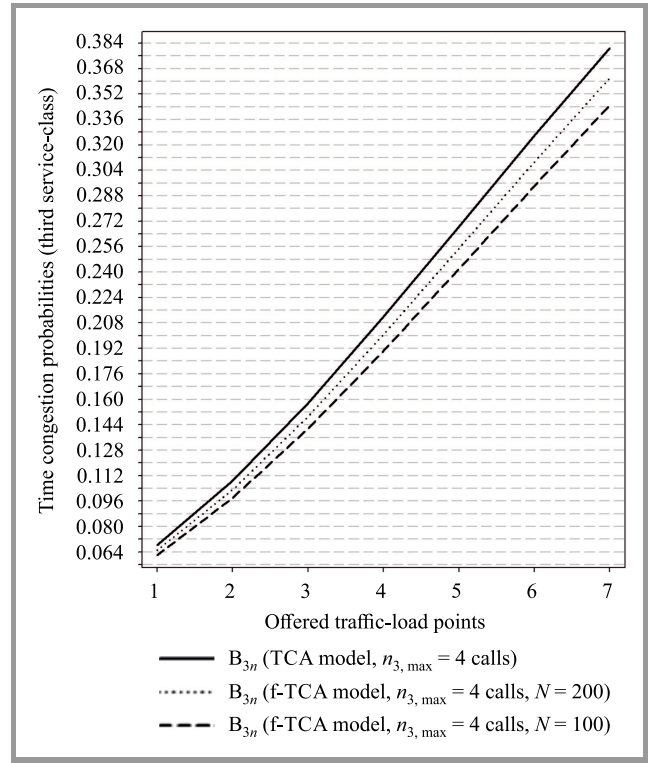


Fig. 8. Time congestion probabilities – third service-class.

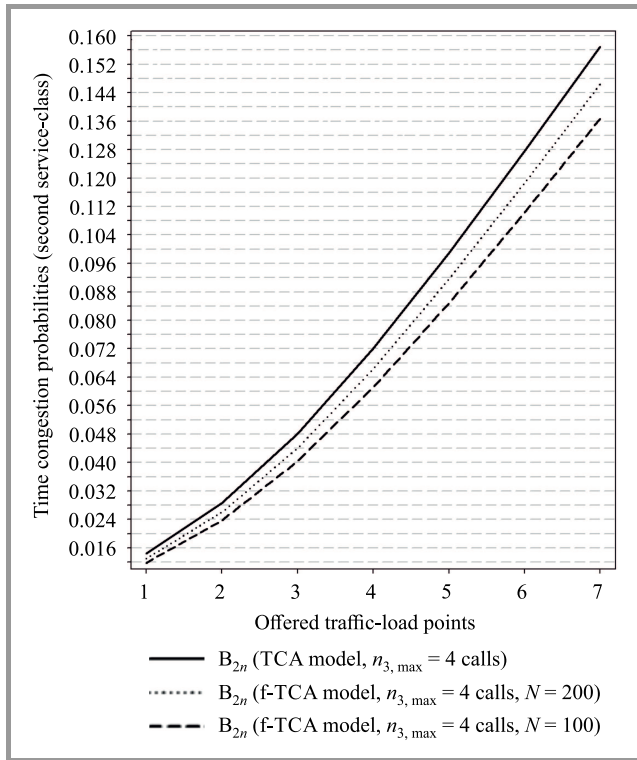


Fig. 7. Time congestion probabilities – second service-class.

policy (see Figs. 2–4) unless $n_{k,\max}$ becomes quite high. In that case the threshold does not affect the call admission control and the TCA policy behaves as the CS policy.

- The consideration of quasi-random traffic results in lower congestion probabilities compared to Poisson traffic, since the number of sources that generate traffic is limited (see Figs. 6–8).

5. Conclusion

We analyze, with the aid of reversible Markov chains, a multi-rate loss system of a single link that services random or quasi-random traffic under the TCA policy. The latter may be adopted when handover calls should be prioritized over new calls of a particular service-class. The TCA policy leads to a PFS for the steady state distribution. Based on the PFS, we can determine time and call congestion probabilities, as well as link utilization via convolution algorithms. As a future work, we intend to study the TCA policy in a multi-rate loss system that carries Pascal traffic. The latter is used to describe incoming traffic that is more bursty than classic random (Poisson) traffic.

Appendix

Tutorial

We consider calls of two different service-classes, accommodated in a link with a fixed capacity of $C = 4$ b.u. under the TCA policy. The arriving calls of both service-classes follow a quasi-random process with the following arrival rates: $v_1 = 0.1$, $v_{1h} = 0.05$, $v_2 = 0.05$ and $v_{2h} = 0.025$. Also let the number of traffic sources be $N_1 = N_2 = 10$. As far

as the bandwidth and the service time requirements of both service-classes are concerned, we have: $b_1 = 1$, $b_2 = 2$, $\mu_1 = \mu_2 = 1.0$. The corresponding arrival rates in the case of the Poisson process are: $\lambda_1 = 1.0$, $\lambda_{1h} = 0.5$, $\lambda_2 = 0.5$, $\lambda_{2h} = 0.25$.

Regarding the TCA policy, let $n_{1,\max} = n_{2,\max} = 1$. The state space Ω of this example consists of 9 states presented in Table 2, in the form of $\mathbf{n} = (n_1, n_2)$, together with the corresponding values of j and the blocking states of both service-classes. More precisely, the symbol “*” expresses a blocking event due to unavailability of b.u., while the symbol “#” expresses a blocking event due to the TCA policy (applied to new calls only).

Table 2
State space (n_1, n_2) , j and B_{1n} , B_{1h} , B_{2n} , B_{2h}

n_1	n_2	j	B_{1n}	B_{1h}	B_{2n}	B_{2h}
0	0	0				
0	1	2			#	
0	2	4	*	*	*	*
1	0	1	#			
1	1	3	#		*	*
2	0	2	#			
2	1	4	*	*	*	*
3	0	3	#		*	*
4	0	4	*	*	*	*

In what follows, we present congestion probability calculations by applying the convolution algorithm from Section 3. In the parenthesis, we show the corresponding results assuming Poisson traffic.

Step 1:

- Determine $q_{f,1}(j)$, $j = 1, \dots, C$.

Let $q_{f,1}(0) = 1$ and determine $q_{f,1}(j)$'s based on $q_{f,1}(0)$:

$$\begin{aligned}
 j = 1 &\rightarrow q_{f,1}(1) = q_{f,1}(0) \times \binom{10}{1} \alpha_{1,\text{idle}} \\
 &\Rightarrow q_{f,1}(1) = 1, \quad (q_1(1) = 1), \\
 j = 2 &\rightarrow q_{f,1}(2) = q_{f,1}(0) \times \binom{10}{2} \alpha_{1,\text{idle}} \times \alpha_{1h,\text{idle}} \\
 &\Rightarrow q_{f,1}(2) = 0.225, \quad (q_1(2) = 0.25), \\
 j = 3 &\rightarrow q_{f,1}(3) = q_{f,1}(0) \times \binom{10}{3} \alpha_{1,\text{idle}} \times \alpha_{1h,\text{idle}}^2 \\
 &\Rightarrow q_{f,1}(3) = 0.03, \quad (q_1(3) = 0.25/6), \\
 j = 4 &\rightarrow q_{f,1}(4) = q_{f,1}(0) \times \binom{10}{4} \alpha_{1,\text{idle}} \times \alpha_{1h,\text{idle}}^3 \\
 &\Rightarrow q_{f,1}(4) = 0.002625, \quad (q_1(4) = 0.125/24).
 \end{aligned}$$

Since the normalization constant is $G = 2.257625$, the normalized values of $q_{f,1}(j)$'s are:

$$\begin{aligned}
 q_{f,1}(0) &= q_{f,1}(1) = 0.44294335, \\
 q_{f,1}(2) &= 0.09966225, \\
 q_{f,1}(3) &= 0.0132883, \\
 q_{f,1}(4) &= 0.001162726.
 \end{aligned}$$

The corresponding values assuming Poisson traffic (and since $G = 2.296875$) are as follows:

$$\begin{aligned}
 q_1(0) &= q_1(1) = 0.435374, \\
 q_1(2) &= 0.108844, \\
 q_1(3) &= 0.0181405, \\
 q_1(4) &= 0.0022675.
 \end{aligned}$$

- Determine $q_{f,2}(j)$, $j = 1, \dots, C$.

Similarly, the normalized values of $q_{f,2}(j)$'s are (where $G = 1.55625$):

$$\begin{aligned}
 q_{f,2}(0) &= 0.6425703, \\
 q_{f,2}(2) &= 0.3212851, \\
 q_{f,2}(4) &= 0.0361445.
 \end{aligned}$$

The corresponding normalized values in the case of Poisson traffic are the following (where $G = 1.5625$):

$$q_2(0) = 0.64, \quad q_2(2) = 0.32, \quad q_2(4) = 0.04.$$

Step 2:

Determine the values of $Q_{f,(-k)}$. Since we have two service-classes this step is not required.

Step 3:

Determine $q_f(j)$'s and consequently the time congestion probabilities (or the CBP in the case of Poisson traffic) of all service-classes.

$$\begin{aligned}
 j = 0 &\rightarrow q_f(0) = q_{f,1}(0)q_{f,2}(0) = 0.2846222, \\
 &\quad (q(0) = 0.2786394), \\
 j = 1 &\rightarrow q_f(1) = \sum_{x=0}^1 q_{f,1}(x)q_{f,2}(1-x) = 0.2846222, \\
 &\quad (q(1) = 0.2786394), \\
 j = 2 &\rightarrow q_f(2) = \sum_{x=0}^2 q_{f,1}(x)q_{f,2}(2-x) = 0.2063511, \\
 &\quad (q(2) = 0.2089795), \\
 j = 3 &\rightarrow q_f(3) = \sum_{x=0}^3 q_{f,1}(x)q_{f,2}(3-x) = 0.1508498, \\
 &\quad (q(3) = 0.1509297), \\
 j = 4 &\rightarrow q_f(4) = \sum_{x=0}^4 q_{f,1}(x)q_{f,2}(4-x) = 0.0487771, \\
 &\quad (q(4) = 0.0536961).
 \end{aligned}$$

The normalized values of $q_f(j)$'s are (since $G = 0.975222$):

$$\begin{aligned}
 q_f(0) &= q_f(1) = 0.291854, \\
 q_f(2) &= 0.211594, \\
 q_f(3) &= 0.154682, \\
 q_f(4) &= 0.050016
 \end{aligned}$$

The corresponding normalized values of $q(j)$'s in the case of Poisson traffic are (since $G = 0.970884$):

$$\begin{aligned} q(0) &= q(1) = 0.286996, \\ q(2) &= 0.215246, \\ q(3) &= 0.155456, \\ q(4) &= 0.055306. \end{aligned}$$

The time congestion probabilities are:

$$\begin{aligned} B_{1n} &= \sum_{j=C-b_1+1}^C q_f(j) + \frac{1}{G} \sum_{t=n_1, \max b_1}^{C-b_1} q_{f,1}(t) \sum_{y=t}^{C-b_1} Q_{f,(-1)}(C-b_1-y) \Rightarrow \\ B_{1n} &= \frac{q_f(4) + q_{f,1}(1)[q_{f,2}(2) + q_{f,2}(0)]}{G} \\ &\quad + \frac{q_{f,1}(2)q_{f,2}(0)/G + q_{f,1}(3)q_{f,2}(0)}{G} \xrightarrow{G=0.975222} \\ B_{1n} &= 0.56222, \\ B_{1h} &= \sum_{j=C-b_1+1}^C q_f(j) = 0.050016 \\ B_{2n} &= \sum_{j=C-b_2+1}^C q_f(j) + \frac{1}{G} \sum_{t=n_2, \max b_2}^{C-b_2} q_{f,2}(t) \sum_{y=t}^{C-b_2} Q_{f,(-2)}(C-b_2-y) \Rightarrow \\ B_{2n} &= \frac{q_f(3) + q_f(4) + q_{f,2}(2)q_{f,1}(0)}{G} \xrightarrow{G=0.975222} \\ B_{2n} &= 0.350626, \\ B_{2h} &= \sum_{j=C-b_2+1}^C q_f(j) = 0.204698. \end{aligned}$$

The corresponding values in the case of Poisson traffic are slightly higher: $B_{1n} = 0.56951$, $B_{1h} = 0.055306$, $B_{2n} = 0.35426$, $B_{2h} = 0.21076$.

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Performance Evaluation of Two-Link Multirate Loss Models with Restricted Accessibility

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Abstract—We consider a two-link communication system with restricted accessibility that services Poisson arriving calls of many service-classes and propose a multirate teletraffic loss model for its analysis. In a restricted accessibility system, call blocking occurs even if available resources do exist at the time of a call's arrival. In the two-link system under consideration, each link has two thresholds (offloading and support) which express the in-service calls in a link. The offloading threshold represents the point from which a link offloads calls. The support threshold (which is lower than the offloading threshold) defines the point up to which a link supports offloaded calls. The two-link system with restricted accessibility is modeled as a loss system whose steady state probabilities do not have a product form solution. However, approximate formulas for the determination of call blocking probabilities are proposed. In addition, we also provide a corresponding analysis related to the case of quasi-random traffic (i.e. traffic generated by a finite number of users). The accuracy of all formulas is verified through simulation and is found to be quite satisfactory.

Keywords—accessibility, blocking, non-product form, Poisson, quasi-random, threshold.

1. Introduction

Bandwidth sharing policies are quality of service (QoS) guarantee mechanisms that are necessary for the provision of bandwidth required by calls in a communication link. Assuming that the link is modeled as a loss system carrying call-level traffic, the most common bandwidth sharing policy is the complete sharing (CS) policy. In the CS policy, a new call is blocked if its required bandwidth units (b.u.) exceed the link's available b.u. In addition to the "CS policy" term adopted in this paper, other equivalent terms are also used in the literature, such as "full accessibility" or "full availability" [1], [2]. The latter, however, usually refers to the proportion of time over which the link is available [3].

The simplest loss system that adopts the CS policy is the Erlang loss system [4]. In this system, with its analysis based

on the Erlang loss model, new calls follow a Poisson process, require one b.u. in order to be accepted by the system and have a generally distributed service time. Call blocking occurs if all b.u. are occupied at the time arrival of a given call. The fact that call blocking probabilities (CBP) are determined according to the Erlang B formula has led to an extensive amount of Erlang loss model extensions for the call-level analysis of wired (e.g. [5]–[20]), wireless (e.g. [21]–[33]), satellite (e.g. [34]–[36]) and optical networks (e.g. [37]–[43]).

In [29], a two-link loss system servicing Poisson traffic is considered and studied. Arriving calls belong to a single service-class and each call requests a single b.u. in order to be accepted in a link. Each link may service calls offloaded from the other link. An offloaded call is a new call that arrives in a link but will be served by the other link, subject to bandwidth availability. This offloading mechanism operates with the aid of a high and a low threshold per link, expressing the number of calls serviced by each link. The high threshold is the offloading threshold, while the low threshold is the support threshold (see Fig. 1). The latter expresses the point up to which the link is capable of supporting offloaded calls (from the other link). The offloading threshold determines the point from which call offloading between the two links may start.

Due to the offloading mechanism, there is no local balance (LB) between adjacent states and, therefore, the steady state probabilities of this system do not have a product form solution (PFS) (see the tutorial example of [44]). Thus, the CBP determination can be based either on the accurate but complex method of solving a set of linear global balance (GB) equations, or on an approximate but efficient method that relies on the Erlang B formula and on the assumption (approximation) that the links are independent.

Such an offloading scheme may find a potential application in mobile/Wi-Fi networks. To manage the increasing traffic in mobile networks, traffic may be offloaded to Wi-Fi networks [45], [46]. In order to increase the bandwidth of Wi-Fi access links, recent research focuses on bandwidth

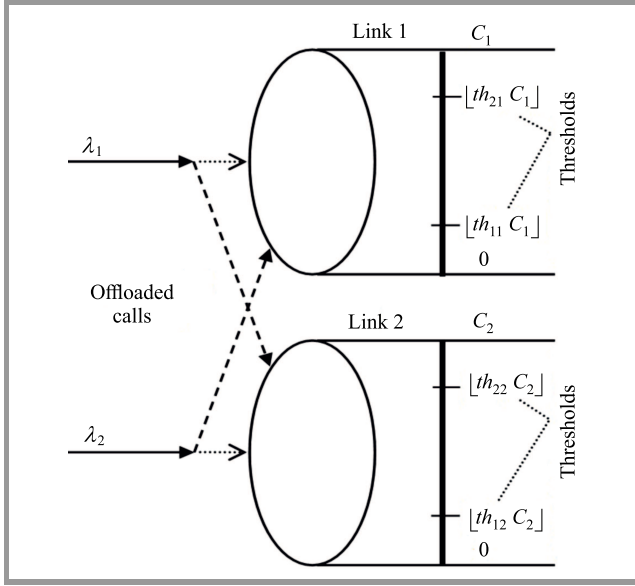


Fig. 1. The two-link loss system.

sharing policies that should be adopted and on the aggregation of backhaul access link capacities. The impact that such aggregation exerts on CBP may be studied via the single-rate model of [29].

In this paper, we extend the model of [29] by considering that the system accommodates Poisson arriving calls of numerous service-classes. Nowadays, this consideration is a sine-qua-non condition in multidimensional network traffic environments. In addition, in the proposed new model we incorporate the notion of restricted accessibility – not only in the case of Poisson traffic, but also in the case of quasi-random traffic (traffic generated by a finite number of users). In a restricted accessibility system, call blocking may occur even if b.u. are available at the time of arrival of a given call. The term “restricted accessibility” covers the following:

- Bandwidth sharing policies, such as the bandwidth (trunk) reservation policy [5], [10], [14], the threshold policy [16], [18] or the probabilistic threshold policy [30], [47]. In the bandwidth reservation policy, call blocking can occur if the available b.u. of the system are reserved at the time of an arrival of a call. In the threshold and the probabilistic threshold policies, a predefined threshold (different for each service-class) is set in order to express the number of in-service calls (of each service-class). If the acceptance of a new call leads to a value that is above that threshold, then call blocking always occurs (threshold policy) or it occurs with a certain probability (probabilistic threshold policy).
- The case where each state of the system (excluding the state where there are no calls in the system) is associated with a blocking probability. Such an approach may be useful when modeling interference between neighboring cells (e.g. in CDMA sys-

tems) [3], [48]. In this paper, we focus on this type of restricted accessibility and propose an approximate method for the CBP calculation which is verified via simulation and is found to be quite satisfactory. The CBP calculation in the proposed two-link model is based on the Erlang multirate loss model (EMLM) [49], [50] which refers to a link that services Poisson traffic generated from different service-classes.

In the remainder of this paper, in Section 2, we review the model of [29]. In Section 3, we propose the extension of [29] which includes the case of many service-classes, as well as the notion of restricted accessibility. In Section 4, we present the corresponding analytical model for the case of multirate quasi-random traffic. In Section 5, we provide analytical and simulated CBP results for the proposed model, assuming the existence of Poisson traffic. We conclude in Section 6.

2. Review of the Two-link Single-rate System

We consider a two-link system of capacities C_1 and C_2 b.u. Each link services single-rate Poisson traffic. Arriving calls require one b.u. in order to be accepted in the system and have a generally distributed service-time with a mean of μ^{-1} . Let λ_l be the call arrival rate in link l ($l = 1, 2$) and let j_l be the occupied b.u. in link l . Then, $0 \leq j_1 \leq C_1$ and $0 \leq j_2 \leq C_2$. Note that j_l expresses the number of in-service calls in link l , since each call requires one b.u.

Each link l ($l = 1, 2$) has a support (low) threshold th_{1l} and an offloading (high) threshold th_{2l} , with $th_{1l} < th_{2l}$ and $0 \leq th_{1l}, th_{2l} \leq 1$. By denoting the largest integer not exceeding x as $\lfloor x \rfloor$ and based on Fig. 1, the role of these thresholds in link l is described in the following manner:

- If $0 \leq j_l < \lfloor th_{1l} C_l \rfloor$, then link l is in a *support mode* of operation. In that mode, the link can service new calls that arrive in link l and offloaded calls from link m ($m = 1, 2, m \neq l$).
- If $\lfloor th_{1l} C_l \rfloor \leq j_l < \lfloor th_{2l} C_l \rfloor$, then link l is in a *normal mode* of operation. In that mode, the link does not service offloaded calls from the other link.
- If $\lfloor th_{2l} C_l \rfloor \leq j_l$, then link l operates in an *offloading mode*. In that mode, a new call that initially arrives in link l is offloaded to link m . If that link is in *support mode* (i.e. $0 \leq j_m < \lfloor th_{1m} C_m \rfloor$), then the call is accepted in link m . Otherwise, if $j_l \leq C_l - 1$, the call is accepted in link l , whereas if $j_l > C_l - 1$, the call is blocked and lost.

Based on the above description, the call admission mechanism applicable to a call that arrives in link l ($l = 1, 2$) consists of two steps:

1. If $(0 \leq j_l < \lfloor th_{2l}C_l \rfloor)$, then the call is serviced via link l .
2. If $\lfloor th_{2l}C_l \rfloor \leq j_l$, then:
 - If $0 \leq j_m < \lfloor th_{1m}C_m \rfloor$, the call is offloaded to link m .
 - If $\lfloor th_{1m}C_m \rfloor \leq j_m$, then link m does not support offloaded calls from link l since it operates in *normal mode*. In that case, the call will be handled by link l . Thus, if $j_l \leq C_l - 1$, the call is accepted in link l . Otherwise, call blocking occurs.

Due to the *offloading* and *support modes* of the two links, there is no LB between adjacent system states and, therefore, the steady state distribution, $P(j) = P(j_1, j_2)$, of such a system cannot be described by a PFS. To determine $P(j_1, j_2)$, two methods exist in the literature.

The first method provides accurate CBP results (compared to simulation results) but is quite complex, since it requires the solution of a set of linear GB equations for each state $j = (j_1, j_2)$ expressed as *rate into state j = rate out of state j*:

$$\begin{aligned} & \lambda_1(j_1 - 1, j_2)P(j_1 - 1, j_2) + \lambda_2(j_1, j_2 - 1)P(j_1, j_2 - 1) \\ & + (j_1 + 1)\mu P(j_1 + 1, j_2) + (j_2 + 1)\mu P(j_1, j_2 + 1) \\ & = \lambda_1(j_1, j_2)P(j_1, j_2) + \lambda_2(j_1, j_2)P(j_1, j_2) \\ & + (j_1\mu + j_2\mu)P(j_1, j_2), \quad (1) \end{aligned}$$

where:

$$l = 1, 2, \quad m \neq l$$

and

$$\lambda_l(j_1, j_2) = \begin{cases} \lambda_l + \lambda_m & \text{if } (j_l < \lfloor th_{1l}C_l \rfloor) \cap (j_m \geq \lfloor th_{2m}C_m \rfloor) \\ 0 & \text{if } (j_l \geq \lfloor th_{2l}C_l \rfloor) \cap (j_m < \lfloor th_{1m}C_m \rfloor) \\ 0 & \text{if } (j_1, j_2) \text{ is a boundary state} \\ \lambda_l & \text{otherwise} \end{cases} \quad (2)$$

Having obtained $P(j_1, j_2)$, we can determine CBP in each link, P'_{b_1} and P'_{b_2} via Eqs. (3) and (4), respectively [29]:

$$P'_{b_1} = \sum_{j_2=\lfloor th_{12}C_2 \rfloor}^{C_2} P(C_1, j_2), \quad (3)$$

$$P'_{b_2} = \sum_{j_1=\lfloor th_{11}C_1 \rfloor}^{C_1} P(j_1, C_2). \quad (4)$$

Equation (3) expresses the fact that call blocking occurs in the first link if all b.u. are occupied (i.e. if $j_1 = C_1$) and, at the same time, the other link does not operate in the support mode (i.e. if $\lfloor th_{12}C_2 \rfloor \leq j_2$). Equation (4) may be interpreted accordingly.

To determine CBP in the system of [29], the following weighted summation can be used:

$$P'_b = \frac{\lambda_1}{\lambda_1 + \lambda_2} P'_{b_1} + \frac{\lambda_2}{\lambda_1 + \lambda_2} P'_{b_2}. \quad (5)$$

Contrary to the first method, the second method is simpler but provides approximate CBP results. The approximation lies on the fact that each link l is modeled as an independent Erlang loss system of capacity C_l ($l = 1, 2$).

The CBP in each link may be approximated by Eqs. (6) and (7), respectively:

$$P_{b_1} = P_1(C_1) P_2(j_2 \geq \lfloor th_{12}C_2 \rfloor), \quad (6)$$

$$P_{b_2} = P_2(C_2) P_1(j_1 \geq \lfloor th_{11}C_1 \rfloor), \quad (7)$$

where $P_l(C_l)$ is the CBP in link l ($l = 1, 2$).

The values of $P_l(C_l)$ in Eqs. (6) and (7) may be determined via the Erlang B formula (see Eq. (8a) for the closed form or Eq. (8b) for the recurrent form):

$$P_l(C_l) = \frac{\frac{a_l^{C_l}}{C_l!}}{\sum_{i=0}^{C_l} \frac{a_l^i}{i!}}, \quad a_l = \frac{\lambda_l}{\mu}, \quad (8a)$$

$$P_l(C_l) = \frac{a_l P_l(C_l - 1)}{C_l + a_l P_l(C_l - 1)}, \quad C_l \geq 1, \quad P_l(0) = 1. \quad (8b)$$

As far as the values of $P_l(j_l \geq \lfloor th_{1l}C_l \rfloor)$ in Eqs. (6) and (7) are concerned, they can be determined by:

$$P_l(j_l \geq \lfloor th_{1l}C_l \rfloor) = \sum_{j_l=\lfloor th_{1l}C_l \rfloor}^{C_l} P_l(j_l), \quad (9)$$

where $P_l(j_l)$ is calculated according to the truncated Poisson distribution:

$$P_l(j_l) = \frac{\frac{a_l^{j_l}}{j_l!}}{\sum_{i=0}^{C_l} \frac{a_l^i}{i!}}, \quad a_l = \frac{\lambda_l}{\mu}. \quad (10)$$

As far as the total blocking probability in the two-link system is concerned, it can be determined via Eq. (5), where P'_{b_1} and P'_{b_2} are replaced by P_{b_1} and P_{b_2} , determined in Eqs. (6) and (7), respectively.

An additional recursive way for the determination of $P_l(j_l)$, $j_l = 1, \dots, C_l$ is based on the link independence assumption. In the Erlang loss model, used to describe each link l , there exist the following LB between states $j_l - 1$ and j_l [44]:

$$j_l P'_l(j_l) = a_l P'_l(j_l - 1). \quad (11)$$

Based on Eq. (11), we can determine the unnormalized values of $P'_l(j_l)$'s considering an initial value of $P'_l(0) = 1$. Then, the normalized values of $P'_l(j_l)$'s are given by:

$$P_l(j_l) = \frac{P'_l(j_l)}{\sum_{x=0}^{C_l} P'_l(x)}. \quad (12)$$

Based on Eq. (12), we can compute P_{b_1} , P_{b_2} and, consequently, the total CBP, via Eqs. (6), (7) and (5), respectively.

3. The Proposed Multirate Loss Model – Poisson Case

In the proposed loss model, each link services Poisson arriving calls of K service-classes. New calls of service class k ($k = 1, \dots, K$) require b_k b.u. in order to be accepted in a link and have a generally distributed service-time with a mean of μ_k^{-1} . Let λ_{1k} and λ_{2k} be the arrival rates in each link of service-class k calls, respectively. We also denote, by j_1 and j_2 , the b.u. occupied in each link. Similarly to Section 2, each link l ($l = 1, 2$) has a support threshold th_{1l} and an offloading threshold th_{2l} , with $th_{1l} < th_{2l}$ and $0 \leq th_{1l}, th_{2l} \leq 1$.

To incorporate restricted accessibility into our model, we assume that each state j_l of link l , except for state $j_l = 0$ where link l is empty, is associated with a blocking probability, $pb_{l,k}(j_l)$. When there are no available b.u. for calls of service-class k in link l (i.e. when $j_l \geq C_l - b_k + 1$), then $pb_{l,k}(j_l) = 1$. Similarly, in the case of an empty system, $pb_{l,k}(0) = 0$.

The procedure of admitting a new service class k call that arrives in link l ($l = 1, 2$) is the following:

1. If $(0 \leq j_l < \lfloor th_{2l}C_l \rfloor)$, then the call is handled by link l . In addition, if $j_l + b_k \leq C_l$, then the call is accepted in link l with probability $1 - pb_{l,k}(j_l)$.
2. If $\lfloor th_{2l}C_l \rfloor \leq j_l$, then:
 - If $(0 \leq j_m < \lfloor th_{1m}C_m \rfloor)$, the call is offloaded to link m and if $j_m + b_k \leq C_m$, the call is accepted in link m with probability $1 - pb_{m,k}(j_m)$.
 - If $\lfloor th_{1m}C_m \rfloor \leq j_m$, then link m operates in *normal mode* and, therefore, does not support offloaded calls. Thus, the call is handled by link l . If $j_l + b_k \leq C_l$, then the call is accepted in link l with probability $1 - pb_{l,k}(j_l)$. Otherwise, the call is blocked and lost.

To calculate the CBP of service-class k calls, we assume that each link is an independent EMLM system under restricted accessibility [48] and, therefore, the CBP of service-class k calls in the first and the second link can be given via Eqs. (13) and (14), respectively:

$$P_{res,b_{1k}} = P_{res,1k}(C_1)P_{res,2}(j_2 \geq \lfloor th_{12}C_2 \rfloor), \quad (13)$$

$$P_{res,b_{2k}} = P_{res,2k}(C_2)P_{res,1}(j_1 \geq \lfloor th_{11}C_1 \rfloor), \quad (14)$$

where $P_{res,lk}(C_{lk})$ is the CBP of service-class k calls in link l ($l = 1, 2$) and $P_{res,l}(j_l \geq \lfloor th_{1l}C_l \rfloor)$ is the probability that link l does not operate in the support mode.

The values of $P_{res,lk}(C_l)$ in Eqs. (13) and (14) can be given by:

$$P_{res,lk}(C_l) = \sum_{j_l=1}^{C_l} G_l^{-1} q(j_l) pb_{l,k}(j_l), \quad (15)$$

where $q(j_l)$ expresses the unnormalized values of the occupancy distribution of link l ($l = 1, 2$) while $G_l = \sum_{j_l=0}^{C_l} q(j_l)$ refers to the normalization constant.

In Eq. (15), the values of $q(j_l)$ can be computed via:

$$q(j_l) = \begin{cases} 1 & \text{for } j_l = 0 \\ \frac{1}{j_l} \sum_{k=1}^K a_{lk} b_k q(j_l - b_k) \times [1 - pb_{l,k}(j_l - b_k)] & \text{for } j_l = 1, \dots, C_l \end{cases}, \quad (16)$$

where $a_{lk} = \lambda_{lk}/\mu_k$ is the total offered traffic-load of service-class k calls in link l .

Regarding the values of $P_{res,l}(j_l \geq \lfloor th_{1l}C_l \rfloor)$, in (13) and (14), they can be calculated by:

$$P_{res,l}(j_l \geq \lfloor th_{1l}C_l \rfloor) = \sum_{j_l=\lfloor th_{1l}C_l \rfloor}^{C_l} G_l^{-1} q(j_l), \quad (17)$$

where $q(j_l)$ is given by (16).

Finally, Eq. (18) is proposed for determining the total blocking probability of service-class k calls in the two-link system:

$$P_{res,b_k} = \frac{\lambda_{1k}}{\lambda_{1k} + \lambda_{2k}} P_{res,b_{1k}} + \frac{\lambda_{2k}}{\lambda_{1k} + \lambda_{2k}} P_{res,b_{2k}}. \quad (18)$$

4. The Proposed Multirate Loss Model – Quasi-Random Case

Contrary to the model proposed in Section 3, we now assume that each link services quasi-random traffic generated by K service-classes. New calls of service-class k ($k = 1, \dots, K$) in link l ($l = 1, 2$) are generated via a finite source population N_{lk} and require b_k b.u. Let $\lambda_{1k,\text{fin}}$ and $\lambda_{2k,\text{fin}}$ be the arrival rates of service-class k idle sources in the first and second link, respectively. Then, $\lambda_{lk,\text{fin}} = (N_{lk} - n_{1k} - n_{2k})v_{lk}$, where n_{lk} refers to the in-service calls of service-class k in link l and v_{lk} is the arrival rate per idle source of service-class k in link l . The corresponding offered traffic-load per idle source is $a_{lk,\text{idle}} = v_{lk}/\mu_k$. Note that if $N_{lk} \rightarrow \infty$ for all service-classes and the total offered traffic-load is constant, then calls arrive in the system according to a Poisson process, resulting in the model described in Section 3.

The process of admitting a new service-class k call that arrives in link l ($l = 1, 2$) is similar to that described in Section 3 and is therefore omitted.

To calculate the time congestion probabilities of service-class k calls, we assume that each link is an independent Engset multirate loss model (EnMLM) under restricted accessibility and, therefore, time congestion probabilities of service-class k calls in the first and the second link can be determined via Eqs. (19) and (20), respectively:

$$P_{\text{fin-res},b_{1k}} = P_{\text{fin-res},1k}(C_1)P_{\text{fin-res},2}(j_2 \geq \lfloor th_{12}C_2 \rfloor), \quad (19)$$

$$P_{\text{fin-res},b_{2k}} = P_{\text{fin-res},2k}(C_2)P_{\text{fin-res},1}(j_1 \geq \lfloor th_{11}C_1 \rfloor), \quad (20)$$

where $P_{\text{fin-res},lk}(C_{lk})$ is the time congestion probability of service-class k calls in link l ($l = 1, 2$) and

$P_{\text{fin-res},l}(j_l \geq \lfloor th_{1l}C_l \rfloor)$ is the probability that link l is not in the support mode.

The values of $P_{\text{fin-res},lk}(C_l)$ in Eqs. (19) and (20) can be determined via the following formula:

$$P_{\text{fin-res},lk}(C_l) = \sum_{j_l=1}^{C_l} G_l^{-1} q_{\text{fin}}(j_l) p b_{l,k}(j_l), \quad (21)$$

where $q_{\text{fin}}(j_l)$ expresses the unnormalized values of the occupancy distribution of link l ($l = 1, 2$), while $G_l = \sum_{j_l=0}^{C_l} q_{\text{fin}}(j_l)$ is the corresponding normalization constant.

In Eq. (21), the values of $q_{\text{fin}}(j_l)$ can be calculated as follows:

$$q_{\text{fin}}(j_l) = \begin{cases} 1 & \text{for } j_l = 0 \\ \frac{1}{j_l!} \sum_{k=1}^K (N_{lk} - Y_k) a_{lk, \text{idle}} b_k q_{\text{fin}} \times \\ (j_l - b_k) [1 - p b_{l,k}(j_l - b_k)] & \text{for } j_l = 1, \dots, C_l \end{cases}, \quad (22)$$

where $Y_k = y_{1k}(j_1 - b_k) - y_{2k}(j_2 - b_k)$ and $y_{lk}(j_l)$ is the average number of service-class k calls in state j_l of link l , assuming that the system accommodates Poisson traffic.

The values of $y_{lk}(j_l)$ are given by:

$$y_{lk}(j_l) = \frac{a_{lk} q(j_l - b_k) [1 - p b_{l,k}(j_l - b_k)]}{q(j)}, \quad (23)$$

where the values of $q(j)$ are computed via Eq. (16) (i.e. by the corresponding Poisson model).

The rationale behind Eqs. (22) and (23) is similar to that of the model from [51] that proposes an algorithm for the approximate determination of time congestion probabilities in the EnMLM.

Finally, regarding the values of $P_{\text{fin-res},l}(j_l \geq \lfloor th_{1l}C_l \rfloor)$, in Eqs. (19) and (20), they are given by:

$$P_{\text{fin-res},l}(j_l \geq \lfloor th_{1l}C_l \rfloor) = \sum_{j_l=\lfloor th_{1l}C_l \rfloor}^{C_l} G_l^{-1} q_{\text{fin}}(j_l), \quad (24)$$

where $q_{\text{fin}}(j_l)$ is determined via Eq. (22).

5. Numerical Examples

In this section, we consider two examples and provide simulation and analytical CBP results of the proposed model assuming Poisson traffic. Simulation results are based on Simscript III [52] and are mean values of 7 runs. In each run, ten million calls are generated. The first 5% of these generated calls are not considered in the CBP results so as to account for a warm-up period.

In the first example, we consider a system with the capacities of $C_1 = 24$ b.u. and $C_2 = 20$ b.u., accommodating two service-classes whose calls require $b_1 = 1$ and $b_2 = 2$ b.u., respectively. Let $\lambda_{11} = 9$ calls/min and $\lambda_{12} = 1$ call/min, for the first link. Similarly, let $\lambda_{21} = 7$ calls/min and

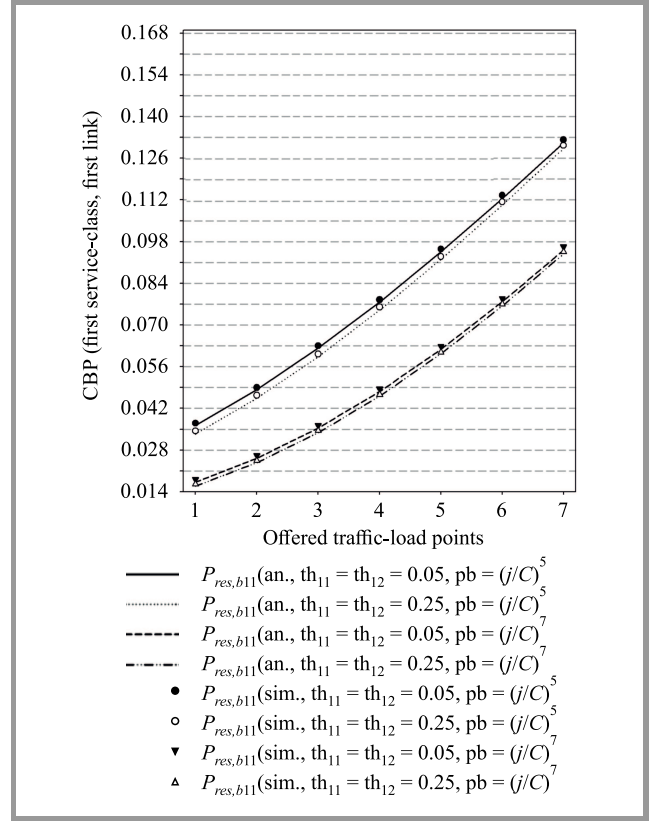


Fig. 2. CBP of the first service-class in the first link (example 1).

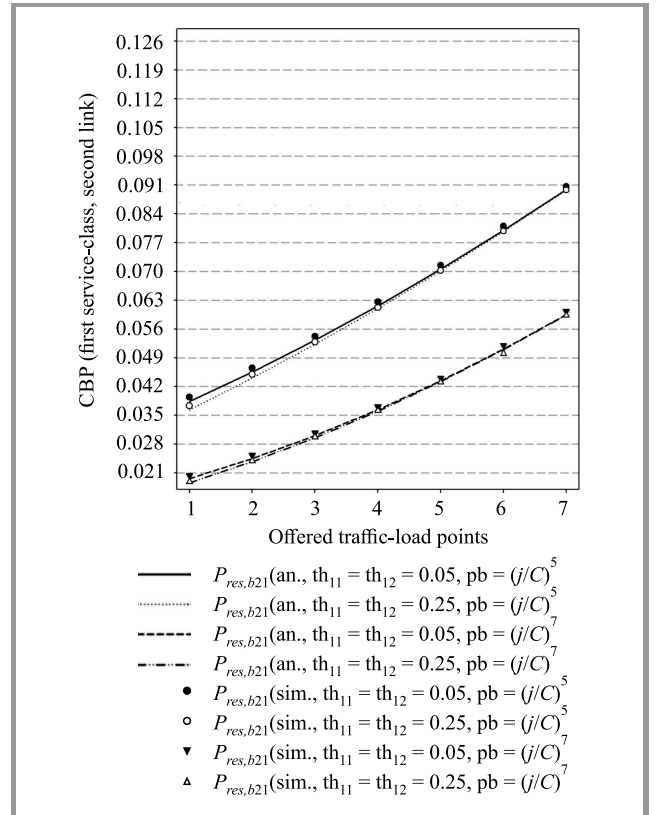


Fig. 3. CBP of the first service-class in the second link (example 1).

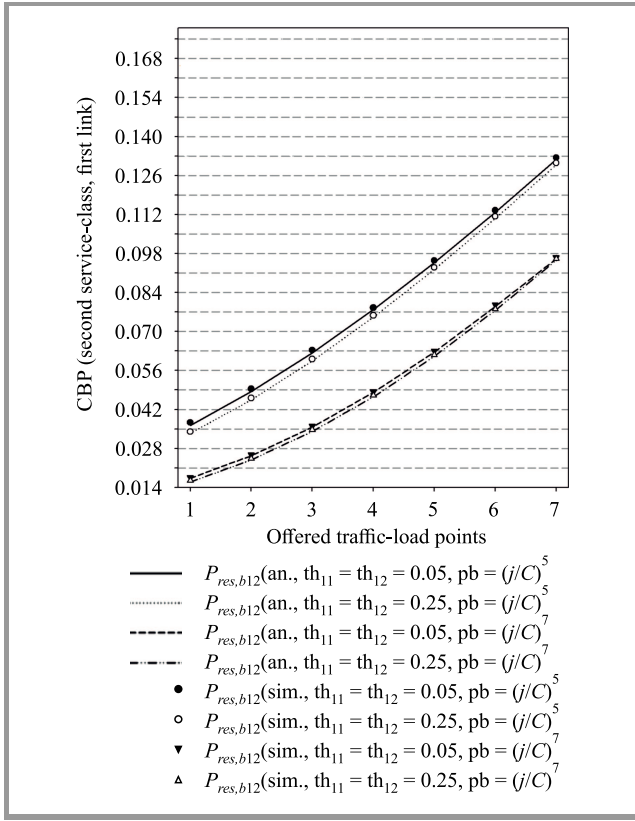


Fig. 4. CBP of the second service-class in the first link (example 1).

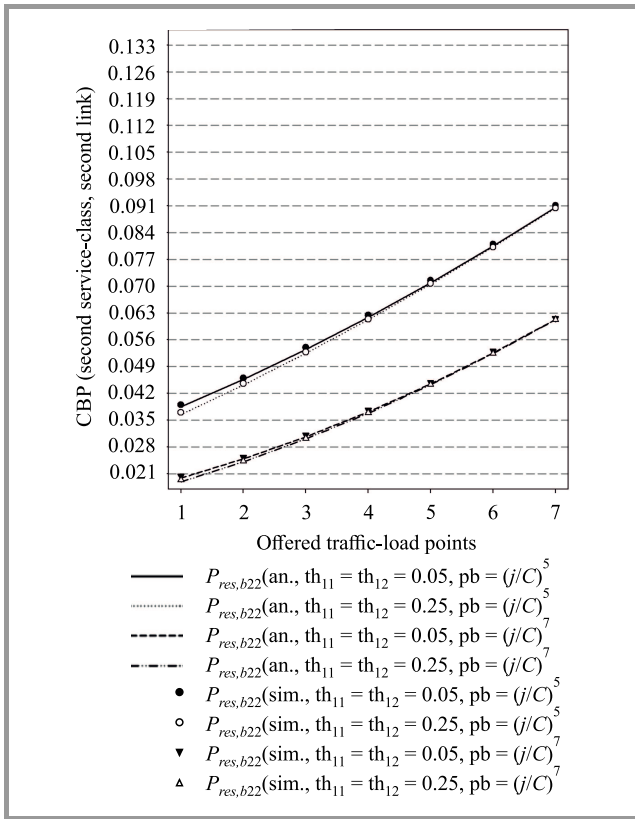


Fig. 5. CBP of the second service-class in the second link (example 1).

$\lambda_{22} = 1$ call/min, for the second link. Also let $\mu_1^{-1} = \mu_2^{-1} = 1.0$ min. Regarding the values of the thresholds, let the offloading thresholds equal $th_{21} = th_{22} = 0.7$, and let us consider two support threshold scenarios: (i) $th_{11} = th_{12} = 0.05$ and (ii) $th_{11} = th_{12} = 0.25$. Finally, regarding the restricted accessibility factors for each link, two sets are studied: (i) $pb_{l,1}(j_l) = pb_{l,2}(j_l) = (j_l/C_l)^5$ and (ii) $pb_{l,1}(j_l) = pb_{l,2}(j_l) = (j_l/C_l)^7$ where $l = 1, 2$.

In the x -axis of Figs. 2–5, λ_{11} and λ_{21} increase in steps of 1.0 and 0.5, respectively. So, point 1 is: $(\lambda_{11} = 9.0, \lambda_{12} = 1.0, \lambda_{21} = 7.0, \lambda_{22} = 1.0)$, while point 7 is: $(\lambda_{11} = 15.0, \lambda_{12} = 1.0, \lambda_{21} = 10.0, \lambda_{22} = 1.0)$.

In Figs. 2–3, we present the CBP for the first service-class in each link, respectively. In Figs. 4–5, the corresponding CBP results for the second service-class are presented. Figures 2–5 show that the analytical CBP results:

- Are close to the simulation results, especially when the values of support thresholds th_{11} and th_{12} are at a reasonable level (e.g. 0.05 to 0.25). Depending on the system, higher values of th_{11} and th_{12} may increase the discrepancy between simulation and analytical CBP results. This behavior may also be observed in [29] and is anticipated due to the fact that both links work independently.
- The choice of $pb_{l,k}(j_l)$ greatly affects CBP. The higher values of set 1 leads to much higher CBP compared to the values of set 2.

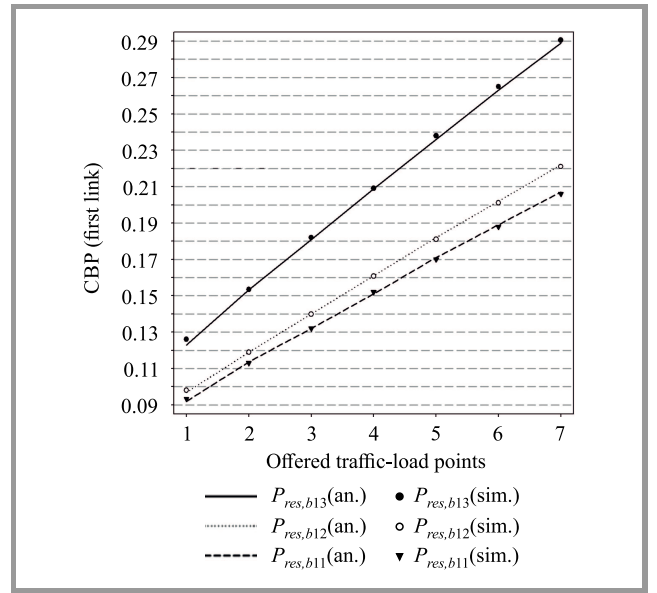


Fig. 6. CBP of each service-class in the first link (example 2).

In the second example, a larger system is considered. More precisely, we study a system of capacities $C_1 = 40$ b.u. and $C_2 = 45$ b.u. that services three service-classes whose calls require $b_1 = 1$, $b_2 = 3$ and $b_3 = 6$ b.u., respectively. Let $\lambda_{11} = 5$ calls/min, $\lambda_{12} = 3$ calls/min and $\lambda_{13} = 2$ calls/min, for the first link. Similarly, let $\lambda_{21} = 5$ calls/min, $\lambda_{22} = 3$ calls/min and $\lambda_{23} = 2$ calls/min, for the second link. Also

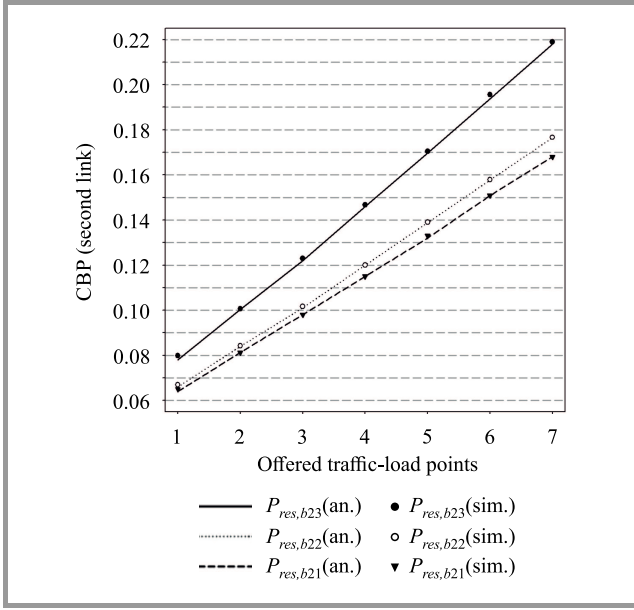


Fig. 7. CBP of each service-class in the second link (example 2).

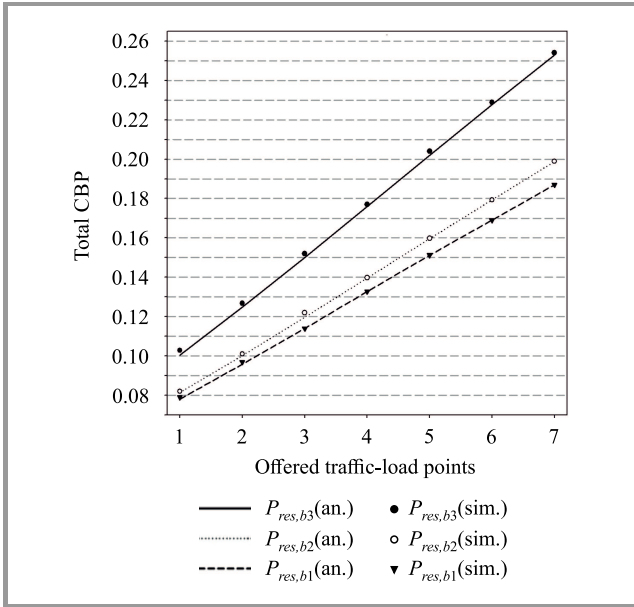


Fig. 8. Total CBP (example 2).

let $\mu_1^{-1} = \mu_2^{-1} = \mu_3^{-1} = 1$ min. Regarding the values of the thresholds let the offloading thresholds equal $th_{21} = th_{22} = 0.7$ and let the support thresholds equal $th_{11} = th_{12} = 0.2$. Finally, regarding the restricted accessibility factors, let $pb_{l,1}(j_l) = pb_{l,2}(j_l) = (j_l/C_l)^7$ where $l = 1, 2$.

In the x -axis of Figs. 6–8, $\lambda_{11}, \lambda_{12}, \lambda_{13}, \lambda_{21}, \lambda_{22}$ and λ_{23} increase in steps of 0.2, respectively. So, point 1 is: $(\lambda_{11} = 5.0, \lambda_{12} = 3.0, \lambda_{13} = 2.0, \lambda_{21} = 5.0, \lambda_{22} = 3.0, \lambda_{23} = 2.0)$, while point 7 is: $(\lambda_{11} = 6.2, \lambda_{12} = 4.2, \lambda_{13} = 3.2, \lambda_{21} = 6.2, \lambda_{22} = 4.2, \lambda_{23} = 3.2)$. In Figs. 6–7, we show the CBP for all service-classes in each link, respectively. In Fig. 8, we present the total CBP results for all service-classes. Figures 6–8 show that the analytical CBP results are again quite close to the corresponding simulation re-

sults. A similar degree of accuracy has been observed for various two-link systems that we studied.

6. Conclusion

We propose new multirate loss models for the call-level analysis of a two-link system with restricted accessibility that accommodates Poisson or quasi-random arriving calls of different service-classes. In this system, each link may support calls offloaded from the other link. The proposed models do not have a PFS for the steady state probabilities due to the restricted accessibility and existence of the offloading mechanism. However, we show that an approximate method does exist for the determination of blocking probabilities, achieving a satisfactory degree of accuracy compared to simulation. As a future work, we intend to analyze the case of interference between the two links, using the proposed model as a springboard for further considerations.

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Modeling of Complex Non-Full-Availability Systems

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Abstract—This article presents an analytical model of complex non-full-availability telecommunications systems. The high degree of accuracy of the model is demonstrated by a comparison with the results of simulation experiments. Due to the introduction of an availability parameter, this model may be used in the future for analyzing real systems, such as, for example, cloud computing infrastructure. This will be possible provided that the function combining the physical structure of the system and the availability parameter is specified. This problem will be addressed in our future work and will constitute the next stage of research undertaken by the authors.

Keywords—analytical model, availability, non-full-availability systems.

1. Introduction

From the traffic engineering perspective, telecommunications and computer systems that are currently under development are characterized by a peculiar but rarely noticed property, namely they constitute non-full-available systems. Non-full-availability is based on the fact that calls arriving at the input end of a system have no access to all of the system's resources. Instead, they only have access to a certain subset of those resources. In the literature, this subset is referred to as availability [1], [2]. Basically, systems of this type have existed in telecommunications from the very inception of the domain. Over time, however, the source of non-full-availability has changed. Originally, non-full-availability resulted from the limited nature of telecommunications networks' resources, e.g. outputs of communications networks and telephone exchange systems, where the number of such outputs exceeded the capacities of output groups. Today, non-full-availability of a system results primarily from the individual traffic management mechanisms that are used by networks operators to optimize the use of the available resources (e.g. resource reservation, compression, priorities and queuing) and, very frequently, from the wide area or distributed nature of the physical infrastructure. Non-full-availability of a system may be illustrated with an example of a dynamic resource reservation system [3], [4]. Its operating principle is based on the system's occupancy limit (boundary) adopted which, when exceeded, causes the resources of the system to be available only to certain pre-defined calls, at the expense of the remaining calls.

Concurrent use of a number of mechanisms means that the degree of dependence between the amount of resource-savailable to a given call and the system occupation level becomes even higher. Physical infrastructure of a cloud computing system may serve as the best example of a modern non-full-availability solution. A high number of devices, the distributed nature of the system, the mechanisms used to optimize of the utilization of resources in cloud computing, the mechanisms that guarantee safe provision of the services, the algorithms employed to reduce power consumption, as well as the nature of the services offered [5]–[8] all make it possible for new calls to be supported and serviced by the resources in the cloud.

There is a number of models available in the literature concerned with analyzing current telecommunications systems. As a rule, however, these models deal with a single traffic management mechanism, while a real system is approximated by a model of a full-availability system (historically known as a full-availability group model), e.g. shown in [9]–[15]. Models of non-full-availability systems, such as the limited-availability group model (LAG) or Erlang's ideal grading (EIG), are used for that purpose less frequently [16]–[19].

Non-full-availability systems belong to a group of the so-called state-dependent systems [20], [11]. This dependence may result from the existence of the function that relates the number of busy resources in a system to the calls that arrive at the input, or, alternatively, dependence between the number of occupied system resources and the decision to admit a new call for service may exist. In the former case, we deal with systems with a state-dependent call stream. In the latter case, we deal with systems with a state-dependent service process. [20] presents general rules applicable to modeling this type of systems with the use of a one-dimensional Markov process, after the so-called state transition coefficient has been taken into account. This coherent methodology makes it possible to model very complicated systems as well, providing that the relevant transition coefficients may be determined.

This article presents an analytical model of a complex non-full-availability system. The structure of the system shows some non-full-availability features, i.e. it is composed of a group of separated resources that are non-full-availability systems themselves. It is the authors' opinion that the model may be relied upon, in the future, to model the

physical infrastructure of cloud computing solutions. The authors of the article are currently conducting research on the conditions under which this may become feasible. The remaining part of the article is structured as follows. Section 2 presents the main assumptions for modeling state-dependent systems. Section 3 provides a description of the structure of this complex non-full-availability system and the corresponding analytical model proposed. On account of the approximate nature of the model, Section 4 presents a comparison between the results obtained with the analytical model and the results obtained during simulation experiments. The article is summarized in Section 5.

2. State-dependent Systems

2.1. Resources and Traffic in Telecommunications Networks

Today's telecommunications networks are primarily packet-switched networks based on the IP protocol. The analysis of a network at the packet level is complicated and time consuming. Papers [9], [21], [22] prove that it is possible to analyze packet networks at the call level, where calls correspond to packet streams (with constant or variable transmission rates) that are sent between a given source and the receiver. Thus, the network traffic represented has the properties of Poisson traffic flow. Thanks to this operation, it is possible to express the amount of resources of the system in dimensionless units, i.e. in resource allocation units (AU). This transition from dimensional units to dimensionless units is known as discretization. The said operation increases versatility of models, since the scope of their operation is not limited to the analysis of the link bit rate (in bps), but is expanded to include other system resources, such as storage memory, RAM memory or processors [23].

The above-mentioned discretization process will be illustrated with an example of a network resource (e.g. a link between routers) with the capacity of C bps. This resource services constant bit rate (CBR) streams and variable bit rate (VBR) streams. In the discretization process, the VBR streams are represented by streams with a constant bit rate, with the transmission speed equal to the so-called equivalent bandwidth (EB). EB for individual call classes may be determined either on the basis of the maximum bit rates or on the basis of heuristic algorithms, in line with the principle that the result of a VBR call service in the network is the same as the result of a CBR call service with the bit rate equal to EB [24]. Most frequently, these algorithms take into account such traffic parameters and networks system parameters as the maximum and the average bit rate of a call, bit rate variance, acceptable delay, jitter and quality of service (QoS) parameters adopted for a given service, etc. It is worth emphasizing at this point that the determination of the EB value on the basis of the maximum bit rates makes it possible to analyze the system under consideration for the worst conditions, i.e. for the highest offered traffic. The knowledge of EB values (the manner in which

EB is determined for individual packet streams is beyond the scope of the present article) for all traffic classes offered to the system makes it possible to determine the resource allocation unit of the system, as the traffic class is composed of all packet streams with similar parameters. The general assumption is that 1 AU is equal to the greatest common divisor of the equivalent bandwidths of all traffic classes that are offered to the system:

$$1 \text{ AU} \equiv c_{\text{AU}} = \text{GCD}(c_1, \dots, c_i, \dots, c_m), \quad (1)$$

where c_i (bps) is the equivalent bandwidth of class i , whereas m is the number of traffic classes offered to the system. The parameter c_{AU} determines the bit rate per a single AU.

After the determination of c_{AU} it is possible to express the capacity of resource V and the demands of individual traffic classes t_i in the AUs, where C is the resource capacity:

$$V = \left\lfloor \frac{C}{c_{\text{AU}}} \right\rfloor, \quad (2)$$

$$t_i = \left\lfloor \frac{c_i}{c_{\text{AU}}} \right\rfloor. \quad (3)$$

Call classes are described by:

- λ_i intensities of call stream of class i ($1 \leq i \leq m$); this parameter characterizes the frequency at which new calls arrive,
- μ_i intensities of service stream of class i ($1 \leq i \leq m$); this parameter characterizes call service time,
- a_i average intensity of traffic of class i (in relation to calls):

$$a_i = \lambda_i / \mu_i. \quad (4)$$

To simplify the analysis of modern networks, an additional assumption may be made that 1 AU corresponds to the bit rate of 1 bps (or 1 kbps):

$$c_{\text{AU}} = 1 \text{ bps}. \quad (5)$$

This choice of the AU value causes the demands of individual classes – Eq. (3) and the capacity of the system – Eq. (2) to be equal, in terms of their value, with the demanded bit rates and the total capacity of the system:

$$c_i = t_i, \quad C = V. \quad (6)$$

2.2. General Model of State-dependent Systems

The general method for modeling state-dependent systems by a one-dimensional Markov process is presented in [20]. The dependence of a system on its state (i.e. resource occupancy) may be a result of the following:

- limited number of traffic sources – in this case, we deal with systems with a state-dependent call stream – a single source can generate a new call only when it is unoccupied. As a consequence, the ration the number of calls that arrive at the input of the sys-

between tem within a given time unit and the number of calls admitted earlier is changed. For instance, for Engset traffic, this number decreases along with an increase in the number of occupied traffic sources;

- system structure – in this case, we deal with a system with a state-dependent service stream. The structure of the system and the call service process may cause only a part of the resources to be available, just as it is the case with switching networks, for example. The structure of the connections within the network and the calls that are currently serviced may then affect the setting up of a connection between a free input and a free output of the network, rendering it impossible;
- algorithms operating the call admission control function – similarly as in the former case, we deal with a state-dependent service stream. Call admission control algorithms may prevent new calls from being serviced under certain operating conditions, e.g. based on a dynamic resource reservation in relation to the total number of busy AUs in the system.

In state-dependent systems, the transition between states depends not only on the parameters λ_i and μ_i , but additionally on the so-called conditional transition coefficient $\sigma_i(n)$, where i is the number of a call classes, and n is the number of busy AUs in the system. This is presented, in a simplified form, in Fig. 1.

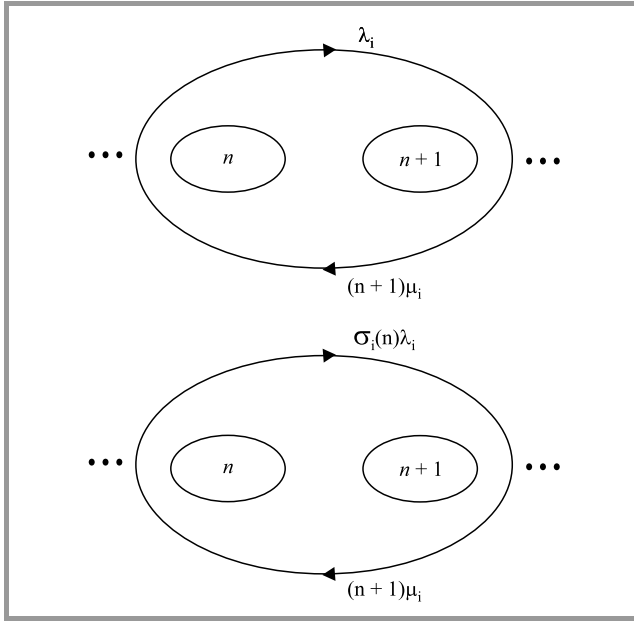


Fig. 1. Transition between states of Markov process: (a) state-independent system, (b) state-dependent system.

It is proved in [20] that in the case of systems in which state-dependence is the result of a number of different and independent sources, the total conditional transition coefficient may be determined in the following way:

$$\sigma_i(n) = [\sigma_i(n)]^R [\sigma_i(n)]^A [\sigma_i(n)]^P, \quad (7)$$

where:

- $[\sigma_i(n)]^R$ is the conditional transition coefficient of class i in occupancy state n , resulting from the properties of a call stream of this class,
- $[\sigma_i(n)]^A$ is the conditional transition coefficient of class i in occupancy state n , resulting from the structure of the system,
- $[\sigma_i(n)]^P$ is the conditional transition coefficient of class i in occupancy state n , resulting from the function of new call admission in the system.

The knowledge of the values of all $\sigma_i(n)$ parameters makes it possible to determine occupancy distribution within the system with the use of the following formula [20]:

$$n[P_n]_V = \sum_{i=1}^m a_i t_i [\sigma_i(n)]_V [P_{(n-t_i)}]_V, \quad (8)$$

where $[P_n]_V$ is the probability of occupancy n AUs in the system.

3. Model of a Complex Non-full-availability System

3.1. Structure of Complex Non-full-availability Systems

The analytical model proposed in the article was developed for a system that is presented, in its simplified form, in Fig. 2. The assumption was that the system was composed of k separated resources (subgroups) with the capacity of f AUs. The total capacity of the system is:

$$V = kf. \quad (9)$$

The division of all AUs within the system into separated full-availability resources renders the system a non-full-available solution. The literature of the subject describes its analytical model, known as the limited-availability group model [25]. In this paper, we assumed that each of k separated resources shows the properties of a non-full-availability system. Non-full-availability of a single separated resource may stem from its internal structure or from the call admission control algorithm. The literature presents a small number of analytical models pertaining to non-full-availability systems, e.g. Palm-Jacobeus formula, modified Palm-Jacobeus formula or the EIG model [1].

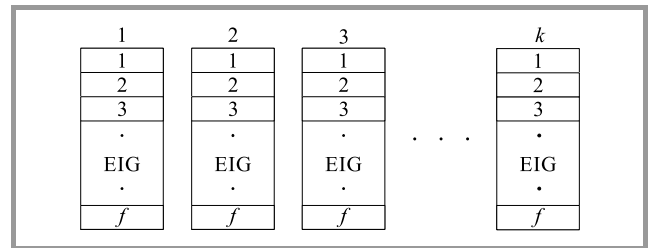


Fig. 2. Simplified diagram of the structure of a complex non-full-availability system.

In analytical models, non-full-availability of resources is characterized by a parameter known as availability d_i ($1 \leq i \leq m$) [1], [26]. This parameter determines the number of AUs in the system to which calls of a given class have access. It is unique for each class of calls (and it may assume different values for each of those classes), while its value is based on the structure of the system and on the call admission control mechanism. It should be stressed that if calls of class i have availability equal to d_i AUs of the system, this does not necessarily mean that all of them have access to the same AUs. Typically, they have access to different AUs of the system. Calls or, to be more precise, their sources that have access to the same AUs in the system are called a load group. In EIG, the number of load groups (for calls of class i) is equal to the number of choices of d AUs from V :

$$g_i = \binom{V}{d_i}. \quad (10)$$

In real world systems, this number is typically lower. Figure 3 shows a simple single-service system with the capacity of 3 AUs and availability of $d = 2$ AUs that illustrates the idea of availability and presents an example of a blocking state.

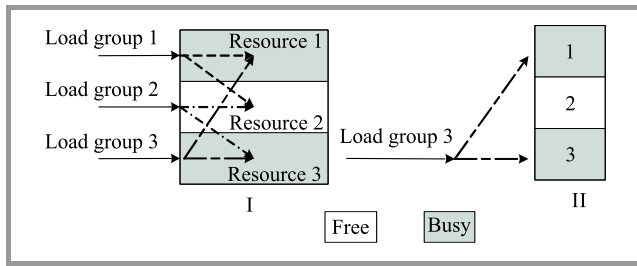


Fig. 3. Non-full-availability system: (a) the availability concept, (b) occupied load group no. 3.

The assumption is that in the complex non-full-availability system under consideration, new calls are admitted according to the following algorithm. Assume that a call of class i that demands t_i AUs for service arrives at the input to the system. In the first instance, the number of unoccupied AUs in the entire system is checked. If the number of free AUs in the entire system is lower than t_i AUs, then the call is lost. When this is not the case, a further check is performed to verify whether in a single separated resource there exist at least t_i unoccupied AUs. This condition is particularly important, as calls may be supported only by AUs that belong to one separated resource. If such resources are not available, then the call is rejected. In the case where the system has separated resources that have at least t_i free AUs available, one of the available resources is chosen randomly and the system checks whether the resource may admit the call under consideration for service. The absence of a possibility to admit a new call may result from the fact that this new call has no access to (i.e. cannot occupy) free AUs in the resource. If the selected separated resource does not offer such a possibility, then the next one

is selected. In the worst case, all k separated resources will be checked. If none of k resources can service the new call, the call is lost again.

3.2. Analytical Model

To determine occupancy distribution in the system under consideration, as given by Eq. (8), and, in consequence the blocking probability, it is necessary to determine the values of the conditional transition coefficient $\sigma_i(n)$ for all call classes offered and for all system occupancy states. The state-dependence of the service stream and, consequently, the non-full-availability in the system under consideration, results from two factors:

- all AUs in the system are divided into k identical separated resources (subsystems), while a call may be completely serviced by one separated resource only,
- each separated resource has limited access to its own AUs which leads, in consequence, to a situation in which - despite a sufficient number of free AUs - the call cannot be admitted for service due to the lack of access to free AUs.

On the basis of Eq. (7), we can write:

$$\sigma_i(n) = [\sigma_i(n)]^{SZ} [\sigma_i(n)]^S, \quad (11)$$

where $[\sigma_i(n)]^{SZ}$ is the transition coefficient that results from the division of the AUs of the system into the separated resources, whereas $[\sigma_i(n)]^S$ results from the non-full-availability of a single separated resource.

Now, let us consider the manner in which parameters $[\sigma_i(n)]^{SZ}$ and $[\sigma_i(n)]^S$ may be determined.

The values of the coefficient $[\sigma_i(n)]^{SZ}$ may be determined as follows. Since calls may be completely serviced only by a single separated resource, this resource has to have at least t_i unoccupied AUs. To satisfy this condition, an appropriate distribution of free AUs between all separated resources is required. An unfavorable distribution of free AUs within the system may be followed by new calls being rejected, as presented in Fig. 4. The said figure shows the distribution of three free AUs in a system that is composed of two separated resources. If, at the input to this system, a call that demands 3 AUs arrives, then it may be admitted for service only in the case of 2 from 4 potential distributions of unoccupied AUs within the entire system.

To determine the conditional transition coefficient, the dependence proposed in [25] can be used:

$$[\sigma_i(n)]_V^{SZ} = \frac{F(V-n, k, f) - F(V-n, k, t_i-1)}{F(V-n, k, f)}, \quad (12)$$

where the function $F(x, k, f)$ is given by:

$$F(x, k, f) = \sum_{i=0}^{\lfloor \frac{x}{f+1} \rfloor} (-1)^i \binom{k}{i} \binom{x+k-1-i(f+1)}{k-1}. \quad (13)$$

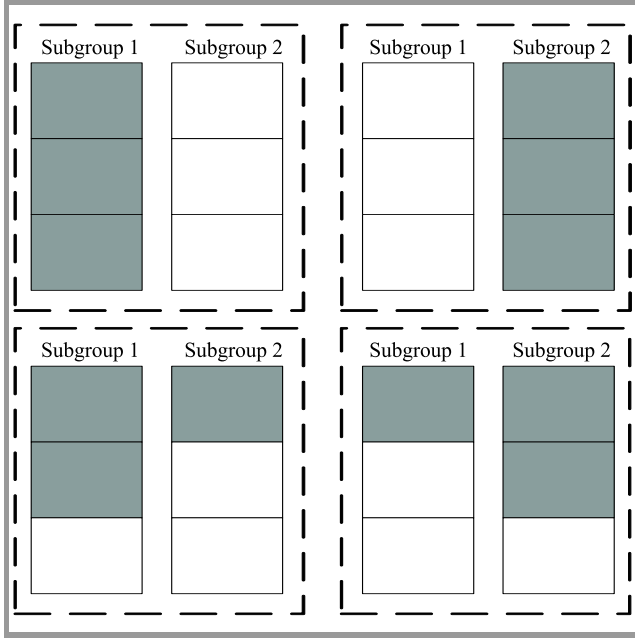


Fig. 4. Distribution of free AUs in separated resources.

Parameter $[\sigma_i(n)]_V^{SZ}$ is calculated based on the assumption that separated resources are of the full-availability variety. This assumption is based on the new connection setup algorithm implemented. According to this algorithm, the separated resources are checked for free AUs first, without checking whether there is access to them.

Each of the separated resources may be the source for non-full-availability as well. Systems in which calls have limited access to their own AUs may be modeled by EIG [26], [1]. In this model, the conditional transition coefficient is determined with the use of the following formula:

$$[\sigma_i(n)]_f^{EIG} = 1 - \sum_{d_i-t_i+1}^z \frac{\binom{d_i}{x} \binom{f-d_i}{n-x}}{\binom{f}{n}}, \quad (14)$$

where: $z = n - t_i$, if $(d_i - t_i + 1) \leq (n - t_i) < d_i$, $z = d_i$, if $(n - t_i) \geq d_i$.

In order to substitute Eqs. (12) and (14) into Eq. (11), two problems have to be solved first. Firstly, the occupancy of a single separated resource changes from 1 to f , while the occupancy of the whole of the system changes from 1 to V . Secondly, the call admission algorithm allows all separated resources that have at least t_i free AUs to be checked. In the proposed analytical model, the first problem was solved in the following way: since the assumption was that the choice as to the separated resources was random, then it could be adopted that approximately (rough estimate) each of them had equal loads and, what followed, the average number of occupied AUs in each resource was equal to $\frac{n}{k}$, where n was the occupancy state in the entire system. The other problem, in turn, was solved thanks to the adoption of the assumption that since service admission of a new call depends on the state of each of the separated resources, then

product dependence must exist between the transition coefficients in each of the separated resources and the resultant transition coefficient $[\sigma_i(n)]^S$. The proposed Eq. (15) was defined as follows: since the call admission algorithm for new calls assumes the possibility of checking all separated resources, then the admission of a new call is possible if such a possibility exists in just one of them:

$$[\sigma_i(n)]^S = 1 - \left(1 - \sigma_i\left(\frac{n}{k}\right)\right)^k. \quad (15)$$

By taking into consideration Equations (12), (14) and (15), we can ultimately write:

$$\begin{aligned} \sigma_i(n) &= [\sigma_i(n)]^{SZ} [\sigma_i(n)]^S = \\ &= [\sigma_i(n)]^{SZ} \left[1 - \left(1 - \sigma_i\left(\frac{n}{k}\right)\right)^k\right] = \\ &= \frac{F(V-n, k, f) - F(V-n, k, t_i-1)}{F(V-n, k, f)} \times \\ &\quad \left[1 - \left[\frac{\sum_{d_i-t_i+1}^z \binom{d_i}{x} \binom{f-d_i}{\lfloor \frac{n}{k} \rfloor - x}}{\binom{f}{\lfloor \frac{n}{k} \rfloor}}\right]^k\right]. \end{aligned} \quad (16)$$

4. Results

To verify the proposed analytical model, the results obtained with the use of the model were compared with the results of the simulation. For this particular purpose, a simulation model of the non-full-availability system under consideration was devised and implemented in the C++ language. The simulator used the event scheduling methodology. To determine a single measurement, 5 series of simulations with 1,000,000 calls of the class that demanded the highest number of AUs for series each, were performed. The results obtained are presented in the form of a function of traffic offered to a single AU of the system:

$$a = \frac{\sum_{i=1}^m a_i t_i}{V}. \quad (17)$$

Another assumption was that the total traffic offered was divided between individual call classes in the following manner: $a_1 t_1 : a_2 t_2 : \dots : a_m t_m = 1 : 1 : \dots : 1$.

Figure 5 shows the result for the following system: $k = 3$, $f = 25$ AUs. The system was offered three classes of calls ($m = 4$) that demanded 1, 2, 3 and 4 AUs for service, respectively, while the availabilities were 8, 10, 12 and 15 AUs (Case 1). Figure 6 shows the result for the following system: $k = 3$, $f = 15$ AUs. The system was offered three classes of calls ($m = 3$) that demanded 1, 2 and 3 AUs for service, respectively, while the availabilities were 6, 8 and 10 AUs (Case 2). Figure 8 shows the result for the following system: $k = 3$, $f = 21$ AUs. The system was offered three classes of calls ($m = 3$) that demanded 1, 2

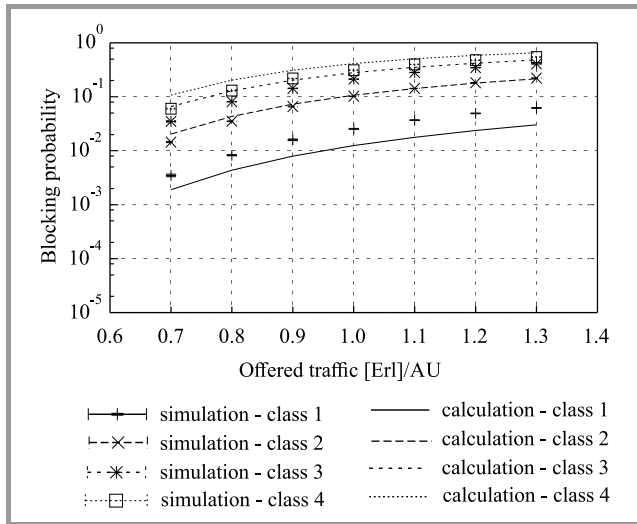


Fig. 5. Blocking probability in a complex non-full-availability system (Case 1).

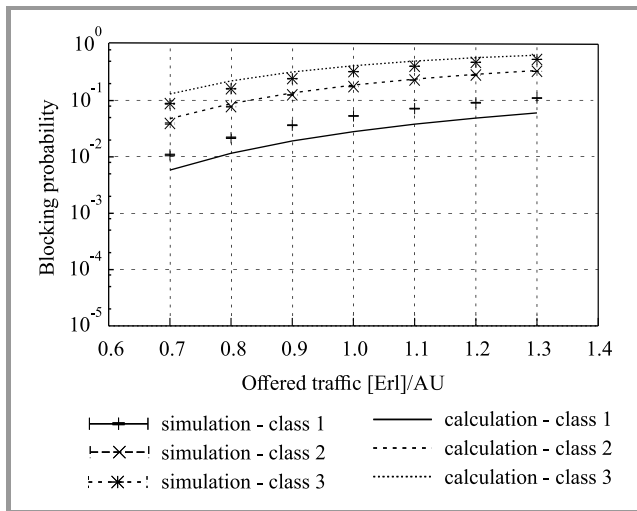


Fig. 6. Blocking probability in a complex non-full-availability system (Case 2).

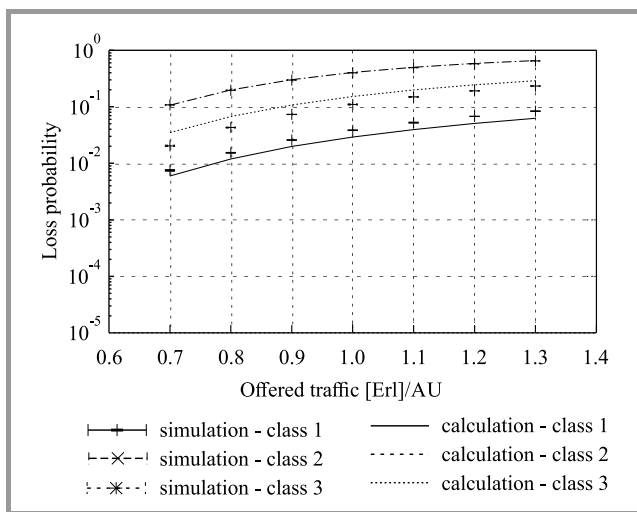


Fig. 7. Blocking probability in a complex non-full-availability system (Case 3).

and 5 AUs for service, respectively, while the availabilities were 7, 10 and 18 AUs (Case 3).

The model presented is an extension of the research described in article [27]. The previous model, in the scenario with low availability for the class requesting the least number of resources, showed some significant inaccuracies. Figure 5 shows the results obtained using models described in [27], [4], and the model described in the current article in connection with the following system: $k = 3$, $f = 24$ AUs. The system was offered three classes of calls ($m = 3$) that demanded 1, 3 and 4 AUs for service, respectively, while the availabilities were 5, 10 and 13 AUs (Case II). Thanks to the modification, the problem of inaccuracies in the youngest class has been eliminated.

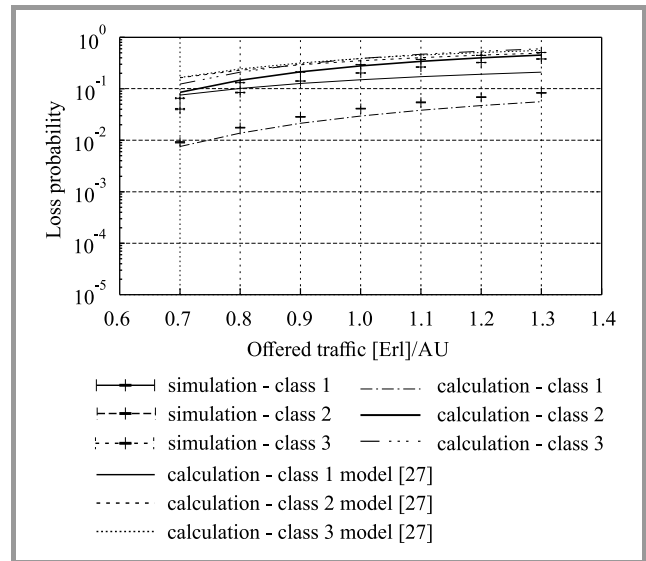


Fig. 8. Blocking probability in a complex non-full-availability system (Case 4) calculate by 2 models.

The analytical model proposed in the article is of the approximate variety. However, the presented results of the study show that the model is efficient at approximating the non-full-availability system under consideration. The highest error rate under investigation occurs for the class that demands the lowest number of AUs. However, this inaccuracy is much lower than that in the model presented in [27]. This error is based on a certain underestimation of the blocking probability, but the repeatability of this error makes it possible to introduce a relevant adjustment to the model. The accuracy of the model, as confirmed in the study, makes it possible to rely on the model for modeling non-full-availability systems, such as those used in cloud computing. This will be feasible, however, only when appropriate equations that make it possible to determine the availability in such systems are proposed. This will be the subject of further research the authors intend to conduct.

5. Summary

This article presents the architecture of a complex non-full-availability system and its analytical model developed by

the authors, enabling to determine loss coefficients and the blocking probability in this system. Though the proposed model is of the approximate nature, the results obtained in the study clearly indicate that is its highly accurate. The approximation errors that occur in the results, and are reported in the investigation, are of the fixed nature, which makes it possible to introduce simple corrections and adjustments that would nullify these differences during the further development stages. The authors are of the opinion that further research into complex non-full-availability systems is of utmost importance due to the dynamic growth of IT services and increasingly more sophisticated systems that are offered to users. The model presented in the article may become, in the future, a useful tool in modeling real telecommunications systems, such as data centers or cloud computing solutions. The fact that it requires a small amount of computer resources to perform calculations and is characterized by a relatively short time of operation is an additional advantage of the model presented.

Acknowledgements

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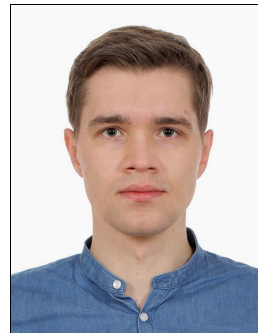
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V2X Communications for Platooning: Sensor Inaccuracy Aspects

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Abstract—Platooning is a future approach to autonomous driving in which vehicle-to-vehicle and vehicle-to-infrastructure communications play an important role. Tests performed in the past showed that a significant reduction in fuel consumption is possible when cars are traveling in a dense platoon formation. To increase the level of their awareness of the surrounding objects and to maintain a very short distance to the preceding vehicle, highly reliable on-board sensors are required. This paper discusses the impact of sensor inaccuracy on the performance and behavior of an autonomous vehicle platoon that makes use of wireless communications supported by context information from various databases and maps.

Keywords—context information database, sensor accuracy, V2X and 5G.

1. V2X Communications for Autonomous Driving

The fifth generation (5G) wireless system is planned to provide, in the future, ultra-reliable low-latency communications (URLLC) that are necessary to enable various safety-related or life-saving services [1]. One of the numerous ideas behind URLLC, is to use it for the provision of highly reliable services, such as vehicle-to-vehicle (V2V) communications, that may be relied upon in the management of autonomous and driverless platoons formed by vehicles on high-speed roads. Autonomous driving is perceived to be one of the enablers for a coordinated movement of a group of driverless vehicles, led by a platoon leader (platooning). In such a scenario, a group of vehicles exchange various control messages, enabling their autonomous movement at short inter-vehicle distances, without any human supervision. Increased interest in platooning is sparked primarily by the expected revenues, e.g. fuel savings of 7 to 15% for a platoon comprising multiple trucks [2]. Fuel savings result also in a substantial reduction of CO₂ emissions – according to the findings of the Energy project by ITS [3], if truck platooning was used in approximately 40% of all freight traffic operations, a 2.1% reduction in CO₂ emissions along highways could be achieved with the gap be-

tween trucks of 10 m, and the reduction would equal 4.8% if the gap could be decreased even further to 4 m.

Autonomous driving, aiming to improve road safety, may be achieved only with the support of both onboard sensors and inter-vehicular communications. However, to ensure that the automated controller responds faster than a human driver, extreme requirements concerned with reliability of wireless communications, as well as with the quality of on-board sensors used need to be met. For vehicle-to-everything (V2X) communications, the exchange of information between the platoon members and between the platoon and other devices may rely on short-range wireless communication protocols, such as dedicated short-range communications (DSRC), cooperative intelligent transport system (C-ITS) or cellular-V2X (CV2X). On the other hand, onboard sensors are necessary to provide information related to specific mobility parameters, facilitating the operation of autonomous vehicles. The accuracy of sensors used in platooning may cause various levels of deviations that strictly depend on the class of the physical devices installed. In this paper we consider a scenario that is depicted in Fig. 1, where the road train is driving, autonomously, on a high-speed road, in various environments. Stability and reliability are the most important aspects affecting intra-platoon communications. Onboard sensors also need to provide stable results that enable real-time and reliable

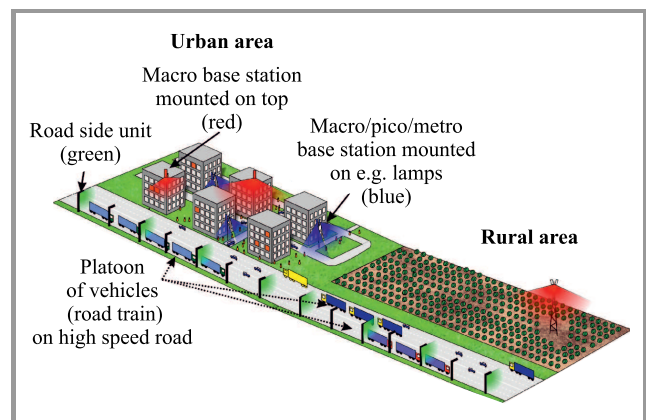


Fig. 1. Vehicle platooning and the surrounding wireless-communications environment.

monitoring of distances between the vehicles and of various operating parameters of the trucks forming the platoon.

In order to meet such strict requirements concerning reliability, exchange of information between the platooned trucks and the communication infrastructure (roadside units, small cells or macro cells) is required to efficiently manage prospective interference, both between individual platoons, and between the platoons and external sources of interference.

This paper focuses primarily on the impact that sensor inaccuracies (caused by imperfections) exert on the efficiency of V2X-aided platooning. In Section 2, we outline practical implementation of adaptive cruise control mechanisms. Section 3 presents an analysis of the impact that the quality of various sensors exerts on reliability and performance of the entire system. In Section 4, we discuss the main challenges related to platooning that relies on wireless communications, and provide a brief overview of the potential solutions improving reliability of the solution. Finally, Section 5 concludes the work.

2. Implementation Issues

Real-world implementation of the platooning technology is a big challenge. Driving needs to be controlled and coordinated with the use of sensors and actuators relying on a sophisticated algorithm. In addition, messages need to be exchanged between vehicles in the platoon, using wireless communications. The most important factors impacting the efficiency of platooning include the following: control system latency and accuracy of information collected by sensors. Cooperative adaptive cruise control (CACC) uses information both from local sensors and from other vehicles in the platoon (passed via a wireless link). CACC allows to boost platooning performance if, in addition to the requirement of a higher degree of accuracy of the control loop and of the sensors, wireless communication supports rapid, reliable and low-latency message exchange between the vehicles [4], [5].

The concept of autonomous driving, applied to a limited set of adjacent vehicles forming a platoon, has attracted a lot of interest in recent years. The CACC algorithm is proposed as a control mechanism for such a structure. Many researchers have been recently evaluating the performance of platooning that relied on the IEEE 802.11p standard for V2V communications. The SARTRE project [2] reported an experiment with two trucks and three cars forming a platoon. It has been proved that the platoon was able to drive at the maximum speed of 90 km/h, with the inter-vehicle distance ranging from 5 to 7 m. On the other hand, the tests described in [3], [6] allowed to achieve the speed of 80 km/h with the target inter-car distance equaling 10 m. This was the result achieved during the Energy ITS project performed in Japan for a platoon of three fully-automated trucks moving on an expressway.

From the point of view of wireless communications, CACC suffers from insufficient reliability of message exchange be-

tween the vehicles. A past study on the use of CACC while exchanging messages using the IEEE 802.11p standard revealed that even a non-significant increase in the density of vehicles on a highway may cause a congestion of the wireless channel. This will deteriorate performance of the CACC algorithm [7], [8]. One of the solutions of this problem is to use a dual-band transceiver that may operate simultaneously at two different frequencies (wireless channels). However, it should be noted that even if two ITS frequency channels are utilized, channel congestion is still possible if the scenario involves a high number of neighboring vehicles and a large amount of data to be exchanged. To solve this problem, the limit of two channels needs to be removed. This can be achieved by using the dynamic spectrum access technology, i.e. by relying on the cognitive radio concept. It is forecast that such an approach will be capable of improving the reliability of wireless communications and, indirectly, the performance of CACC used for vehicle platooning.

3. Role of Sensors in Wireless Communications within Platoons

On-board sensors constitute an essential part of a CACC-controlled vehicle system. They measure the basic physical quantities necessary for the proper operation of the control algorithm. Sensors used in a vehicle may be divided into two types. The first category is used to monitor the vehicle's environment, while sensors of the other type are responsible for the evaluation of the vehicle itself. The first class includes long/medium range radar (LRR/MRR) sensors and forward collision warning (FCW) sensors. They determine the distance to the preceding vehicle and its instantaneous speed. The other group includes sensors that determine the instantaneous speed and acceleration of the vehicle in which they are installed.

3.1. Sensor Inaccuracies

Sensor inaccuracies may be determined in different ways. The significance of sensor errors may be described using the following categories: absolute accuracy, error distribution, error values corresponding to a given percentile of error distribution or output noise.

In order to add inaccuracies of the analyzed sensors to the system under consideration, it is assumed that sensor inaccuracies may be modeled using normal distribution truncated to $\pm 3\sigma$. For the sensor accuracy distribution model in question, standard deviation that corresponds to a given sensor error has been determined. For the purpose of this paper, we investigated some off-the-shelf, average quality units that are installed or could be installed as onboard sensors.

The sensors investigated may be divided into the following groups, based on the parameter they measure: distance, relative velocity, ego velocity and acceleration. Depending on

their tasks, these sensors may be characterized by different ranges and accuracy levels. The task of distance sensors is to determine distances to adjacent objects, i.e. moving vehicles and stationary obstacles. Relative speed sensors evaluate the speed of the surrounding vehicles with respect to the ego vehicle. In addition to sensors that monitor the position of adjacent vehicles, sensors that determine the speed and acceleration of the vehicle on which they are mounted are also required. For measuring the velocity of the ego vehicle, electro-mechanical sensors may be used, with their accuracy depending on the quality of the vehicle's tires.

3.2. Sensors Measurements

It is the objective of the distance sensors to measure range to adjacent moving vehicles or obstacles. Sensors offer different ranges and measurement accuracies, depending on their purpose. Table 1 presents a set of commercially available units with information taken from catalog notes, measurement reports, and scientific papers, presenting their features. The tables present designated model parameters as well.

Table 1
Sensors model

Sensor reference	Accuracy	Std. dev.
Distance sensors		
[9]	92% = 0.5 m	0.2860
[10] – Type 1	90% = 0.6 m	0.3650
[10] – Type 2	90% = 1.5 m	0.9120
[10] – Type 3	90% = 0.9 m	0.5470
SRR contin. [11]	Accuracy = 0.2 m	0.0667
Relative speed sensors		
LLR3 Bosch [12]	Accuracy = 0.12 m/s	0.0500
MRR Bosch [13]	Accuracy = 0.11 m/s	0.0367
[9]	85% = 1.11 m/s	0.7700
[10] – Type 1	90% = 0.25 m/s	0.1520
[10] – Type 2	90% = 0.23 m/s	0.1400
[10] – Type 3	90% = 0.23 m/s	0.1400
Acceleration sensors		
VBOX [14]	Accuracy = 0.3924 m/s ²	0.1308
SMB type 225 [15]	Output noise = 3.5 mg	0.0343
SMB type 227 [16]	Output noise = 5.0 mg	0.0491
Kistler [17]	90% = 0.23 m/s ²	0.0123

Relative speed sensors that evaluate the velocity of surrounding vehicles constitute another group of measuring devices. As in the case of distance sensors, specific parameters and accuracies of those sensors are shown in Table 1. Additionally to sensors that monitor the position of adjacent vehicles, CACC requires information on the speed and acceleration of the ego vehicle. In the case of ego

vehicle velocity measurement, two options may be considered: electromechanical (accuracy depends on the quality of the vehicle's tires) or GPS-based devices.

To ensure a state-of-the-art approach to velocity measurement, electro-mechanical sensors are taken into account. For the purpose of simulations, it has been assumed that the device will be an electro-mechanical sensor with accuracy that is proportional to the actual speed of the vehicle, with the proportionality factor of $k = 1.0\%$ (the measured value is higher than the actual one).

The CACC formula considered takes also into account the acceleration rate of the platoon leader and the preceding car, so acceleration sensors should be used as well. For acceleration measurements, only high-precision GPS-based sensors are used (Table 1).

Due to the large number of devices to be tested, sensors representing the specific parameters (mean value and standard deviation) were identified. Mean errors and standard deviation values for all tested sensors are summarized in Table 2.

Table 2
Parameters of the sensors used

Sensor	Mean error	Standard deviation
Distance sensor	0	0.2794
Rel. velocity sensor	0	0.1439
Velocity sensor	1%	0
Acceleration sensor	0	0.0410

3.3. Simulation Model

Before we present the results of our simulation investigations aimed at determining the influence of imperfect sensors on the platoon's performance, we discuss the simulation assumptions, platoon structure, as well as the parameters of the so-called jamming vehicle considered in the simulations. The role of the jammer vehicle is to disturb the fluent movement of the platoon. Its velocity varies in accordance with a predefined pattern in 30-second cycles (looped for the course of the simulation). Each cycle lasts 30 s and consists of three phases: braking (speed decreases from 130 km/h to 30 km/h), acceleration (acceleration factor equal to 1.5 m/s², with the speed reaching 130 km/h again) and, finally, constant speed travel for approximately 2.04 s.

3.4. Actuation Lag

After the acceleration value has been calculated by the ACC/CACC controller, it needs to be used by the system. This, however, is not done instantaneously, but with a delay referred to as actuation lag. In the simulator, this delay is modeled as a first order low-pass filter applied to the output of the cruise controller:

$$P(s) = \frac{1}{(\tau s + 1)}, \quad (1)$$

where τ is the time constant of the filter. In [18], a list of factors that affect actuation lag in the case of a human-driven vehicle is presented. Taking all these factors into account, we can conclude that time constant τ could be as high as 500 ms.

Recent papers [19], [20] indicate that in the case of human-driven vehicles the total lag value could be decreased to approx. 200 ms. Moreover, a further decrease in actuation lag to a few tens of milliseconds is expected [8]. Therefore, two different actuation lag values, i.e. 20 and 200 ms, are investigated here.

3.5. Simulation Assumptions

In the considered scenario, we investigated a homogeneous platoon traveling in one lane of a highway. The first car in the platoon is the so-called leader and is preceded by another vehicle that interrupts the fluent movement of the platoon – the so-called jammer. The speed of the jammer changes (periodic breaking and acceleration), affecting the platoon's movement. With the help of control mechanisms, vehicles in the platoon attempt to respond to the jammer's behavior by adjusting their speeds and accelerations, in order to maximize platoon performance (i.e. they attempt to minimize the length of the platoon) and to avoid crashes. Within the platoon, two controlling algorithms are considered. One is the adaptive cruise control (ACC) algorithm (used by the platoon leader) and the other one is the CACC algorithm used by the remaining vehicles.

In the ACC algorithm, the speed of and distance to the preceding car are measured by built-in (or onboard) sensors, while in CACC, in addition to sensor measurements, messages from other vehicles that are transmitted by radio are relied upon. In this work, communication between the vehicles is performed based on the IEEE 802.11p standard.

3.6. Simulation Parameters

The set of parameters valid for all simulation runs and the configurations parameters used in IEEE 802.11p are presented in Table 3.

In the simulation, sensor inaccuracies are investigated separately to determine their impact on the platoon's performance. Then, all inaccuracies considered are combined to obtain the overall impact of all inaccuracies on the platoon's performance.

The performance of the platoon with the sensors adopted was evaluated in terms of mean inter-vehicle spacing. Inter-vehicle spacing corresponds to the bumper-to-bumper distance measured, averaged over time, for all pairs of subsequent vehicles in the platoon, and for all simulation runs over the chosen target CACC distance. It is expected that the average inter-vehicle spacing closely approximates the target CACC distance, i.e. the minimum target distance that guarantees collision-free performance of the platoon.

Moreover, it needs to be mentioned that we used, in the simulation, an improved version of the CACC algorithm introduced in [8], referred to as modified CACC, offering

Table 3
Simulation and IEEE 802.11p configuration parameters

Parameter	Value
Number of simulation runs	100
Simulation time	900 s
Platoon size	10 cars
Engine actuation lag	20/200 ms
ACC headway time	0.2 s
BSM message size	300 bytes
CACC message size	16 bytes
Spectrum band	ITS-G5A(5895–5905 MHz)
Channel estimation	Ideal
Channel model	ITU Vehicular-A
Path loss model	Winner+ B1 LOS
Antenna height	3.5 m
Shadowing distribution	Log-normal
Shadowing std. dev.	3 dB
Decorrelation distance	25 m
TX power density	23 dBm/MHz
Noise power	−174 dBm/Hz
Noise figure	9 dB
TX/RX antennas	1/1
Power control	Off
MCS	QPSK, $R = 1/2$

significantly better performance that is reflected by lower mean inter-vehicle distances.

3.7. Sensors Inaccuracies

The first part of the investigations performed is related to the impact of sensor quality on the behavior of the platoon. Figures 2 and 3 show the mean distances between platoon vehicles, assuming that the vehicles are equipped with distance and velocity sensors used in the CACC algorithm. These distances are calculated as the average of all bumper-to-bumper distances between the vehicles in the platoon, measured throughout the entire simulation period. The results are compared with the reference values which correspond to the situation in which all sensors work perfectly.

The introduction of inaccuracies results in a slight degradation of the platoon's performance. The said degradation equals approx. 0.2 m for the distance sensor and 0.4 m for the speed sensor in the case of the 20 ms scenario, and 0.3 m for the distance sensor and 0.15 m for the speed sensor in the case of the 200 ms scenario.

Another sensor that has been analyzed is the in-vehicle speed sensor. The mean distances obtained for different actuation lag values are shown in Fig. 4.

Based on the above results, it can be clearly stated that in the case of velocity sensor, the inaccuracies exert a nega-

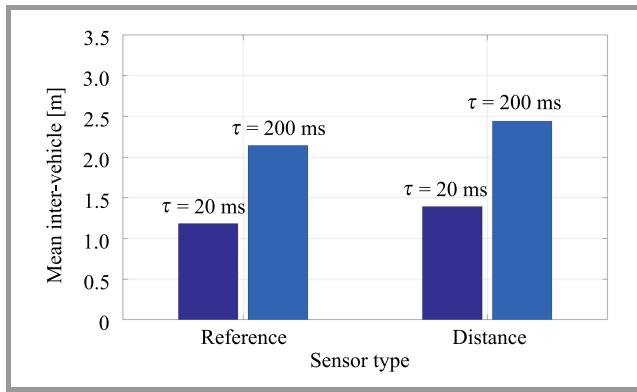


Fig. 2. Mean platoon inter-vehicle distance for distance sensors used in the CACC algorithm.

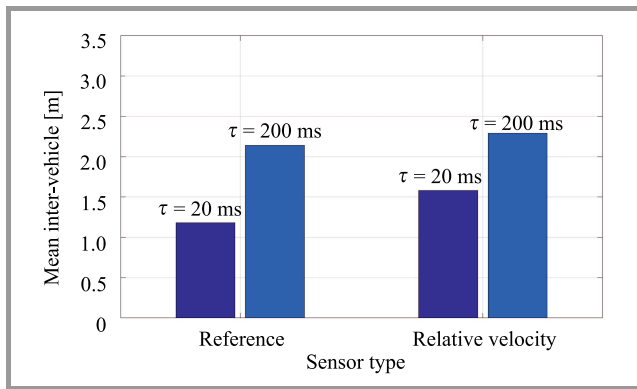


Fig. 3. Mean platoon inter-vehicle distance for relative speed sensors used in the CACC algorithm.

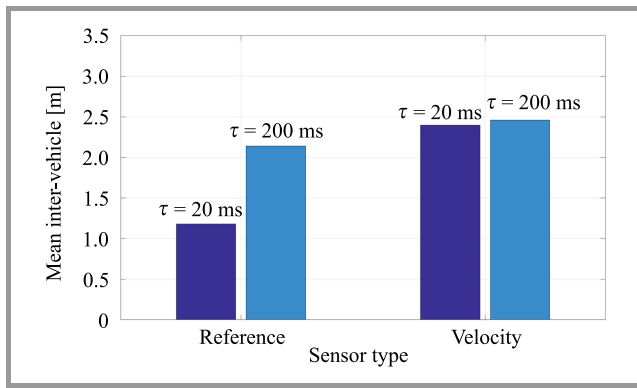


Fig. 4. Mean platoon inter-vehicle distance for speed sensors used in the CACC algorithm.

tive impact on the behavior of the platoon. This impact is reflected by a significantly higher mean distance between vehicles, which increases by 1.28/0.26 m for 20/200 ms scenarios, respectively.

The last device (Fig. 5) that has been analyzed is the vehicle acceleration sensor which determines the instantaneous acceleration factor of the ego vehicle. According to the literature, as such a unit should operate based on GPS data, it may achieve very good parameters. Tests performed with such an acceleration sensor show that the target distance

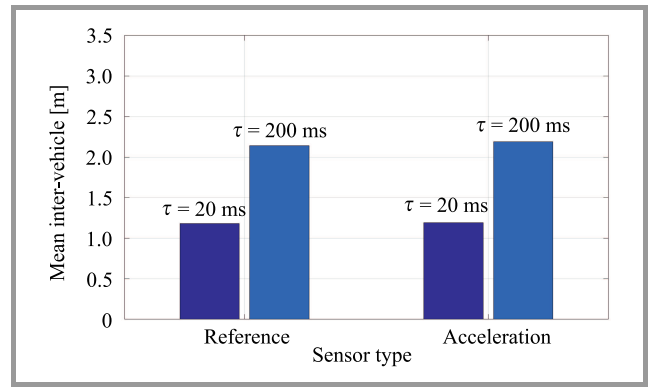


Fig. 5. Mean platoon inter-vehicle distance for acceleration sensors used in the CACC algorithm.

is only slightly affected, even for the lowest-quality sensor. Moreover, the mean inter-vehicle distance deterioration observed is as low as 0.01 m, compared with the reference values.

Although the presented acceleration sensors are characterized by great accuracy, their largest drawback is that they need to be within range of the GPS signal. Thus, these sensors might experience some difficulties in the so-called urban canyon scenarios or in tunnels. To overcome this, additional post-processing of the velocity data will be required.

Finally, the combined effect of all sensors on the behavior of the platoon was also analyzed, and distances between vehicles within the platoon were determined (Fig. 6). The results obtain clearly indicate that the quality of the devices used exerts a great impact on the platoon. Even the best quality units affect the platoon performance significantly, causing an increase in the observed mean distance between vehicles, equaling one meter.

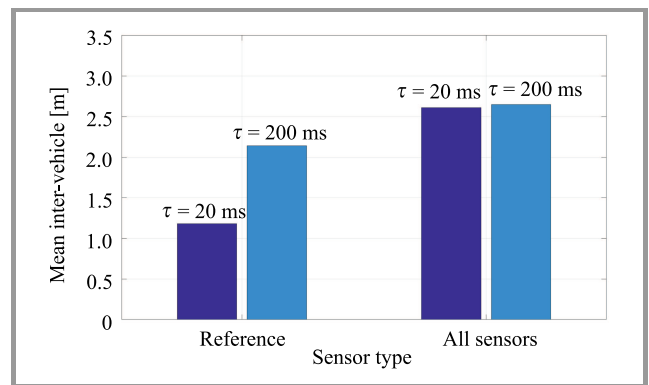


Fig. 6. Mean platoon inter-vehicle distance for all sensors used in the CACC algorithm.

4. Challenges Affecting Wireless Communications

V2V communications are usually proposed to be performed using cellular networks (e.g. LTE, 5G) or IEEE 802.11p

systems. However, it was observed that these technologies use fixed wireless channels, which lowers reliability when a high number of users (cars, trucks, platoons) is present.

A potential solution is to base intra-platoon communications on the so-called white spaces, i.e., frequencies that are unused by the licensed system at a given location, at the time of the platoon's arrival, such as the cognitive radio concept. This approach to spectrum usage is known as dynamic spectrum access (DSA). Its application is motivated by the number of spectrum occupancy measurements carried around the world. It has been revealed that 50% to 80% frequencies in the range of up to 6 GHz are unused. Therefore, potential frequency resources that may be reused exist. If a given frequency band is currently not occupied by a specific licensed system, it may be used for secondary system transmissions (without a license to use this band) [21]. In our case, the role of the secondary system may be played by the platoon of vehicles. Moreover, the higher the frequency, the more white spaces may be expected. Carrier frequencies of above 6 GHz and DSA may be considered for short-range V2V communications [22].

In the case of V2V communications, access to timely and accurate information about the surrounding environment may be very important. When vehicles are aware of the status of other objects on the road ahead of them, they can change the route or at least modify the driving parameters to reduce fuel consumption. Decreasing the probability of a crash is of the highest importance. However, in the context of DSA for V2V communications, availability of specific context information databases is required. The databases will store various data structures depending on the applications. In order to enable efficient management of the spectrum resources, information about transmissions using licensed systems and about secondary users (platoons utilizing DSA) is needed. A scenario may be imagined in which the platoon leader communicates with the database providing suggestions as to which frequency band should be used, taking into account interference received from and caused to other platoons. However, information about spectrum usage is the simplest of potential applications of such a database. It is envisaged that context information databases may store much more information, with their range varying from the number of trucks in each convoy, to their speed, to the current length of the platoon. This may be a source of typical speeds achieved on a given road depending on weather conditions. The structure of the database system should be, most likely, hierarchical. Long-term information about the environment may be stored in remote databases, while short-term information is stored in their local counterparts [23].

4.1. Sensor Inaccuracy vs. Information Available in Databases

As shown in Section 3, sensor inaccuracy directly impacts the performance of CACC. Error-affected information pro-

vided by the sensors, combined with limited reliability of wireless links may render effective platooning impossible due to an excessive number of errors/variances in the information fed to the CACC control algorithm. Therefore, novel solutions have to be proposed, such as dynamic spectrum access for V2V transmissions. If raw sensing information is to be exchanged between vehicles, the mm wave band, offering a higher bandwidth, may be considered [22], [24]. In such a case, throughput required along the links between vehicles may increase to hundreds of Mbps, while requirements concerning latency and reliability remain the same. Although the millimeter wave band allows for wider bandwidth, it poses serious challenges as well, e.g. reduces the transmission range. Therefore, lower frequency bands may be considered for intra-platoon communications as well [22]. In any frequency band of choice, a dynamic spectrum management mechanism is needed enable the selection of a frequency band that guarantees the best possible transmission conditions. The spectrum management mechanism may consider a licensed spectrum for inter-platoon communications as well. Proper selection of the frequency band may be aided with context information stored in dedicated databases. Multiple parameters stored therein may be used to improve the frequency selection procedure.

5. Conclusions

This article discusses various aspects of vehicle platooning, in particular the impact that inaccuracy of onboard sensors on the performance of a CACC-based platoon. Additionally, it considers sensing inaccuracy and limited reliability of wireless communications. Given the limited accuracy of the sensors, challenges for future V2V wireless communication have been formulated, including restrictions concerning delay, reliability and perceived collaboration. A suggested potential solution to increase CACC reliability is to use dynamic allocation of numerous frequency bands, including the millimeter wave band, to improve data transmission speed and to reduce the load of current ITS bands.

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An Expert System for Automatic Classification of Sound Signals

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Abstract—In this paper, we present the results of research focusing on methods for recognition/classification of audio signals. We consider the results of the research project to serve as a basis for the main module of a hybrid expert system currently under development. In our earlier studies, we conducted research on the effectiveness of three classifiers: fuzzy classifier, neural classifier and WEKA system for reference data. In this project, a particular emphasis was placed on fine-tuning the fuzzy classifier model and on identifying neural classifier applications, taking into account new neural networks that we have not studied so far in connection with sounds classification methods.

Keywords—audio descriptors, bird species, fuzzy classification of audio signals, MPEG-7, spectral features of sound.

1. Introduction

Automation of sound recognition processes is an important part of modern-day technology, widely applied in different areas, such as, for instance, automatic speech-based emotion recognition [1] and speech analysis [2]. Nowadays, with well-developed human speech recognition techniques available, work is also performed on automatic recognition of musical instruments. Here, we focus on developing specific methods for automatic recognition of animal species. We commenced with bird species [3] that are recognized based on the sounds they generate. The methods may find practical application in agriculture, where birds often pose a significant threat to crops (fruit, vegetables, cereals). Proper identification of bird species present in the area where the crops are grown is a prerequisite for the application of appropriate mitigation measures.

Another aspect that inspired us to conduct this research is the potential use of this method for classifying bird species to support the work of ornithologists and foresters aiming to quantify bird populations inhabiting a given area. Hopefully, the methods we develop may also be relied upon to automatically classify sounds in other fields, for instance in speech recognition [4]. The authors have already achieved considerable success in recognizing musical instruments [5], [6] and other sounds generated by technical devices. This project constitutes a subsequent stage of the research activities planned. Both methods are relevant for agriculture, wind farms and airport monitoring. A bird identification method based on camera snapshots

and relying on deep learning techniques was presented in papers [7], [8]. Research [9] used deep learning techniques while processing images from an unmanned aerial vehicle. A classification based on machine learning is presented in [10]. On the other hand, in [11], decision trees were used. Here, approaches that differ from the ones described previously are presented. We propose to classify bird species by using such methods, as neural networks and fuzzy logic with sounds descriptors.

The first section contains a short introduction to problems faced while recognizing bird species. Section 2 is devoted to the research methodology applied. It contains information about the MPEG7 standard, the methodology used to detect bird sound features and a proposal to develop a proprietary research tool. It also presents the result of bird species classification work supported by some classification algorithms included in the WEKA package. It also shows research results obtained with the use of the neural classifier. Section 3 presents the concept of a linguistic audio signal classifier. The last two sections summarize the research conducted. A preliminary model of an expert system is also proposed.

2. Study Methodology

This work focuses on the concept of building an expert system, and on achieving some novel results in the area of using neural networks for the recognition of bird species based on their sound. In order to perform the study, the

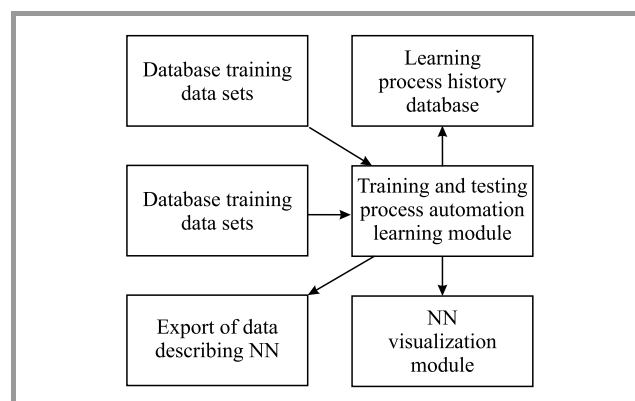


Fig. 1. Development of a proprietary research tool.

Table 1
Results retrieved for the WEKA classifier

	Corn crake	Hawk	Black- bird	Chiff- chaff	Eurasian pygmy owl	Meadow pipit	Lesser whitethroat	Firecrest	Cuckoo	House sparrow
Used k-NN, cross-validation method	98	91	69	84	96	91	98	96	100	78
Used random forest, cross-validation method	98	87	80	82	98	93	100	93	98	80
Used J48 decision tree, cross-vali- dation method	95	80	76	84	95	85	100	87	87	67

authors developed software supervising the learning process of neural networks. The concept of the solution under development is shown in Fig. 1. The development of a test tool facilitated full control over the learning process, offering the ability to relatively quickly assess how effective the specific networks were in the classification of the samples tested. The study itself was conducted using a three-layer neural network with three different neural activation functions. In the first phase, input vectors were determined by using low-level descriptors (LLD) from the MPEG-7 standard [12], [13]. The objective was to identify the vector of traits that will be best for automatic classification of bird species based on the sounds they record and produce. The methods developed in [3] have been used to determine window length in the frequency domain. Such a parameter is understood here as a fragment of the signal in the time domain, which was downloaded at the same time. The length of the window is determined by the spectrum resolution:

$$f_r = \frac{f_s}{n}, \quad (1)$$

where: f_r – adopted spectrum resolution (10 Hz), f_s – sampling frequency (44,100 Hz), n – number of samples for the experiment (4410).

If the tested sound is shorter than the window length ($n = 4410$), then the missing values are filled with zeros [5], [6]. In addition, the Blackman window was used for this study. Based on the MPEG-7 standard for audio signals, we have used the spectral centroid (SC) parameter describing the domain of signal frequency for the spectrum:

$$SC = \frac{\sum_{i=0}^n A(i) \cdot i}{\sum_{i=0}^n A(i)}, \quad (2)$$

where: $A(i)$ is the amplitude of the i -th partial (harmonic) and i is the frequency of the i -th partial.

The irregularity of spectrum Ir may be expressed as:

$$Ir = \log \left[20 \sum_{i=2}^{N-1} \left| \log \frac{A(i)}{\sqrt[3]{A(i-1) \cdot A(i) \cdot A(i+1)}} \right| \right], \quad (3)$$

where: $A(i)$ is the amplitude of the i -th partial (harmonic) and N is the number of the available harmonic.

Three characteristics were obtained from each spectrum: SC , Ir and Sum – the sum of all amplitudes in the spectrum. The combination of descriptors gives a 24-component feature vector that describes the bird sound.

2.1. Results obtained for the WEKA System

Reference values obtained from the popular WEKA system were used for testing and validation of neural classifier's capabilities. The methods used included the following: k -nearest neighbors, random forest, Jrip.

Table 1 shows the performance of the WEKA classifier while working on samples retrieved from 10 bird species. The numerical values in the table represent the recognition of the individual bird species.

2.2. Results Obtained for the Neural Classifier

Paper [3] shows that the characteristics vectors obtained by using the defined descriptors contain information allowing the classes that correspond to the bird species whose sound recordings were analyzed to be identified. Networks consisting of 2 and 3 layers and using selected activation functions were analyzed in the study. The learning set had 50 characteristics vectors (50 sets of input signals – expected output values) for 10 classes. The aim of the tests was to identify, in practice, the level of separation (distinction) of the particular classes corresponding to individual bird species. The tests were to determine the further direction of work, i.e. to select and modify the outcomes of the initial sample preparation phase (with sound descriptors included). The learning process was carried out using the Delta rule [14], [15].

$$r = \sigma = [d_i - f(w_i^t x)] f'(w_i^t x), \quad (4)$$

$$\sigma w_i = c(d_i - y_i) f'(w_i^t x), \quad (5)$$

where: r – signal learner, σw – correction weights, c – constant learning process, d – error learning, y – matrix outputs, active function, x – matrix inputs.

The mean classification efficiency for the tested activation functions equaled 78%. It also needs to be noted here that such results were achieved for a relatively small learning set, containing 50 samples only. For the average best performing activation function:

$$f(x) = \frac{1}{1 + e^{-\beta x}}, \quad (6)$$

the average of 1200 learning cycles was required. An important observation made at this stage is that some species were identified in a unanimous way, and some were "distorted" by other input data sets (belonging to another species), as shown in the following graphs generated for Eq. (6).

Therefore, we used a modest set of samples for the teaching set. The results obtained allow us to assume that this method is effective in classifying sound signals of various bird species. With a larger set of learners, the achieved level of recognition should be higher. The results obtained from the application of neural networks in various configurations, with different neural activation functions [3] prove to be useful for the development of an expert system. Furthermore, assuming that the expert system can also be deliberately developed as a sensory network (extensive area applications), the change/programming of the scope of actions may be limited to updating the value of the neurons' weights only. This will be a major advantage in the case of low-energy networks, for example. Hence, the use of neural networks to build one of the modules of the expert system seems to be justified.

2.3. Linguistic Audio Signal Classifier

In this research, we aim to develop a methodology for designing an expert system, with one of its modules based on fuzzy logic [16]. In previous studies, in order to develop a fuzzy classifier of sound signals, fuzzy sets were defined as an acceptable range of a given characteristic for the particular bird species. As a result, the values represented by triangular fuzzy sets [3], [16] indicating individual bird species are obtained within the range of a given linguistic variable. The fuzzy sets for the classifier inputs indicating the individual bird species are defined as:

$$\begin{aligned} Bird_i &= \Lambda(x; x_{mean} - 2\Delta_L, x_{mean} + 2\Delta_L) \\ \Delta_L &= X_{mean} - x_{min}, \Delta_L = x_{mean} - x_{min} \end{aligned} \quad (7)$$

where: i – number of bird species x_{min} , x_{max} , x_{mean} – the minimum/maximum/mean value of the descriptor for a given species.

An assumption that is not usually used in fuzzy systems is a feature that is specific of the model adopted. In linguistic variables representing audio descriptors, we ignore the completeness and continuity of the model. We also do not assume that the values of the membership functions of adjacent fuzzy sets add up to unity [17], [18]. the introduction of a sufficiently wide media for the fuzzy sets reduces the risk of error in the operation of the proposed fuzzy system.

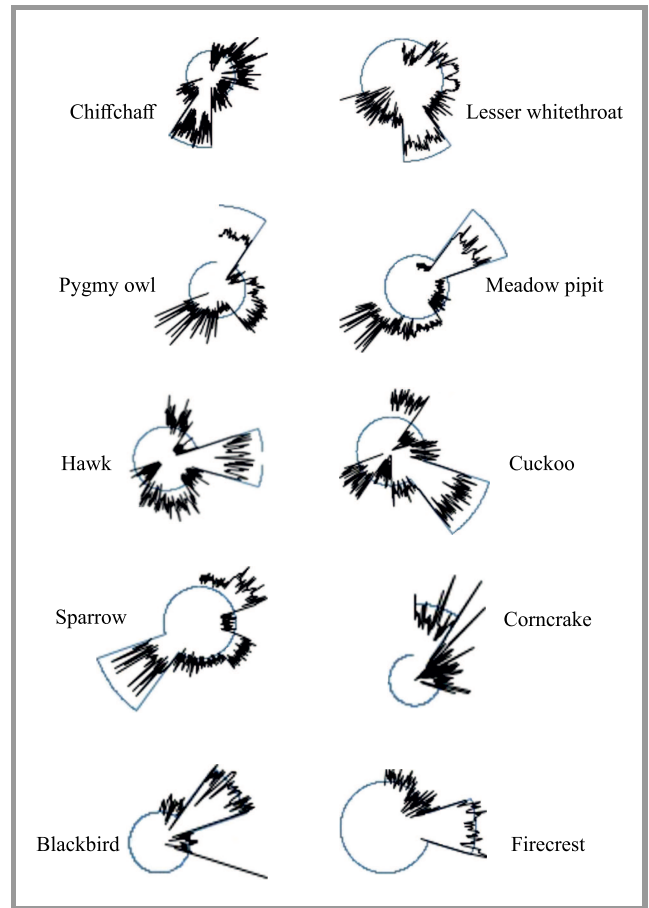


Fig. 2. Effectiveness of the neural classifier in identifying selected bird species.

Hence, the double delta defining the left- and right-hand side fuzzy bank in Eq. (5). Although such an omission of the recommended properties of fuzzy systems seems to be problematic, it is worth noting that a similar approach has already been successfully applied to the classification of lower subspecies of irises [19]. The adopted rule model is defined in the following way:

$$\begin{aligned} \text{If } i_1 \text{ is } B_1 \wedge i_2 \text{ is } B_1 \wedge \dots \wedge i_i \text{ is } B_1 \\ \text{Then } B_1 \text{ is } rec \wedge B_2 \text{ is } rej \wedge \dots \wedge B_i \text{ is } rej \end{aligned} \quad (8)$$

where: i_k – input number k , B – bird (species) number, rec – fuzzy set for the recognition and rej – for the rejection.

The resulting concept of building a modular fuzzy classifier has an advantage consisting in the form of a high scalability of the solution, as the existing structures does not have to be disturbed while adding more species of birds. the expansion may consist in simply adding another fuzzy set of blurred characteristic sets of the new species and in adding a rule that recognizes the new data class. The tests conducted so far have yielded results whose effectiveness ratio equals 65%. These results are not excellent, but we are currently conducting research on improving the data (descriptors) preparation stage and we assume that fuzzy and neural classifiers will complement each other as a result of higher/better classification results.

3. Expert System Concept

Figure 3 shows the modular structure of the designed expert system. It may be used in different hardware implementations, i.e. single-board computers and microcontroller-based embedded systems. The fuzzy logic module is to be an element that complements the neural classifier [20].

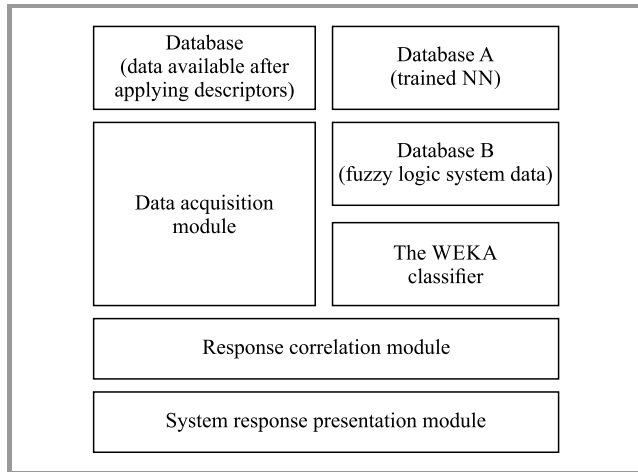


Fig. 3. Concept of an expert system.

4. Conclusions

This work summarizes the second stage of research focusing on sound recognition. In the first stage, the effectiveness of specific descriptors used in the MPEG-7 standard is identified. The present stage focuses on developing methodologies and models required for the expert system. Work planned in the future will allow to develop and test a prototype of the expert system and search for other effective descriptors, also for identifying objects other than bird species. In the future, we intend to develop methods and tools for the construction of expert systems, as indicated in [13], and applications that use the so-called deep learning techniques, for example those described in [20].

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Retransmission as a Replacement Forward Error Correction in Noiseless LV Networks used for Narrowband PLC

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Abstract—The article describes a project concerned with deploying PLC technology in railway light signals. The proposal increases reliability of communication between railway light signals and the railway automation center, relying on frame retransmissions instead of the FEC technique. The paper presents the results of long-term FER measurements performed in a narrowband PLC-based communication system. The said results are used as communication reliability metrics. Deployment of dual channels in order to increase communication reliability is discussed as well, as is efficiency of FEC-based convolutional coding. The results are verified in a real world environment.

Keywords—FEC, FER, narrowband PLC, PSK.

1. Introduction

Multichannel support is a receiver feature enabling the device to listen to a number of channels simultaneously, until the carrier signal is detected. Thereafter, only the channel on which the carrier signal was detected is considered for further data reception. After data has been collected or the time limit has expired, the receiver returns to the multichannel carrier detection mode. The multichannel option is a common feature of short-range communication devices, both of the wired and wireless variety [1]–[3].

This article presents a communication system that is based on the narrowband power line communication (PLC) technology implemented in the ST7580 chip [3]. The ST7580-based modem allows the receiver to operate in the single or dual channel mode. It forms a part of an LED-based signaling device used for rail traffic control. LED light signals used in railway traffic are more reliable due to the fact that they communicate with railway automation centers which control and monitor signal states. In the past, current flow was the only parameter monitored. Hence, reliability the signal status monitoring process was lower. It was simply assumed that if the current flows, the signal is working. Unfortunately, this was not always the case. The PLC modem allows to use the railway signal's power line both for power supply and data transmission purposes. Here, the use of wireless technology is irrational because of three reasons: security, long communication dis-

tances (excessively long for radio communications due to time delay restrictions characterizing the multi-hop technique [4]–[6], and strict reliability requirements. The use of other communication media, e.g. optical fibers, is linked with high investment costs.

The packet used for monitoring the state of railway traffic light signals comprises data about LED current and voltage, unit address and a control field. It is more than 100 bytes long and must be delivered to the railway automation center within 300 ms maximum. If not, the railway traffic on a given track is stopped due to security reasons. Hence, the new solution to be used in railway traffic signaling systems needs to offer a more reliable method of assessing the condition of light signals. Remote detection of signal damage is a welcome feature as well.

2. Communication System Requirements and the Problem Definition

The PLC modem is controlled by a microcontroller via UART operating at 57.6 kbps. To shorten the time of booting the system after power-up, the modem is not configured, meaning that it works with a default line frame format shown in Fig. 1.

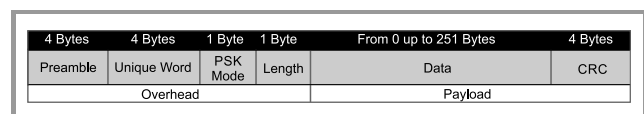


Fig. 1. Default format of the PLC line frame.

The length field defines the data volume plus the CRC code. The frame overhead is modulated using the most robust modulation possible, i.e. B-PSK coded modulation, where “coded” means that forward error correction (FEC) based on convolutional error correction coding is enabled. The potential modulation modes, together with their bitrates, are shown in Table 1, according to [7].

The duration of transmission of a 10 byte frame overhead is: $10 \times 8 \text{ bits} : 4800 \text{ bps} = 16.7 \text{ ms}$. The time required for communication between the microcontroller and the mo-

Table 1
Modulation modes and bitrates available in the ST7580 chip

Modulation	B-PSK	Q-PSK	8-PSK	B-PSK coded	Q-PSK coded	B-PSK coded with PNA
Bit rate	9.6 kbit/s	19.2 kbit/s	28.8 kbit/s	4.8 kbit/s	9.6 kbit/s	2.4 kbit/s

Table 2
Time values for the state monitoring communication frame

Modulation	B-PSK	Q-PSK	8-PSK	B-PSK coded	Q-PSK coded	B-PSK coded with PNA
On line state frame duration	117 ms	67 ms	50 ms	217 ms	117 ms	417 ms
State frame communication time	177 ms	127 ms	110 ms	277 ms	177 ms	477 ms
State frame delivery time after powering the signal up	202 ms	152 ms	135 ms	302 ms	202 ms	502 ms

dem is another design parameter that needs to be taken into consideration. The railway light sign state monitoring packet is 116 bytes long. With additional of the remaining bytes of the local frame including two frame check sequence (FCS), it is 122 bytes. With UART working at 57.6 kbps bitrate and with one start and one stop bit, the local transmission time would be: $122 \times (1 + 8 + 1)$ bits : 57600 bps = 21.2 ms. This time should be doubled due to using duplex in communication and then increased by the time needed for the modem initialization. All together it gives: $2 \times 21.2 \text{ ms} + 18 \text{ ms} = 60.4 \text{ ms}$. Another important time is LED checking time (25 ms). The time of payload data transmission depends on the modulation used and length. The time values depend on the modulation used are summarized in Table 2.

Table 2 also shows the time elapsed from power-up the sign to the state frame delivery to the railway automation center. It increases the communication time by 25 ms. Figure 2 consists of oscillogram taken during the transmission of the state frame modulated with 5 types of available modulation of 5 state frames.

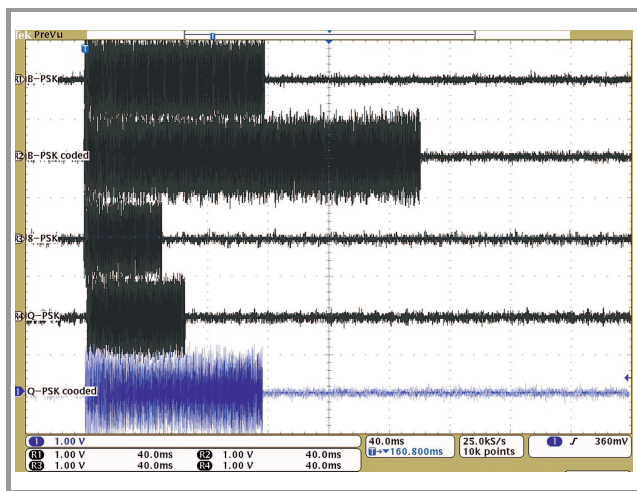


Fig. 2. Oscillogram of five frames: B-PSK, B-PSK coded, 8-PSK, Q-PSK and Q-PSK coded.

B-PSK coded [8], being twice slower. All frames presented in Fig. 2 carry the same information, even though their duration is different. The duration of Q-PSK coded frame is the same as B-PSK frame, while the Q-PSK coded modulation is more robust than B-PSK, so that, the Q-PSK coded modulation disqualifies B-PSK.

3. Modulation Selection

To meet the time and reliability requirements, that monitoring data must be delivered to the railway automation centre up to 300 ms. In real the time available for communication is decreased by 25 ms (LEDs checking time), i.e. is 275 ms. Analysing data from Table 2, it may be concluded that B-PSK coded and B-PSK coded with PNA modulations cannot be used because they are too slow in terms of throughput. From the four types of remaining modulations to be chosen the most reliable is Q-PSK coded (see Fig. 3) where two charts of FER vs. SNR for Q-PSK coded and B-PSK are shown.

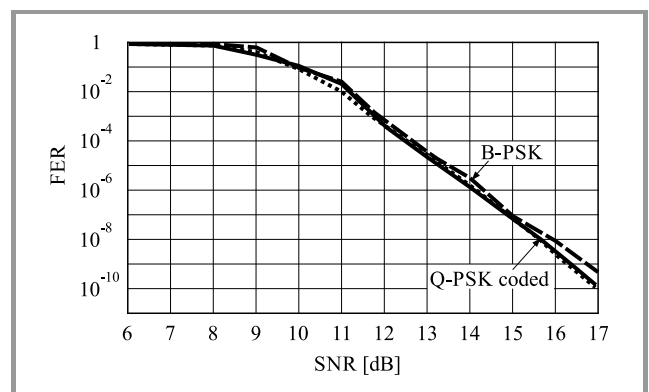


Fig. 3. FER vs. SNR for 37-bytes frames in B-PSK and Q-PSK coded modulations.

Finally, B-PSK was selected for the comparison, as B-PSK modulation has the same bitrate as the Q-PSK coded solution. Subsequently, three modulation types, namely 8-PSK,

Q-PSK and Q-PSK coded were considered, with Q-PSK coded being the most robust of them all, and with Q-PSK and 8-PSK characterized by an extra feature: they are fast and offer state frame transmission lead-times of 127 ms and 110 ms, respectively, meaning that the state frame may be communicated twice when Q-PSK or 8-PSK modulation is used.

4. Frame Retransmission vs. FEC

Retransmission is a method of achieving good reliability by sending the same information more than once. It is used in communication systems whose physical or logical links are unreliable [9]. The frame is retransmitted if the sender does not receive any handshake information. The sender may also retransmit the frames by default [10], without any additional confirmation needed. The FEC method implemented in ST7580 increases the length of the frame more than two-fold. Thus, two frames may be transmitted within the same period of time, using the same type of modulation, if the frames are not coded. In Fig. 4, two FER vs. SNR lines are shown for Q-PSK and Q-PSK coded.

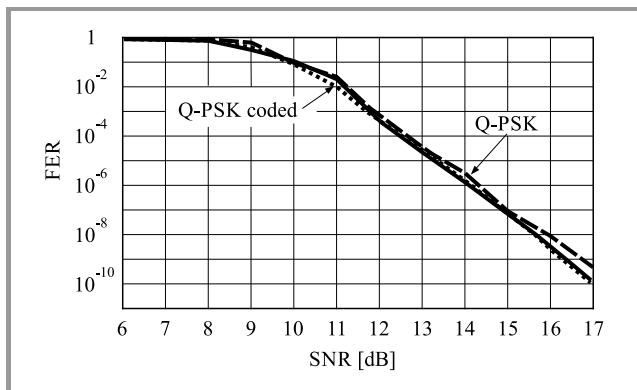


Fig. 4. FER vs. SNR for 37-byte frames: performance of Q-PSK and Q-PSK coded modulations.

From Fig. 4, one may see that FER is the error probability, and with the same noise level affecting the communication of both frame types, reliability is still worse while relying on retransmission rather than on FEC. With frame retransmission, FER is lower because the probability is the product of two components for each frame. Assuming that noise is a short-term phenomenon, FER may be lower. Assuming also that noise of the narrowband variety, deployment of the dual channel mode may be useful.

5. Single Channel and Dual Channel FERs vs. SNR in Improving Q-PSK Modulation Performance

To demonstrate the advantages that dual channel operation of the receiver offers in terms of the quality of communi-

cations, three FER vs. SNR charts are presented (Fig. 5). They depict the following: FER on a 72 kHz low band channel, FER on a 86 kHz high band channel and FER on a dual channel (72 and 86 kHz).

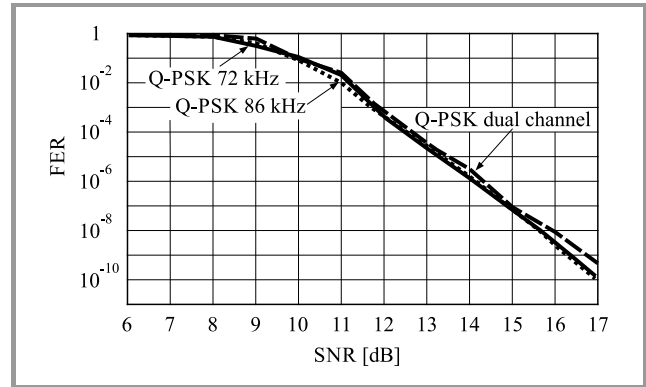


Fig. 5. FER vs. SNR for 37-bytes Q-PSK modulated frames, received in the following receiver modes: single low channel, single high channel and dual channel.

From Fig. 5, one may conclude that by turning on the dual channel mode in the receiver, transmission quality is lowered, but only slightly. Therefore, the dual channel option may remain turned on at all times, which allows the receiver to receive frames transmitted both on low and high channels.

6. Experimental Results

To check the efficiency of the proposed method, a mixed network was used, where old and new railway signal monitoring systems, i.e. those relying on the flow of current and PLC-based, were operated simultaneously. After power is supplied to the railway signal, it sends a Q-PSK coded state frame and current flow should be detected as well. During the period of several minutes over which the experiment lasted, three state frames were transmitted every second: Q-PSK coded (alternately over high/low channel), Q-PSK over high channel and Q-PSK over low channel. On the receiving side, one error free Q-PSK coded frame and at least one error free Q-PSK frame are expected every second. If they are not received, a counter is incremented. There are two counters: one for missed or erroneous coded frames (MECFC), and the other one for missed or erroneous non-coded two frames (MEN2FC). The measurement results obtained from ten LED signal-controller pairs during the experiment that lasted half a year show that the summarized MECFC is 141, and MEN2FC is 36. The total observation time was 6.5 million seconds. The average SNR level, calculated based on the number of error free and erroneous frames received, was 23.8 dB. For 8 signal-controller pairs, the reading of MECFC counters was always greater than the reading of MEN2FC counters, and for 2 pairs, the readings were the same and equaled 0.

7. Conclusions

This paper has demonstrated that the FEC technique based on convolutional coding is more effective than sending the same information twice within the same time period, but without relying on FEC. Activation of the dual channel receiver mode in the PLC modem does not improve transmission quality but allows the modem to transmit data over two communication channels. Therefore, it allows transmit two consecutive frames carrying the same data in two different ways. Based on experimental results, it may be concluded that using frame retransmission over different frequency channels renders the communication more reliable than when relying on FEC. This statement applies only to low voltage networks with low noise levels, such as signaling cables used in railway automation systems. Errors caused by external interference are rare. Due to its the short duration and narrowband character, the frame retransmission mechanism helps achieve good reliability levels. Future work will focus on the compression of data transmitted within state monitoring frames. By shortening frame length and applying 8-PSK modulation, retransmission could be used more than once, which would probably improve communication reliability even further.

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