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Preface

This issue of the *Journal of Telecommunications and Information Technology* aims at documenting state of the art research, new developments, and directions for future investigation in the field of wireless networks. The issue contains twelve articles of researchers at the beginning of their career as well as works of renowned scientists, presenting their contributions to a broad range of topics related to wireless networks. The topics cover subjects ranging from various applications of wireless networks, architectures of wireless mesh networks, analytical and simulation methods for modeling wireless networks, to new protocols and devices for wireless networks.

This issue is opened by the article *Ubiquity of Client Access in Heterogeneous Access Environment* by Krzysztof Gierłowski. It presents an extensive overview of new mechanisms and technologies used to provide ubiquity on network access in heterogeneous environment of today's access systems. It also takes the discussion of challenges which must be addressed while attempting to fulfill topics such as handover control and mobility management.

The problem of mobility and handover management is also addressed in the second article, entitled *Evaluation of the Delay-Aware NUM-Driven Framework in an Internetwork Environment*. The authors: Maciej Urbański, Mateusz Poszwa, Paweł Misiorek and Dario Gallucci present a new architecture for handover management in multi-service wireless mesh networks. The proposed solution consists of DANUMS (Delay-Aware Network Utility Maximization) and WiOptiMo systems. The systems cooperate by exchanging measurements of transmitted traffic in order to improve the network utility as well as to optimize seamless handovers. The high effectiveness of the proposed architecture has been evaluated in a dedicated test-bed.

Tomasz Bilski in his paper *New Threats and Innovative Protection Methods in Wireless Transmission Systems* discusses several issues related to latest innovations in wireless communications systems, from the point of view of new threats and new vulnerabilities of this innovations. At the same time these new technologies may be used as new means for data protection. Cryptography, physical layer security, energy usage, handover, secrecy capacity and out-of-band authentication issues have been raised in this article.

In the next paper, *Lessons Learned from WiMAX Deployment at INEA*, Karol Kowalik, Dawid Dudek, Michał Kołodziejski, Bartosz Musznicki, Eugeniusz Grzybek and Jacek Jarzina share their experience related to deployment of broadband WiMAX-based service in Wielkopolska region with the readers. The experience gained during the deployment of connectivity for Internet and telephony services to around 5,500 households across the 30,000 sq. km region concerns accuracy of theoretical propagation models, quality of service features in 802.16e standard, and techniques for throughput maximization in multi-path environment.

The authors of the article *DVB-T Channels Measurements for the Deployment of Outdoor REM Databases* focus on outdoor measurement of the spectrum occupancy in the TV band, in Poznań (Poland) and Barcelona (Spain). Adrian Kliks, Paweł Kryszkiewicz, Krzysztof Cichoń, Anna Umbert, Jordi Perez-Romero and Ferran Casadevall evaluated stability of the TV channels in both cities, during both drive-tests and indoor measurements. The obtained results allow for further works on TV White Space Communications Systems.

The present issue of the Journal also includes the article *Adaptive Algorithms Versus Higher Order Cumulants for Identification and Equalization of MC-CDMA* by Mohammed Zidane, Said Safi, Mohamed Sabri, Ahmed Boumezzough, and Miloud Frikel. The authors provides the results of a comparative study of blind and adaptive algorithms, elaborated for Multi-Carrier Code Division Multiple Access (MC-CDMA). The effectiveness of both group of algorithms has been evaluated in a simulation environment, by calculating bit error rate for different values of SNR.

A new method for traffic characteristics estimation in cellular systems with Wideband Code Division Multiple Access (W-CDMA) radio interface is proposed by Ioannis D. Moscholios, Georgios A. Kallos, Maria A. Katsiva, Vassilios G. Vassilakis, and Michael D. Logothetis in the article *QoS Equalization in a W-CDMA Cell Supporting Calls of Infinite or Finite Sources with Interference Cancelation*. The authors consider a multirate loss model for the calculation of time and call congestion probabilities in a W-CDMA cell. According to this model, the calculation of time and call congestion probabilities are based on approximate but recursive formulas, whose accuracy is verified through simulation experiments.

Test-beds, emulation and simulation environments for wireless networks are described in the next two papers. In the article: *On IPv6 Experimentation in Wireless Mobile Ad hoc Networks* by Monika Grajzer and Mariusz Głąbowski selected topics on performing IPv6 protocols experimentation in wireless, IPv6-only mobile ad hoc networks (including both simulation and testbed-based evaluation) are presented. The selection of open-source simulation environments is presented and the comparison of simulation and emulation experimentation methods is also provided.

The authors of the paper *Review of Simulators for Wireless Mesh Networks*, Piotr Owczarek and Piotr Zwierzykowski, present issues related to the simulation tools and the main advantages of simulation techniques. In order to help researchers in selection of an appropriate simulation environment, the authors present statistical information gathered during a literature survey of a number of research articles from the most popular publishers in which the selected simulators were used in initial system design.

The next two articles propose new routing protocols for wireless networks. Takuma Koga, Kentaroh Toyoda and Iwao Sasase in their article *Priority Based Routing for Forest Fire Monitoring in Wireless Sensor Network* propose a new protocol for forest fire monitoring system. The main aim of their study was to lower the probability of packets' loss and to lower end-to-end delay. These goals were achieved by elaborating a new priority policy routing methodology.

In the article entitled *On-demand QoS and Stability Based Multicast Routing in Mobile Ad Hoc Networks*, P. I. Basarkod and Sunilkumar S. Manvi propose quality of service and stability based multicast routing protocol for real time applications offered in MANETs. This on-demand protocol, called OQSMR, utilizes information sent between neighboring nodes, including node and link stability factor, bandwidth availability and delays. The effectiveness of the proposed protocol has been evaluated in a simulation environment.

In the last article *Design of a Superconducting Antenna Integrated with a Diplexer for Radio-Astronomy Applications*, Massimo Donelli and Pascal Febvre present the design of a compact receiving front-end diplexer, optimized for radio-astronomy applications. At the design stage of the diplexer, an evolutionary Particle Swarm Optimization algorithm has been applied. The authors fabricated the diplexer prototype which confirmed a good accuracy of numerical and experimental results.

Mariusz Głąbowski and Maciej Piechowiak
Guest Editors

Paper

Ubiquity of Client Access in Heterogeneous Access Environment

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Abstract—With popularization of mobile computing and diverse offer of mobile devices providing functionality comparable to personal computers, the necessity of providing network access for such users cannot be disputed. The requirement is further reinforced by emergence of general purpose mobile operating systems which provide their full functionality only with network connectivity available and popular XaaS (Everything as a Service) approach. In this situation and combined with the fact that most Internet-based services are able to function efficiently even in best effort environment, requirement of ubiquity of network access becomes one of the most important elements of today's computing environment. This paper presents a general overview of the the vast group of mechanisms and technologies utilized in modern attempts to efficiently provide ubiquity on network access in heterogeneous environment of today's access systems. It starts with division of users interested in ubiquitous network access into broad groups of common interest, complete with their basic requirements and access characteristics, followed by a survey of both already popular and new wireless technologies suitable to provide such access. Then a general discussion of most important challenges which must be addressed while attempting to fulfill the above goal is provided, addressing topics such as handover control and mobility management.

Keywords—handover, mesh networks, mobility, technological networks, ubiquitous access, wireless networks.

1. Introduction

Very high and still growing rapidly popularity of mobile end-user devices, along with their considerable robustness and functionality falling into range previously reserved only for personal computers, results in raising demand for means of easy network access for such devices [1]. Moreover, concepts such as XaaS (Everything as a Service) and architecture of popular operating systems designed for mobile devices make presence of such access still more critical for users, as its lack will result in significant available functionality reduction.

In this situation, network access ubiquity becomes one of the most important requirements for environments such as metropolitan areas, industrial installations or various personal and cargo transportation systems. Many new concepts, like Smart Cities, Smart Grids, assisted living or intelligent transportation systems, depend on its presence. One can argue, that currently the above requirement surpasses in its importance even the ability to maintain a high level of transmission quality.

Wireless network technologies play a crucial role as networks access technologies, as cable-based solutions tend to be of limited utility in case of easily portable or mobile devices. As a result, a number of popular wireless technologies emerged, staring with Personal Area Networks (i.e. as ZigBee), through highly popular Local Area Networks (for example: Wi-Fi installations) and ending with Wide or Regional Area Network installations (mainly 2G/3G/4G technologies). A high number of wireless systems, utilizing this assorted set of technologies, have been deployed by numerous operators in high demand areas, creating massively heterogeneous access network environment. Additionally, many supporting technologies were developed, i.e., broadband mesh networks (providing self-forming, highly resilient network structures and good radio coverage in varied environments) or cognitive radio solutions, allowing for much better efficiency in radio frequency resource utilization, by taking advantage of currently unused transmission channels owned by external systems – for example unused TV channels.

Unfortunately, this diverse set of access systems does not necessarily guarantee constant, uninterrupted network access. In fact, many additional functions should be provided to consolidate such a diverse collection of access systems (divided by both technological and organizational boundaries) and offer users an ubiquitous network access.

2. Ubiquitous Network Access Usage Groups and Requirements

The necessity of communication convergence and ubiquity of network access is driven by both "technology push" and "business pull" [2]. New devices, access technologies and protocols create wide range of possibilities which can be offered to a user. At the same time customer demand, lower entry barriers for infrastructure and service operators, new business opportunities lead to new installations development and new services resulting in further popularization of mobile computing technologies.

Users interested in ubiquitous network access can be roughly divided into three main groups: popular access, infrastructure systems and technological networks, special systems and environments.

The first, popular group of users is mainly interested in obtaining uninterrupted access to Internet resources. Such users require relatively low Quality of Service (QoS), but

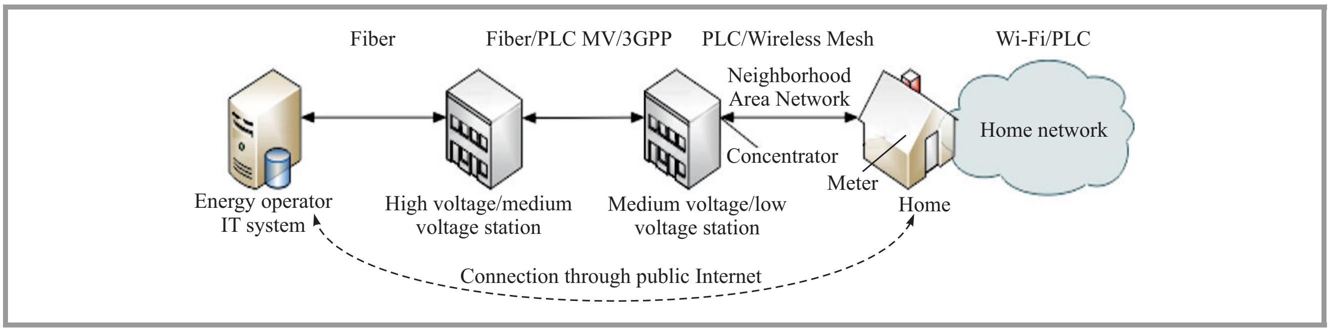

Fig. 1. Advanced Metering Infrastructure system as an example of technological network.

at the same time they are going to utilize wide variety of applications to different services access. Moreover, their subjective level of Quality of Experience (QoE) for a given QoS level depends not only on a requirements of a particular service they access, particular hardware and implementation they use, but also on their personal preferences and expectations. As such, providing very high QoS level (for example: hard QoS guarantees) is unnecessary, especially in case of Internet service implementations which are being developed for a network inherently lacking QoS guarantees.

At the same time, popular access systems need to correctly interface with a large and rapidly growing number of different client access devices. Fortunately, with emergence of universal, general purpose, mobile operating systems, obtaining software compatibility is much easier than in past years, when each hardware device utilized a dedicated firmware implementation.

From access system operator's view, popular user group consists of a potentially anonymous high number clients interested in obtaining access to a high number unspecified services, with comparatively low QoS requirements, from which throughput can be considered the most important. It is also worthy of mention, that in this user group, necessity of providing ubiquitous network access for mobile users can be considered both technologically simplest (due to low QoS requirements) and most rewarding, as not only many new Internet services are well prepared for handling connectivity parameters fluctuations frequent in mobile wireless environment, but mobility of users itself creates demand for new services – for example location aware solutions for navigation or micro-payments.

Infrastructure and technological networks can be considered an opposite end of the scale compared to popular users. They serve a well defined, closed user groups, interested in obtaining a highly reliable access to a strictly defined group of services. Specific QoS requirements can differ greatly, but they can always be precisely defined.

Energy distribution-related computer networks can serve as a good examples of technological networks. With such systems as smart grids [3], Supervisory Control and Data Acquisition (SCADA) [4], Distribution Automation (DA) and Advanced Metering Infrastructure (AMI) [5], energy-related systems are omnipresent in populated and technologically developed areas. Of course other examples of this type, such as: emergency communication systems, metropolitan transport control systems, bulk warehousing and transport support networks, building automation or Internet of Things deployments cannot be discounted.

As can be seen in Fig. 1, an example DS-AMI system is a complex deployment, consisting of data acquisition and processing center, which can be connected to energy transmission and distribution substations with a diverse set Wide Area Network (WAN) technologies. Taking into account that such stations are located over large geographical areas, creation of infrastructure can be a significant investment for even big companies, eased in some part by the fact that it can be co-located with energy distribution grid.

Elements of the communication network system located relatively close to end-users are created with use of different technologies. Neighborhood Area Network (NAR) responsible for providing data transmission capabilities between distribution stations and metering equipment at customer premises, most often utilize Power Line Communication (PLC) solutions [6], thus reusing already present power distribution installation, or Wireless Mesh Network (WMN), creating resilient, multihop, wireless communication system, which coverage area extends with each participating end-user device.

Such large communication networks, which, due to their very connection with power distribution grids are able to provide coverage in practically all technologically developed areas, can be a very well suited as means of providing infrastructure for ubiquitous network access solutions. At the same time, an opposite trend can also be observed – instead of creating such complex systems, expensive in both creation and maintenance, it becomes a popular solution to utilize already present communication infrastructure in place of described structure chosen elements. One of the most popular examples include use of public EDGE/UMTS operator data services in place of WAN infrastructure. There is also high interest in idea of utilizing an already existing, general purpose Internet access present at customer premises for creating direct link between metering equipment and central data acquisition center.

From the above example, it is evident that technological networks can be seen as both efficient provider and highly interested client of ubiquitous network access solutions. However, regardless of the choice between these two

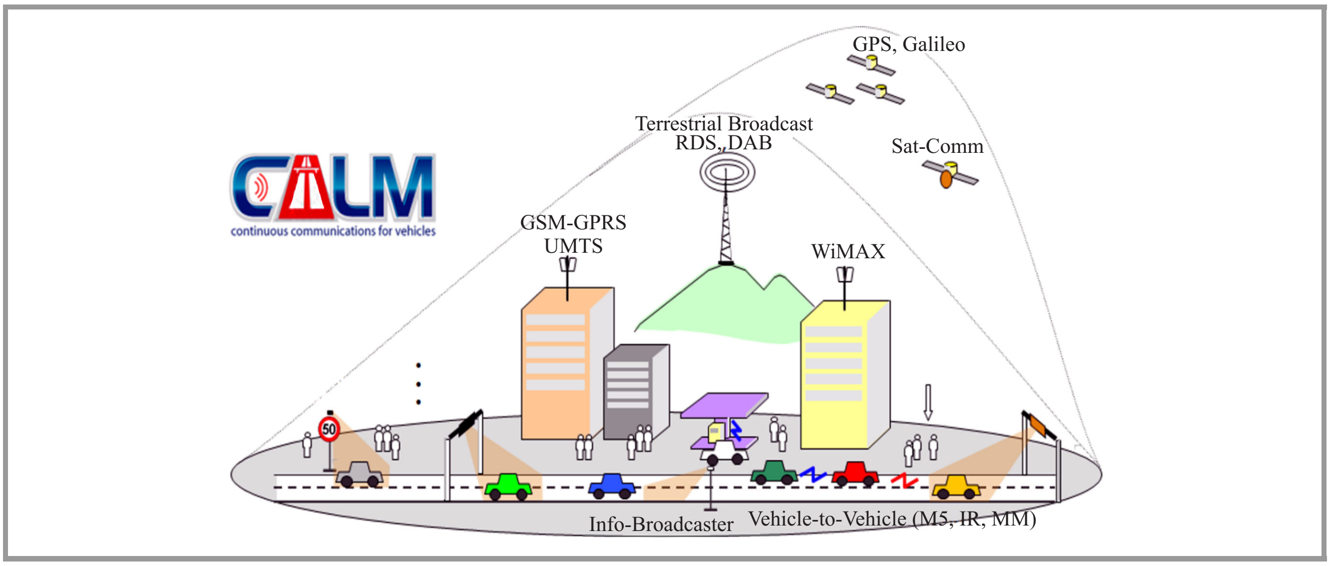

Fig. 2. Continuous Air Interface for Long and Medium distance usage scenarios.

possibilities made by particular energy companies, there are two emerging characteristics of such networks, which can be observed universally: move towards open standards and widespread employment of IPv6 communication. They both stand in contrast to earlier technological network deployments, which tended to utilize specialized, proprietary solutions, often compatible only with products of the same manufacturer. Development of mature open standards, able to provide necessary level of both functionality and reliability, combined with falling costs of industrial automation hardware capable of supporting IP-based communications make this evolution direction the most attractive one.

Another important example of specialized infrastructure solutions are vehicular networks. In their case both some requirements of popular access and some requirements specific for technological networks have to be addressed. A large user group utilizing diverse range of hardware solutions have to be supported, but, at the same time specifications for these devices being strictly followed could be counted on, due to legal requirements concerning devices allowed to integrate with vehicle systems. Moreover, apart from communication protocols and procedures, also services for this environment tend to be clearly defined, which leads to higher predictability of required QoS level. Many of these services, on the other hand, can have considerably higher QoS requirements then general Internet ones – especially in case of safety-related, automated solutions, i.e. collision avoidance mechanisms.

Continuous Air interface for Long and Medium distance (CALM) [6] can serve as an example of standardized solution for vehicular environment. The standard defines comprehensive set of elements necessary for creating a fully functional system, covering:

- a diverse set of access technologies, starting with wired access, and including wide range of wireless technologies such as IrDA, Personal Area Networks (PANs), short range RF broadcasts, Wireless Local and Metropolitan Area Networks (WLANs and WMANs) and cellular technologies (2G/3G);
- network layer mechanisms and protocols for handling communication within complex network structures – based on IPv6 protocol stack;
- network and service convergence solutions, allowing seamless integration with external network systems (including IPv4/IPv6 Internet) and both CALM-aware and proprietary services;
- application implementation and integration, for creating application level service providers and clients able to both seamlessly function in CALM network environment and take advantage of its additional functions, i.e. as user's location awareness;
- management and control mechanisms for all defined layers.

Two additional characteristics of this standard require a special attention in general context of ubiquitous network access. The first observation is based on the following list of communication scenarios which are supported in CALM environment: Vehicle to Infrastructure (V2I) Non-IPv6, Vehicle to Vehicle (V2V) Non-IPv6, V2V and V2I Local IPv6, V2I Mobile IPv6 (MIPv6) and Network Mobility (NEMO) (see Fig. 2) [7]. Non-IPv6 scenarios are included sorely for purposes of compatibility with existing proprietary solutions. The remaining scenarios clearly divide communication into direct interactions between 2 system elements (both V2I and V2V) – where basic IPv6 mechanisms are used for sake of simplicity and performance and universal, general purpose IPv6 communication mode. The observation particularly interesting from author perspective is that in case of general purpose communication, the use of network layer mobility management solutions, in this case

Mobile IP (MIPv6) [8] and Network Mobility (NEMO) [9] is mandatory. That clearly indicates the importance of this group of network mechanisms in complex heterogeneous access system environment.

The second observation is that despite high level of independence between services and access technologies used by client, the system allows services to utilize specific characteristics of a particular access technology to provide additional functionality. For example, low range transmission technologies can be used to broadcast warning messages over limited areas without need for inclusion of higher layer range control solutions.

Information about current user location proves to be very useful in providing services to mobile users. With currently available mobile devices being comparable to popular stationary computers in terms of their performance characteristics, one of their main limitations seems to be the user interface – required to be easily usable on small displays and with user input methods severely limited in their range and precision. With such constraints, mobile user's ability to efficiently absorb and filter large amounts of information by use of such an interface is strictly limited, so steps should be taken to further prepare information provided to him, taking into account his personal preferences and current needs. For this task, information about user's whereabouts can be of high value – for example: user entering public transport vehicle will probably be interested in ability to make necessary payments for a very specific line, tariff etc. instead of obtaining full and comprehensive information about a city's public transport system.

With precise geolocalization being both well researched and still difficult task, at Gdańsk University of Technology scientists have been researching the use of context localization – obtaining information about user proximity to various access network infrastructure elements. There is a high number of frequent tasks where precise geolocalization is both an error prone and not particularly efficient method, while context localization proves to be both easy and well suitable. For example, in already mentioned public transport example, is proves very difficult to clearly state if the user is on board of a given vehicle (Fig. 3) – due to both localization errors (with required precision being rather high) and unpredictable vehicle mobility.

At the same time, by a simple measurement of signal strength from on-board wireless access point, the above task can be easily fulfilled.

While the two broad user groups mentioned above cover a vast majority of ubiquitous network access users, there are also some specific environments and uses, where providing ubiquity of network access requires dedicated approach. As an example for such environment the author chosen a broadband maritime networking.

There is currently a number of systems and technologies used to provide digital communication between maritime vessels themselves and between them and shore infrastructure. However, due to their changing locations, unpredictable propagation characteristics, long communication ranges etc. available solutions tend to be costly and offer low transmission throughput (as can be seen in Table 1). Such limitations confine their employment to basic navigational, reporting and safety related applications.

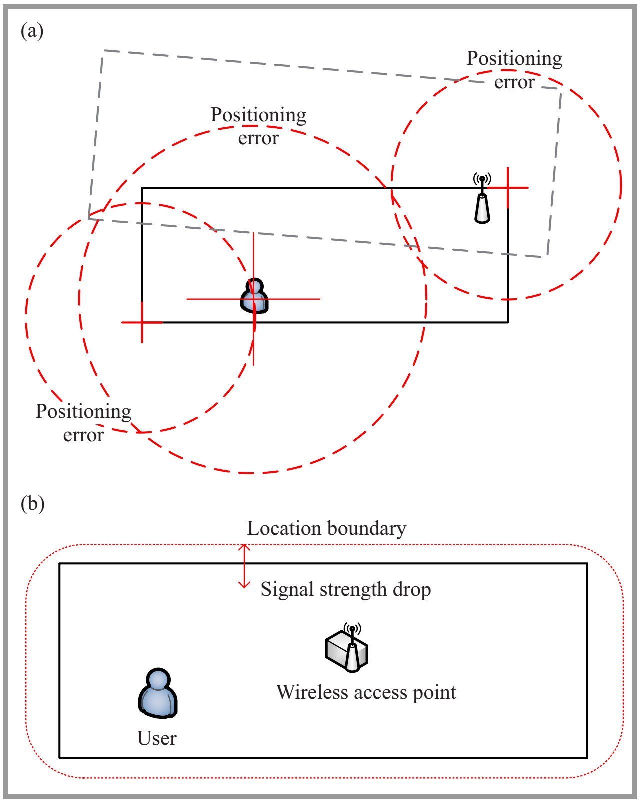

Fig. 3. Public transport vehicle scenario comparison: (a) geolocalization and (b) context localization.

Table 1
Comparison of maritime data transmission systems

System	Transmission type	Throughput
NAVTEX	HF, MF	300 b/s
DSC	VHF	1.2 kb/s
GPS	NMEA 0183	4.8 kb/s
AIS	VHF	2×9.6 kb/s
EPIRB	COSPAS-SARSAT	100 b/h
SSAS		100 b/day
SafetyNET	Extension of NAVTEX to Inmarsat coverage	100 message/day
Other satellite-based systems	Inmarsat, VSAT, ...	64 kb/s – 4 Mb/s

There are, whoever, multiple other uses for broadband data transmission in maritime environment (Fig. 4), especially with recent emergence of enhanced-Navigation (e-Navigation) initiatives, aiming to provide ship officers with comprehensive, integrated services suite for both safety and efficiency of maritime traffic [10], [11].

With obvious inadequacy of currently available solutions, such as expensive satellite communications and range limited shore cellular base stations, the issue of extending the

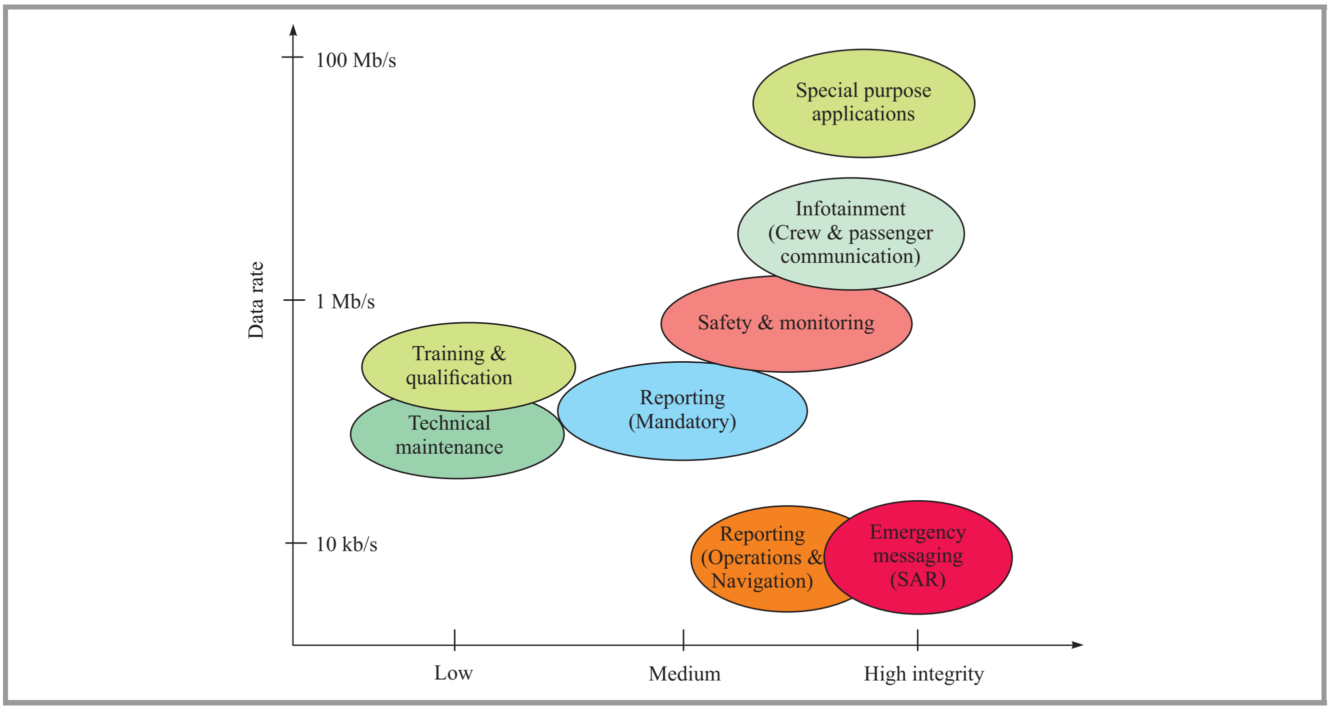

Fig. 4. Maritime services.

availability of broadband network access over sea has been a subject of research at Gdansk University of Technology. Proposed solution includes employment of a number of modern concepts, i.e., self-organizing Wireless Mesh Networks (WMNs), cognitive radio technologies and communication procedures differentiation based on ship location, to create an integrated, self-configuring, heterogeneous network access system. The above mechanisms seem to bring many advantages and facilitate the task of providing ubiquitous network access in diverse deployment scenarios encountered in maritime service.

3. Wireless Access Technologies and Architectures

There are currently many wireless transmission technologies, which can be divided into many groups and types. From author's perspective, the most interested in technologies which can be classified as Wireless Local and Metropolitan Area Networks (WLANs and WMANs). They, supplemented by immensely popular cellular technologies of 2nd to 4th generation (2G-4G, discussed in later section), form practically all popular, and vast majority of all modern broadband network access systems.

While WMAN technologies, such as WiMAX (IEEE 802.16 [12]) are currently deployed only in specific scenarios, being replaced as leading general-purpose, operator-level access technologies by Long Term Evolution (LTE) [13] standards maintained by 3GPP, Wi-Fi WLAN technologies based on IEEE 802.11 family of standards are next to omnipresent in technically developed areas. Despite the fact that first IEEE 802.11 standard has been proposed over 15 years ago, constant and rapid evolution, driven by actual user needs, has led to its constant and rapidly increasing presence, making Wi-Fi the WLAN technology of choice.

The evolution of Wi-Fi technologies can be divided into 3 distinct stages, corresponding to increasing levels of standard's technological maturity. The first Wi-Fi standard, IEEE 802.11-1997 [14], offered transmission speeds up to 2 Mb/s over radio and infrared media, utilizing contention-based medium access mechanisms. From the network administrator's point of view, it lacked almost all elements and functions necessary for utilizing it as an efficient and reliable element of a complex network system, and had to be regarded as not much more as a proof of concept, showing the possibility of creating a low-cost wireless transmission solution.

In this situation, the first stage of development of IEEE 802.11 standard addressed the most pressing requirements necessary for the discussed technology to be used in production grade systems: available throughput, elements of QoS management and security. As a result it became possible to reach transmission rates up to 54 Mb/s in both 2.4 GHz (IEEE 802.11g [15]) and 5 GHz (IEEE 802.11a [16]) ISM bands. Moreover, multiple optimizations and advanced mechanisms allowing both traffic prioritization and hard QoS guarantees were defined in IEEE 802.11e [17]. However, it was never implemented in practice. On the basis of IEEE 802.11e, Wireless Multimedia Extensions (WME) [18] specification has been developed, covering only traffic prioritization and assorted optimizations of transmission efficiency and power-saving functions.

To address gaping holes of initial Wired Equivalent Privacy (WEP) [14] security mechanisms, an IEEE 802.11i [19] extension has been defined, introducing cryptographically sound suite of security mechanisms. At this point, an IEEE 802.11-2007 [20] release of standard have been published, marking development state allowing the use of Wi-Fi technology in production grade systems, and beginning the second stage of standard evolution.

With the strictly necessary functionality present in IEEE 802.11-2007 standard, further development concentrated on still lacking, monitoring and management tasks. With extensions such as IEEE 802.11k (Radio Resource Measurement) [21] and 802.11v (Wireless Network Management) [22], it becomes possible to improve network efficiency by controlling not only infrastructure devices, but also wireless clients, which have been impossible previously. There are also multiple extensions dedicated to interworking and creation of complex network systems, i.e., IEEE 802.11u (Interworking with non-802 networks) [23], IEEE 802.11r (Fast Roaming) [24] or IEEE 802.11s (Mesh Networking) [25]. Growing ability of Wi-Fi networks to function in complex network environment, created the need for protection of its management traffic, which, up until this point, have been transmitted unprotected as IEEE 802.11i covers only user's traffic protection. For this purpose IEEE 802.11w (protected Management Frames) [26] extension have been introduced. In parallel with these management-related improvements, the work towards improving available throughput is has continued, resulting in IEEE 802.11n (higher throughput improvements using MIMO) [27] specification, allowing for transmission speeds up to 600 Mb/s (depending on number of spatial streams and channel width). There is also a first, service-related extension to Wi-Fi standard – IEEE 802.11p (Wireless Access for Vehicular Environments) [28], dedicated to use of Wi-Fi in vehicular networks.

A new update of main standard follows, marked IEEE 802.11-2012 [29], specifying Wi-Fi as a fully mature technology, with well recognized place in both popular home deployments, corporate networks and sizable access systems.

At present, Wi-Fi technology diversifies to cover multiple possible deployment scenarios. There are some extensions concerning its use for efficient handling of multimedia traffic (IEEE 802.11aa – Robust Audio/Video Streaming) [30], and growing management traffic prioritization (IEEE 802.11ae [31]), but the most prominent are transmission related improvements.

There are concurrently 4 separate extensions being developed, dedicated to radio transmission mechanisms for different usage scenarios:

- IEEE 802.11ac [32] – aiming to provide very high throughput (over 1 Gb/s) in traditional 5 GHz band, suitable for general-purpose popular deployments,
- IEEE 802.11ad [33] – designed for very high throughput (up to about 7 Gb/s), but very short ranged transmissions, suitable for indoor, line-of-sight interactions between mobile devices and infrastructure,
- IEEE 802.11ah [34] – operating at frequencies under 1 GHz, created to extend network coverage at the cost of transmission rate, which makes it well suited for monitoring/automation systems,
- IEEE 802.11af [35] – introduces cognitive radio mechanisms to Wi-Fi, allowing transmissions in unused TV frequency channels.

By adopting such diverse development directions, authors of IEEE 802.11 standards family clearly aim to make it the standard of choice for diverse needs created by varied deployment scenarios necessary for ubiquitous network access.

One of the very interesting elements being introduced to modern wireless access networks (including Wi-Fi) are cognitive radio mechanisms. They allow these networks to utilize radio frequency channels assigned to other technologies, as long as they will not negatively impact functionality of the primary owner of the channel. The most common example involves use of TV Whitespace (unused TV channels) for data transmission. There are currently two such solutions in process of standardization dedicated to the task: IEEE 802.22 [36] and already mentioned IEEE 802.11af [35].

The first, IEEE 802.22 has been designed in point-to-multipoint architecture, to provide Internet access service for stationary or nomadic users over large areas. With typical Base Station (BS) transmission range of 33 km and maximum of about 100 km (Fig. 5), its deployments are categorized as Regional Area Networks (RANs) [35]. The utilized frequency range depends on local regulatory rules, but generally fall in sub-one gigahertz range (for example 54–862 MHz), which ensures good propagation and coverage over long distances. With each TV channel of 6 MHz being used, the system is able to provide asymmetric data transmission rate of 23 Mb/s. As avoiding disruption of primary service take absolute priority, IEEE 802.22 standard includes a number of mechanisms to prevent such occurrence. Each client terminal (Customer Premises Equipment – CPE) is identified by the system and knows its current geographic location, which allows it to consult a dedicated database to obtain a list of RF channels which can possibly be used. Building on that basis, the system gives its BS complete control over CPE activity, which allows fast reconfiguration as needed. Moreover, sophisticated spectrum sensing mechanisms are included in both BS and CPEs, ensuring real-time reaction for presence of primary service signal in channel which has been considered free.

The described technology, apart from improving radio resources utilization efficiency, provides an important tool for providing ubiquitous network access, due to its long range and through coverage. With IEEE 802.22 technology, it becomes relatively easy to provide network access

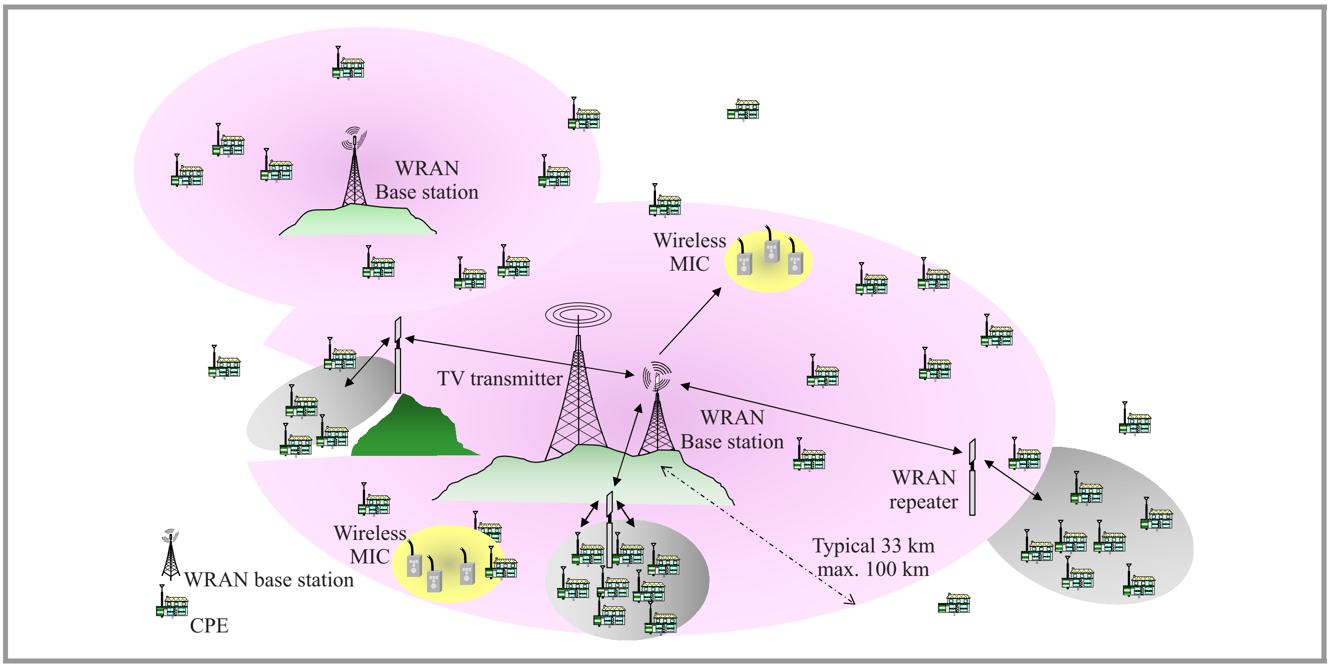

Fig. 5. IEEE 802.22 Regional Area Network.

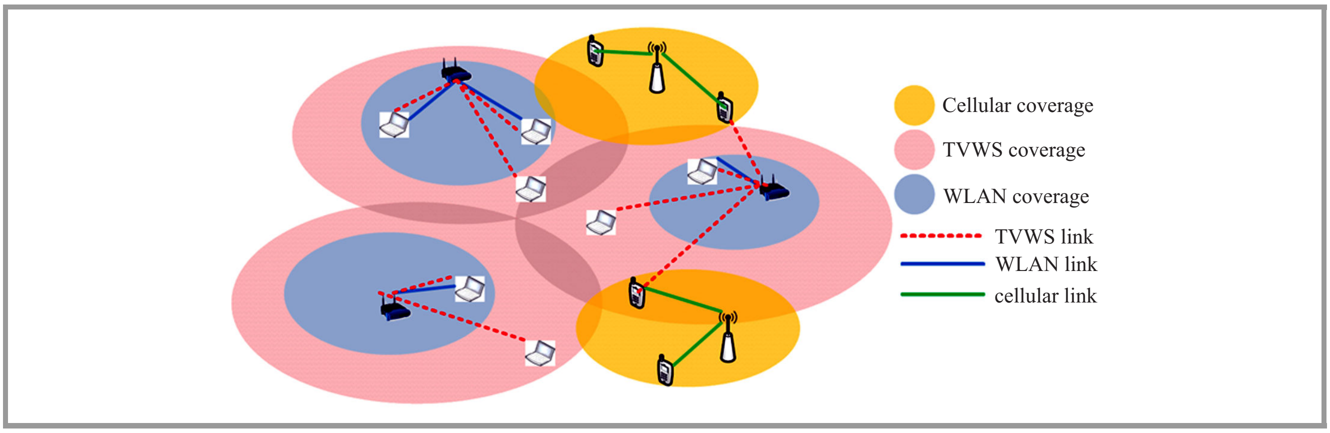

Fig. 6. Predicted usage scenario of IEEE 802.11af access technology.

over extended areas, which can then be supplemented with shorter-ranged, but capable of higher throughput, technologies such as WLANs, WMANs and cellular systems.

The second of the discussed cognitive radio technologies, IEEE 802.11af [36], utilizes mechanisms very similar to these present in IEEE 802.22 technology and also takes TV Whitespace advantage. It is, however, designed for much smaller ranges and with maximum allowed transmission power of 100 mW, can be used to extend range of Wi-Fi APs (Fig. 6) [37]. Despite possible problems of coexistence with IEEE 802.22, this technology promises to close the gap between cheap but very short ranged Wi-Fi coverage and much more costly WMAN technologies (i.e. WiMAX) and cellular systems.

Another of relatively new approaches to providing through coverage without the necessity of deploying extensive, fixed infrastructure consists of deploying a broadband wireless system of devices capable of forwarding received traffic in highly automated manner. Highly developed auto-configuration, dynamic routing, fault management, monitoring etc. mechanisms make such systems a very robust solutions. They are generally labeled Wireless Mesh Networks (WMNs), despite the fact that the description covers at least two popular, yet vastly different approaches to deployment of an access system. The first one, which can be called pre-designed WMN, consists of a number of devices in a network structure designed and deployed by network operator, which take advantage of mesh functions to provide network connectivity for dedicated access points and their connected clients. In this case mesh nodes can be homogenous and possess considerable resources (such as multiple wireless interfaces, crucial for efficient transit traffic forwarding), while the network structure itself can be optimized during design stage.

The second one, which can be described as ad-hoc WMN, allows each connecting client to act as fully functional mesh

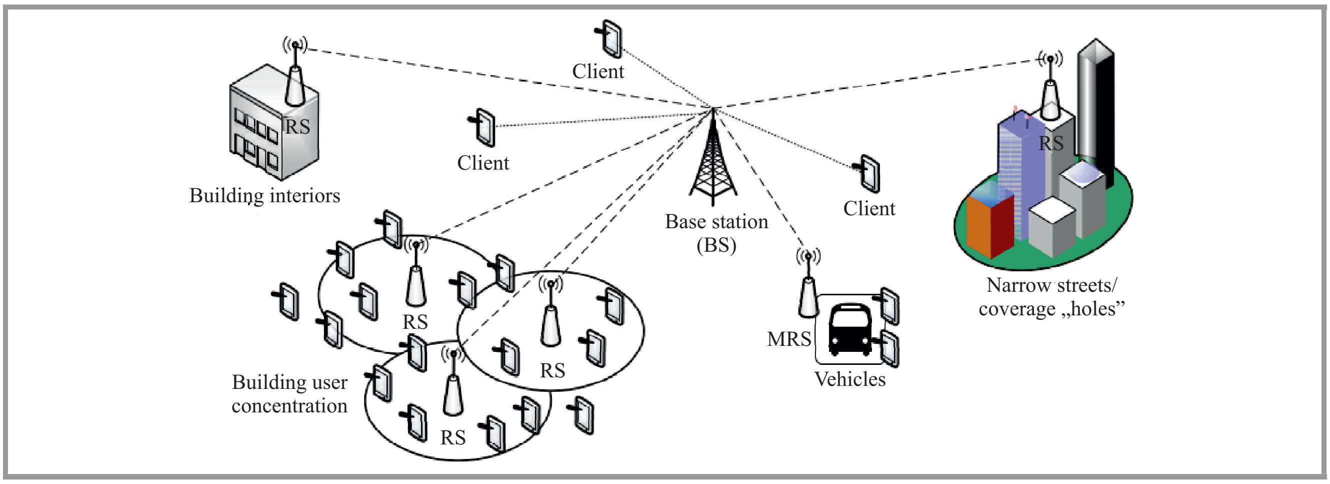

Fig. 7. Relay stations.

node, so each such client can extend its overall traffic forwarding capacity and coverage. Such ability allows a severe reduction of the necessary operator provided infrastructure compared to classic point-to-multipoint systems. It is also a significant step towards network access ubiquity, as such mesh systems tend to provide through coverage even in difficult propagation conditions.

Mesh networks can be used in a variety of roles, starting from small ad-hoc systems, through a highly robust and redundant access network infrastructure, and ending with emergency or military communication networks or self-organizing office/building/campus integrated infrastructure/access systems. In all of these scenarios, the main mesh networks advantages include autoconfiguration and self-forming capabilities.

One of the most promising mesh solutions currently being developed is an IEEE 802.11s standard [25], aiming to create a broadband, fully autoconfigurable, dynamically extending, and secure mesh solution, based on widely popular Wi-Fi technology. It is designed to serve in wide variety of environments, starting with small ad-hoc, isolated networks (for example: laptops or smartphones groups), through industrial/sensor network deployments, office LANs, and ending with large, self-extending, public access systems. The fact that this solution is based on cheap and popular Wi-Fi technology and can be deployed on existing hardware makes it one of very few mesh solutions able to successfully appear and remain on popular WLAN market. Additionally, a number of design decision have been made to make an IEEE 802.11s mesh as compatible and as easy as possible to integrate with existing network systems.

Due to mesh network mechanisms complexity and the fact that described automation level of management functions is rarely necessary in case of pre-designed access network infrastructure, mesh architecture is slow to gain popularity in such deployments. In vast majority they retain classic point-to-multipoint architecture, with base stations acting as points of network attachment to clients. However, due to relatively high cost and infrastructure requirements of fully functional base stations, a relay stations concept have been introduced, which be seen as a simplified form of multihop transmission. Relay stations are responsible for providing network access to clients, but only under direction of already deployed, fully functional BS, which allows them to be significantly simpler and cheaper. Moreover, while BS requires dedicated network connection to the infrastructure, relay station can be connected to its governing BS using the same mechanisms as clients, instead.

As a result, different variants of relay stationscan be deployed (Fig. 7) – starting from simple range extenders utilizing in-band communication with governing BS, through somewhat more costly ones which can to be chained forming multihop structure, and ending with versions able to internally perform most operations necessary for servicing clients and are useful for offloading governing BS in areas of high client density.

Examples of technologies which define relay stations include more advanced WiMAX variants (IEEE 802.16j [38]) and already mentioned IEEE 802.22 [35].

With such diverse access technologies at our disposal it is natural, that access system providers will differentiate deployed technologies to best suite their technical and economic needs. Even if a single access network of particular operator will have homogenous composition, the network environment of a mobile end-user interested in retaining ubiquity of access will be a heterogeneous one. In this situation, presence of efficient mechanisms for handling a seamless change of his point of network attachment, both within the structure of a single access network and across their boundaries, is of utmost importance. The tasks required for this process can be roughly divided into two processes:

- handover support – ability to seamlessly connect to new point of network attachment and configure all necessary mechanisms for network access;
- mobility management – ability to retain client's identity and current network sessions despite handover, to allow for continued high-layer service access.

4. Handover Support

Ability to change a point of network attachment in a manner which will minimally disrupt network connectivity of a mobile user is a task of paramount importance in modern access systems. The process is not a straightforward one even in within a homogenous system under consolidated management, and it only gets more difficult when we need to perform it across administrative network boundaries or between two different access technologies.

Due to the task complexity and many different scenarios which contribute to its necessity, taxonomy of handover types and solutions is an extensive one [39], which makes it impractical to include in this paper. Instead, a review of some general areas for process optimizations and a few chosen approaches to the problem in example environment of the most popular WLAN technologies are presented.

The handover process in general can be divided in to a number of distinctive phases, including:

- handover detection – decision that it is necessary to perform handover. In case of some simple WLAN technologies it amounts to detection that client alredy lost network access;
- network search – obtaining information about new access networks possible for client use and choosing the one to connect to;
- association – attempt to connect to a chosen network;
- authentication – providing authentication information for new network's access control mechanisms. Theoretically optional, in practice a required step;
- higher layers configuration – after obtaining link-layer connectivity, it is necessary to reconfigure higher layer (mostly network layer) mechanisms.

Example time values necessary to perform the above steps in case of popular Wi-Fi technology and IP-based network are provided in Table 2.

Table 2
Comparison of maritime data transmission systems

Layer	Item	Best case [ms]	Worst case [ms]
L2	802.11 scan (passive)	0 (cached)	1000
	802.11 scan (active)	20	300
	802.11 association	4	80
	802.1x auth (full)	750	1200
	802.1x fast resume	150	300
	Fast handoff	10	80
L3	DHCPv4	200	500
	IPv4 DAD	0 (DNA)	3000
	IPv6 RS/RA	5	10
	IPv6 DAD	0 (optimistic DAD)	1000
	MIPv6 MN->HA	0	200

Some stages of the handover process can introduce significant delays – in particular, network search and authentication. Moreover, IP configuration, which includes obtaining a new IP address and verification if it is not duplicated by Duplicate Address Detection function can be quite lengthy. If we take into account, that Wi-Fi utilizes hard-handover, which means that existing connection is released as first step of handover process, each delay results in longer disruption of client's connectivity. Many approaches to handover optimization have been proposed, for example IEEE 802.11r [24] extension of Wi-Fi standard includes fast resume/fast handoff mechanisms which allow for drastic reduction of authentication phase.

In this situation, during author research activity at Gdańsk University of Technology the issues related to network search phase was addressed. By performing network search while the client is still connected to its current access point, significantly reduce handover time and resulting disruption of services can be achieved. However, if reduction QoS of existing network connection is unwanted, this process of background scanning can be a lengthy one, poorly suited for fast moving users. In this situation be decided to use a dedicated, physical interface for this purpose, which solved the problem and allowed to use much more sophisticated criteria in choosing new access point than simply current signal strength. Example results of experiment combining the described handover optimization with Proxy Mobile IPv6 mobility management protocol implementation (see Section 5) are presented in Table 3. The experiment consisted of a single Wi-Fi handover during MPEG (2 Mb/s) video transmission. For estimation of QoE level a Degradation Category Rating (DCR) 5 points MOS scale has been employed [40].

Taking the research further in this direction, the author decided to make the disruption of network connectivity largely independent of handover time, by introducing soft-handover to Wi-Fi technology. In this case the described preemptive scanning is performed and when decision handover is made, the connection to old access point is not disconnected, until a new one is finalized. By use of this method at most one IP packet at handover is lost, which makes it next to transparent to user [41].

Implementation of the above mechanisms utilizes standard tools of popular operating systems introducing only additional management functions by means of scripting language, which makes it both highly universal, compatible, hardware independent and suitable for vertical (inter-technology) handover [40], [41].

Another approach to handover optimization have been demonstrated by concept of Virtual Cell [42], made possible by popularization of wireless installations based on Wireless Network Controller (WNC) architecture. In their case, instead of multiple fully functional access points (APs) able to forward network traffic between wireless and wired network by generally recognized rules, there is only one central entity responsible for traffic handling, and all

Table 3
Impact of IEEE 802.11/PMIPv6 handover on MPEG video transmission

Scenario	No. handover	PMIPv6 with standard handover	PMIPv6 with optimized scanning
MOS (DCR)	4.86 ±0.09	2.34 ±0.17	3.98 ±0.14
Mean delay delta [ms]	3.98 ±0.001	4.99 ±0.05	4.63 ±0.04
Mean jitter [ms]	3.97 ±0.001	4.97 ±0.05	4.61 ±0.04
Mean packet loss	78 ±37	3498 ±500	1065 ±250
Connectivity gap [s]	–	5.05 ±0.78	1.45 ±1.00

access points (called Lightweight Access Points – LWAPs) are responsible sorely for forwarding it towards WNC. Such an approach, while creating evident problems with scalability, provides level of control over network system which have not been possible before. Proprietary Virtual Cell technology takes even more radical approach – the WNC is able to control activity of APs to the extent which makes it possible for the network to be presented to standard Wi-Fi client as a single virtual AP which relocates between hardware APs, following the client. As a result, client never experiences link layer handover and disruptions of his connectivity related to virtual AP relocation do not exceed 5 ms. At the same time, there is no need to differentiate frequency channels between neighboring APs, as WNC is able to coordinate transmissions (including that of standard Wi-Fi clients) to avoid interference. That, in turn, makes it possible to place APs in much denser manner, thereby extending system capacity as client number is concerned.

Apart from the above techniques, designed to improve link layer handover processes efficiency, there are also similar solutions for network layer-related handover stages. For example, it is a popular approach to omit DAD procedures, relaying on proper functioning of address assignment solutions, and accepting marginally probable address conflicts, to obtain significant improvement in handover performance.

5. Mobility Management

However, even efficient handover itself does not guarantee, that mobile user will be able to continue his activities uninterrupted. It is highly probable, that, due to change of location in network structure, his network address will also be changed, resulting in disconnection of existing network sessions. To prevent such occurrence, it is necessary to employ mobility management mechanisms.

Despite the fact, that there are various sets of network layer mechanisms, the author is going to concentrate on IP-based networks, as by far the most popular ones. Moreover, many problems of efficient mobility support encountered in IP networks are also valid for different network layer solutions.

The single most important consideration is the fact, that an address in IP network serves dual purpose – it both uniquely identifies the client and describes its location in a network structure. Due to this characteristic, a change in client's point of network attachment significant enough to place him in different location within network-layer system structure, must also result in change of his IP address – which, in turn, results in change of his identity, as far as network mechanisms are concerned. To prevent such occurrences and allow the user to preserve his network sessions continuity, a number of mobility management mechanisms have been proposed. The most universal approach is to implement them in network layer, thus allowing them to provide mobility support for different higher layer protocols and applications in transparent manner (for example by allowing the client to retain his IP address). However, this approach requires a network protocol stack modification and additional mechanisms inclusion.

Such requirements resulted in slow deployment of network layer mobility management solutions, and significant number of application layer solutions have been deployed instead [43]. They perform efficiently for a single specific application or service. There are also some propositions of mechanisms located in other ISO-OSI layers, for example in transport layer, but they have not gained significant popularity.

To provide ubiquitous network access, the network layer mobility management is most interesting, due to mentioned transparency for higher layer mechanisms and independence of lower layer transmission technologies. In their case, available solutions can be divided into three groups, based on general architecture of a given solution:

- client-side solutions – require additional mechanisms to be included in client's network protocol stack, but will function in any access network;
- network-side solutions – all necessary mechanisms are located within an access network, while client equipment need not to be modified in any way;
- mixed solutions – require both client device and access network mechanisms modification.

From the above groups, mixed solutions are relatively poorly suited for this purposes, requiring both network devices of access systems and client devices modification, which complicates their popular deployment. This group includes well known Mobile IPv4 [44] protocol, developed over 18 years ago, which functions in a manner very similar to client-side Mobile IPv6 [8] solution described below, but, due to limitations of IPv4 protocol, requires additional element to be present in access network being currently used by mobile client.

Client-side solutions, such as Mobile IPv6 [8], allow modified client devices to retain their IP address as they move through unmodified access systems. This is a powerful ability as far as ubiquitous mobility support is concerned, providing users with global mobility support (macro-mobility – see Fig. 8). Moreover, the emergence of a relatively small group of general purpose operating systems for mobile devices, makes the requirement of client-side IP protocol stack modification an actual possibility even in mass deployments.

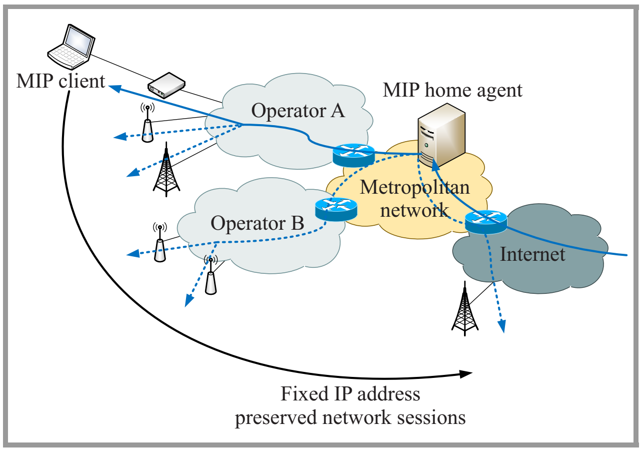

Fig. 8. Mobile IPv6 client-based mobility management solution.

Its basic principle of operation involves the client (called Mobile Node – MN) possessing two IP addresses: an unchanging home address and a Care-of-Address (CoA) obtained in visited access network. After obtaining CoA, MN contacts Home Agent (HA) entity responsible for managing its mobility and registers a mapping between its home address and current CoA. The traffic for MN's home address is routed to HA, which delivers it, by means of tunneling, to CoA registered by MN. Traffic in opposite direction can be delivered using standard IP mechanisms (resulting in possibly harmful triangle-routing) or by using reverse tunnel from MN to HA.

Network-side solutions, in contrast, allow unmodified clients to retain their IP addresses, as long as they move within access network where this solution has been deployed. Such characteristics makes their general deployment more problematic than in case of client-based solutions, but ability to support any client device in a transparent manner can be highly beneficial. The most popular example of this approach is Proxy Mobile IPv6 [45]. In its case, access routers (called Mobility Access Gateways – MAGs) are responsible for detecting that client has moved between them, and will inform Local Mobility Anchor (LMA). LMA then tunnels the traffic to appropriate MAG, to be delivered to client (Fig. 9). To provide mobility service transparency, new MAG also impersonates the previous one, by assuming the same link layer and network layer (IP) address.

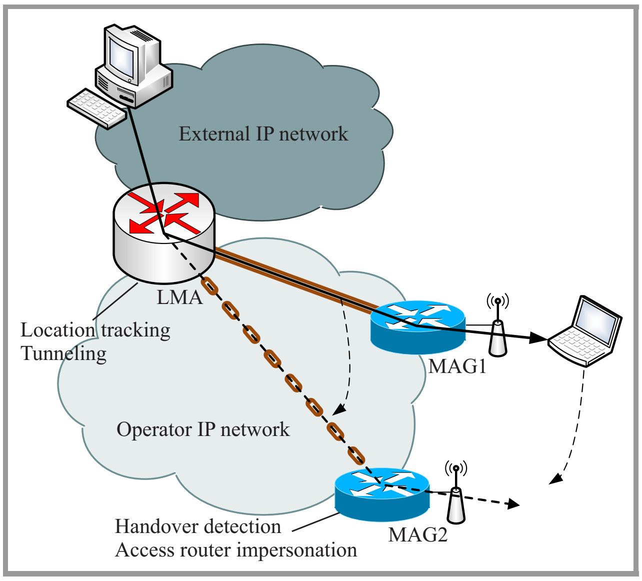

Fig. 9. Proxy Mobile IPv6 - network-based mobility management solution.

As was already mentioned in Section 4, presented research concerning mobility and handover support mechanisms resulted in fully functional implementation of PMIPv6 protocol, complete with related security feature (authentication, accounting, confidentiality and traffic transmission integrity) on Linux platform [40]. It is interesting to note, that such an implementation has been possible to create with exclusive use of scripting language (with associated ease of deployment and high level of compatibility) while still retaining high performance. This possibility derives from the fact, that the necessary functionality falls into management category, while performance intensive data handling functions are readily available in most of modern operating systems.

6. Cellular Network Evolution

When discussing ubiquity of network access it is impossible not to mention modern cellular telephony networks, due to both their popularity and almost universal coverage in technically developed areas. Until recently however, development of the above technologies (standardized mainly by 3GPP [13]) proceeded separately to popular WLAN/WMAN solutions and has been aimed at different goals. Discussed systems have been created as means of commercially providing users with relatively small and well defined services set, mainly related to direct human communication. In this situation it is natural, that standardization moved towards solutions allowing creation of a tightly integrated, homogenous system, complete with a comprehensive management mechanisms set. Specification also took full advantage of clearly defined service set, which

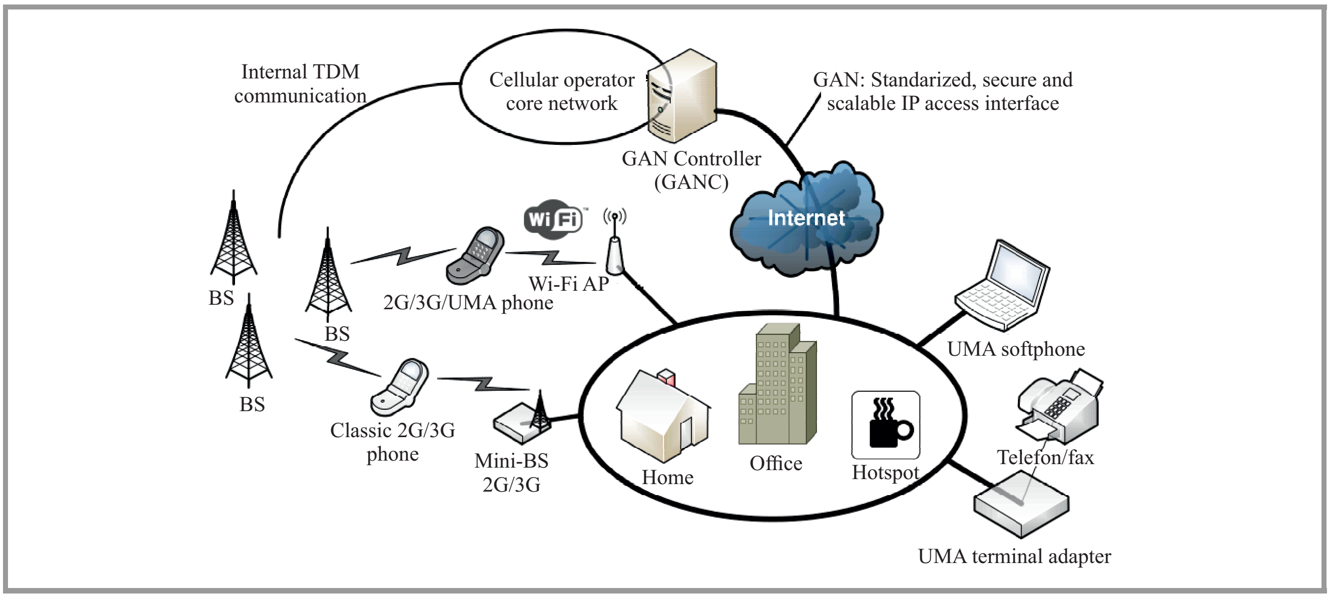

Fig. 10. Generic Access Network structure.

allowed significant simplifications in utilized mechanisms, compared to general purpose system.

It is worthy of note, that the first of the popular digital cellular communication systems, called Global System for Mobile Communications (GSM, 2nd Generation – 2G) provided only channel switching capabilities. Packet switching has been introduced later, in its first modernization: Enhanced Data Rates for GSM Evolution (EDGE – 2.5 Generation – 2.5G), by introducing additional elements to GSM infrastructure and retaining clear separation between channel and packet switching subsystems.

As user demand for additional services requiring packet data transmission started to grow, somewhat ad-hoc introduced packet switching capabilities of EDGE system have been substantially upgraded and tightly integrated with system infrastructure in Universal Mobile Telecommunications System (UMTS, 3rd Generation – 3G) network. Apart from higher possible data rates, there is also a visible trend towards simplification of Radio Access Network (RAN) which is a system part responsible for managing base stations and connecting them to the core network, by limiting number of separate elements and integrating their functions within base stations (called NodeBs).

At this point, with efficient packet switching capabilities of 3G system, first significant initiatives to integrate cellular and computer networks are observed. One of the most interesting is creation of Generic Access Network (GAN) specification, also known as Universal Mobile Access (UMA) technology, which defined mechanisms required to obtain access to services provided by cellular operator's core network with use of IP protocol (see Fig. 10).

By allowing such access, it become possible to obtain services provided by cellular communication network independently of access technology, as long as IP communication of sufficient quality could be maintained between a client and a GAN Controller (GANC). Moreover, client accessing services by means of GAN is always directly connected to his home network and able to access all of its services.

This revolutionary step has been a first standardized and widely recognized move towards new approach to services in communication networks. To date, it was the network that provided certain services to which users could subscribe. With the new approach, there are users, interested in obtaining access to a number of services, which they can access using different network access systems. Such an approach, combined with All-IP trend (providing all possible services by means of IP communication) and aim to provide ubiquity of network access by means of heterogeneous access systems, pointed the way towards the latest development in cellular communications – 4 Generation networks (4G), named Long Term Evolution Advanced (LTE Advanced) [13].

In case of 4G network the existing 3G infrastructure have been abandoned entirely, to be exchanged for RAN in form of a distributed set of sophisticated Enhanced NodeB (eNodeB) base stations, interconnected by IP protocol (over any available transmission technology) amongst themselves and with core network (Fig. 11). This independence of transmission technology (in contrast to a strictly defined allowable technology set of previous generations) allows for much easier deployment and maintenance of infrastructure, enables infrastructure sharing etc. Moreover, new core network architecture (Evolved Packet Core – EPC) also exclusively utilizes IP communication.

Such an architecture also makes it easy to provide support for GAN, but 4G network moves the idea two steps further, by providing mechanisms to integrate computer network technologies (such as WLAN, WMAN, cable modems, etc.) directly with EPC and by moving services outside of network core – 4G network does not provide any services itself, except packet data transmission. This ability and design decision should be considered an enormous step to-

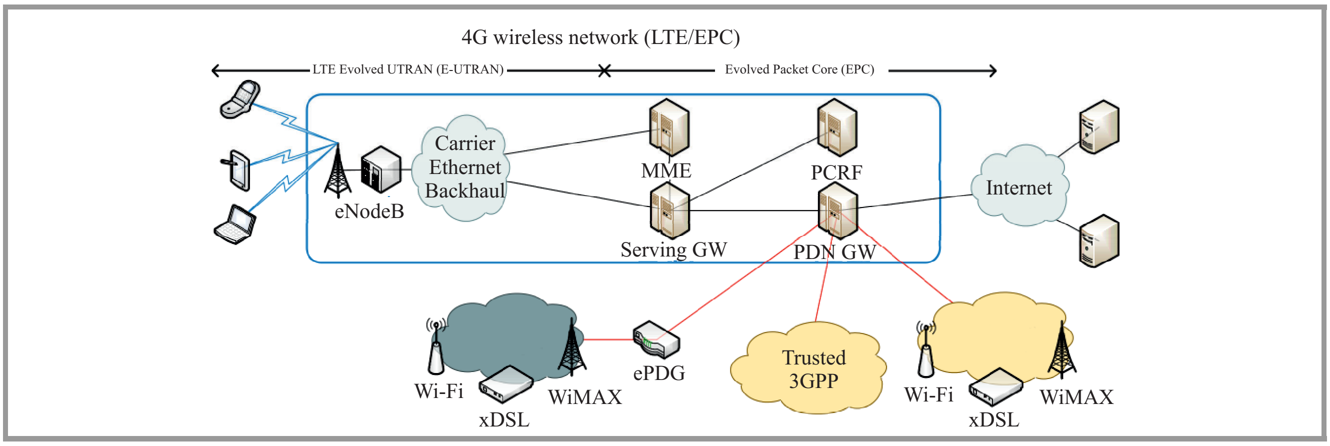

Fig. 11. 4G (LTE Advanced) network architecture.

wards integration of two separate, immensely popular network access technology types, and reconfirms All-IP-based, access agnostic approach to services.

Another element worthy of note is the fact, that 4G network employes well-known IP-based mobility management solutions to provide users with mobility support – for example: mobility of 4G users moving between eNodeBs belonging to different Serving Gateways (see Fig. 11) is supported with use of PMIPv6 [45] and mobility within connected computer network access technologies is supported with use of MIPv4 [44], PMIPv6 [45] or Dual Stack MIPv6 (DS-MIPv6) [46].

7. Conclusions

The provided general survey of various aspects related to a growing need for ubiquitous network access clearly shows that evolution of many separate elements, such as transmission techniques, access network technologies, system architectures and high layer software solutions begin to converge towards this difficult, common goal. Feasibility of successfully reaching this objective in real-world deployments, combined with emergence of approaches such as All-IP, XaaS and cloud services, seem to bring a real possibility of obtaining not only ubiquitous network access, but also to ubiquity in access to high layer services.

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Paper

Evaluation of the Delay-Aware NUM-Driven Framework in an Internetwork Environment

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Abstract—Nowadays, due to emergence of cloud services, even the basic uses of personal computers may require the access to the Internet. In this paper modifications to Delay-Aware Network Utility Maximization System (DANUMS) are presented, which enable it to be deployed in an internetwork environment. The proposed solution consists of DANUMS and WiOptiMo systems, which cooperate by exchanging measurements of transmitted traffic in order to improve the network utility. Additionally, WiOptiMo enhances mobility by providing facilities for soft handover. Experiments presented in this paper illustrate the benefits gained from the integrated system application.

Keywords—DANUMS, Network Utility Maximization, seamless mobility, WiOptiMo, Wireless Mesh Networks.

1. Introduction

Wireless Mesh Networks (WMNs) are regarded as the suitable technology for enabling permanent Internet connection, which provide many advantages for end users as well as for network operators. Comparing to Ethernet and standard Wi-Fi networks end users can stay connected to the Internet even where wired connection is not possible or direct path for wireless signal from Access Point (AP) is blocked. Another important aspect is the fact that the WMN is self-organizing with almost no configuration needed when adding a new node is required. From the operators' point of view wireless mesh nodes are easy to be deployed and relocated, what makes the operator network extremely adaptable and expandable. In parallel, the same area coverage may be reached with smaller cost since the need for wired network backbone is reduced. The last but not the least, WMN reliability also increases with the growth of the network.

However, WMNs require more sophisticated mechanisms that assure connection stability. The presented solution responds to these needs by allowing seamless gateway switching without breaking the connection what additionally improves the stability and increases overall QoS.

It has been admitted that the attainable bandwidth in wireless networks is significantly lower than those possible in wired networks. In order to optimize their utility, a Network Utility Maximisation (NUM) framework [1] can be used. In this paper the application of the delay-aware version of NUM framework has been evaluated, which in parallel to flows' rate, takes the delay flow characteristic into account. The presented research has been conduced within the work in the Carrier-Grade Delay-Aware Resource Management for Wireless Multi-Hop/Mesh Networks (CARMNET) project, which concentrates on implementation of mesh network in a way extending the backbone network of a telecommunications service provider [2]. As a result of work in the project, the integrated CARMNET system has been developed. The system contains the Delay-Aware Network Utility Maximization System (DANUMS) subsystem (responsible for resource management) and the WiOptiMo subsystem (responsible for mobility) as core elements [3].

The most popular scenario of wireless network use involves one or more gateways providing access to the Internet. As realization of the NUM model requires precise control over network traffic, it becomes difficult in the internetwork scenario. In particular, the DANUMS allows optimal scheduling in a wireless mesh network by taking the "utility" of flows into account [4]. The utility of a particular flow can be estimated only if relevant measurements are available. These measurements are gathered by DANUMS and disseminated by extensions of the Optimized Link State Routing (OLSR) protocol, including Urgency Reporting Messages (URMs), Queue Reporting Messages (QRMs) and Delay Reporting Messages (DRMs) described in [4]. This monitoring is only conducted within the mesh network, where each node is controlled by DANUMS software. As the flow leaves this controlled environment to reach its destination in another network, other monitoring methods have to be applied in order to ensure proper operation of DANUMS.

The paper contribution contains the description of the integration of the NUM system with mobility extensions. The integrated solution enables preservation of accurate utility estimation in the internetwork environment. The presented experiments also highlight benefits of this integration in the multi-gateway environments such as WMNs.

2. Related Work

IEEE 802.11s Mesh Networking standard [5] was developed as an extension of the IEEE 802.11 standard which is regarded as the foundation of Wi-Fi. Internetworking is realized by Mesh Portals which serve as gateways between local mesh network and the Internet. The Mesh Portal is supported by underlying routing mechanism with selection of the most optimal route for mesh nodes including the mobile ones. As a result of Mesh Portal operation, the connection is served during roaming between networks. However, when Mesh Portal becomes unreachable the connection continuity, and thus session continuity cannot be preserved.

The dynamic topology (caused by node movement or environment change) of WMN also deteriorates the connection quality. One of the reasons of this quality decrease is the fact that once chosen gateway cannot be changed to avoid breaking the already established connections. For the sake of the connection preservation, OLSRd [6] employs additional parameter called *NatThreshold*, which favors the currently used gateway, by means of artificial lowering of measured path cost. This is the simplest solution, which, however, has its disadvantages such as routing loops occurring when the *NatThreshold* parameter value differs between hosts. The SmartGateway and MultiSmartGateway [7] extension for OLSRd allows selection of gateway in a per-flow basis by use of the IP-over-IP tunnel. Such a solution allows better allocation and partially solves the issue of changing topology. However, the long-lived connections cannot be switched to the newly selected best gateway, as this decision can be undertaken only just before connection establishment.

Authors of [8] have proposed another roaming solution for a hybrid WMN. Based on the information regarding neighbor Wireless Mesh Routers (WMRs) and theirs clients, the packet loss during handover could be avoided by means of shifting tunneled connections between inter-connected APs. Such a mobility solution introduces only a little delay during handover, but requires that every WMR is interconnected in a single network.

Another approach to maintain mobility which does not require any support from lower network layers is Mobile IP. In that case the mobility is manage by allowing the mobile node to use two IP addresses: a fixed home address and a care-of address that changes at each new point of attachment. This is achieved by implementing a highly sophisticated solution, which makes mobility transparent to the application layer, but is prone to DoS attacks. The Mobile IP is not suitable for the intra-domain movement, which is much more frequent than the inter-domain movement what makes it even harder to be deployed in WMN networks.

In the Spontaneous Virtual Networks architecture [9] the mobility management is moved directly to application layer. It shares features with WMNs, such as self-organization, self-configuration, and distributed architecture without need for intrastructure. However, this approach requires modification to each and every application that is supposed to gain benefits from the solution.

Finally, the important aspect of NUM is ensuring that QoS requirements are met. While DANUMS uses the DRM extension [10] for flow monitoring in the Intranet, it is unable to collect measurements this way, when flows are forwarded to the external networks. Throughput of TCP flows and their delay can be measured by passive monitoring of ACK packets [11]. Such a solution benefits from no additional overhead, but cannot be applied for transport protocols other than TCP. For UDP flows, monitoring without overhead is not possible, but often can be realized through inspection of commonly accompanying protocols such as RTP Control Protocol (RTCP). In the presented solution, the authors have reused already available measurements provided by the WiOptiMo subsystem by implementation of the external measurement interface for the Delay-Aware Network Utility Maximization (DANUM) system [12]. The WiOptiMo measurements original purpose is to support the system QoS features, such as selection of least occupied SNAPT (the WiOptiMo proxy server) [12].

3. Applications in Real-World Scenarios

The need for CARMNET-like solutions can be observed from several perspectives. The first reason is the bigger and bigger need for permanent Internet connection caused by the growing popularity of Web applications and cloud-based services. On the other hand, many services, e.g., media streaming or Voice over IP (VoIP) calls, require uninterrupted connection during the whole session. Moreover, even unstable or slow connection may severely reduce the service utility. The integrated resource allocation optimization and mobility framework, such the CARMNET system, may be regarded as a tool addressing the above-mentioned issues.

3.1. Network Load Optimization

Let us assume two WMN users – *Alice* and *Bob* performing independent VoIP calls using Internet connection. As a result of DANUMS application, the network load may be balanced in order to achieve overall best possible quality for both of users. In order to enable optimal DANUMS operation, measurements needed for flow performance evaluation are provided by nodes constituting the mesh network as well as mobility management entity (SNAPT) located outside internal network. As a result of application of two sources of measurements, DANUMS is able to properly estimate the flow throughput and delay. It is especially important for the case of delay, since delay factor observed outside the WMN may be taken into account.

Steps

1. User Alice and Bob initiate independent VoIP calls to external network.

2. After a short period of time, the quality of Alice's connection decreases caused by the change of network conditions outside the governed WMN.

3. DANUMS is provided with information on the delay generated outside the WMN by means of WiOptiMo measurements.

4. DANUMS allocates network resources in a way increasing the perceived QoS of both connection trying to recompense the bad condition of Alice's connection.

5. Finally, in the case when Alice's connection QoS could not be compensated more network resources are provided to Bob's flow.

As a result of external delay measurements availability, DANUMS compensates the externally caused QoS decrease by reconfiguring network resource allocation in order to optimize summarized quality of both connections. In particular, when Alice does not perceive the service of satisfactory utility resources because of external delays, more resource are assigned to Bob's flow. In contrast, when the information on external delay is unknown, the Alice would perceive a decrease of call quality even if WMN would still have enough (delay/bandwidth) margin that could be used to preserve quality of both connections.

3.2. Seamless Gateway Switching

The benefit of seamless handover may be observed in the scenario, in which a user of wireless network watches a video on demand or uses another streaming service. When the user moves out of the reach of the initial gateway the connection has to be routed through a different gateway. In typical WMN without roaming technology employed, the change of a gateway implies the change of external IP address and the necessity of reestablishing the connection. Figure 1 presents comparison of the scenario of CARMNET handover solution application and the scenario of typical behaviour of WMN without seamless gateway switching. In the picture, each transport layer connection is represented by a double-headed arrow. The network layer path used for a given connection is illustrated by dashed lines between intermediate nodes. The black line denotes currently used path and connection, while the gray lines represent the previous network state. X mark is used to denote the path disruption (thus the connection disruption) which forces the use of a new network path and in consequence a new gateway. As each gateway in this scenario performs Network Address Translation (NAT), the initial connection is also disrupted, so it has to be restored as well. The bold arrow represents switching of a gateway that realizes the connection to the Internet.
Connection a is managed by user's application. In typical scenario, i.e., in scenario without CARMNET mobility support, this connection is abruptly broken when gateway is changed, and must be reinitialized by the application. In the case in which the proposed mobility solution is used, the connection a is transmitted through substitute connection t to the SNAPT server, and the communication can be continued as normal. Even if connection t is disrupted

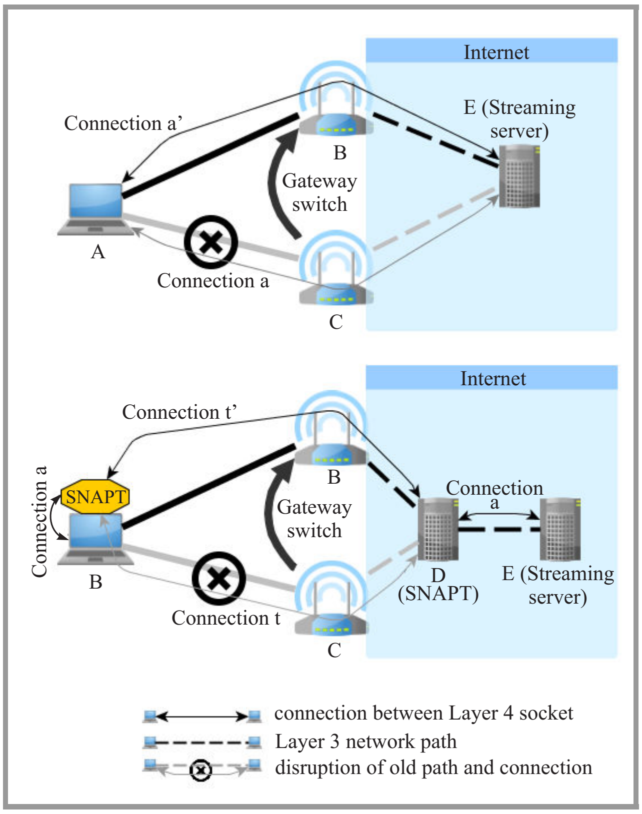

Fig. 1. Gateway switching approach comparison.

and renewed as new connection t', it is not visible for the connection end points on hosts A and E.

Steps

1. User device – denoted as node A – uses gateway C in order to access the Internet.

2. Node A connects to most appropriate SNAPT server.

3. Due to the node movement, the conditions are changing and a new path is chosen in order to preserve the desired QoS parameters.

4. A new path through gateway B is utilized without the need to reestablish the connection, thus without disturbing user experience.

Without the mobility support (such as provided by the proposed solution) the user watching a video and changing an Internet access point at the same time will experience the interrupted or completely broken video playback (depending to the application). As a result of presented proposal application the user will be not even aware of underlying gateway switching – each TCP/IP application which does not explicitly provide the user device IP address is compatible with our approach and may benefit from it.

3.3. Application in M2M Solutions

CARMNET solutions can be applied not only in the case in which users are directly involved, but also for purposes of Machine to Machine (M2M) communication. Devices such as vehicle trackers and path loggers and even those used for medical telemetry need to have the access to the network as often as possible. Mesh networks are particularly suitable for wireless sensors and other home automation equipment due to their ability to self-configure and repair network routes as well as the lack of the need to provide Internet access to each of the node independently. While distributed nature of the CARMNET mesh network allows easy deployment and extension of the network, the integrated WiOptiMo-based mobility solution allows to use multiple alternative Internet connection methods in a flexible way. The WiOptiMo preserves connectivity while switching between different access networks and Internet gateways. This enhancement allows the CARMNET system to provide the mobility support without additional modifications on the application layer.

4. CARMNET Framework

The idea of the CARMNET system has been proposed in [3]. The main goal of the system is to enable the WMN users to share their resources, in particular to share the Internet access. Additionally, the CARMNET vision assumes the work on integration of DANUMS with the Internet provider infrastructure – CARMNET solutions are aimed to be integrated with public wireless networks [13]. The Internet sharing functions (described in detail in [14]) are optimized as a result of the application of the utility-aware resource allocation subsystem, which allows to compare the utility of flows with different requirements with regard to end-to-end delay and throughput [4].

The implementation of the CARMNET framework requires integrated studies in several research areas including wireless network resource management, multi-criteria routing, and seamless handover. In this paper the authors have focused on integration issues concerning two CARMNET system components: DANUMS which is responsible for resource allocation, and WiOptiMo which is responsible for seamless handover functions.

4.1. DANUMS

The DANUM system [4] is aimed to maximise the overall utility of a mesh network defined as:

$$\sum_{r\in S} U_r(x_r, d_r), \tag{1}$$

where S denotes a set of flows within the network: x_r – rate of flow r, d_r – delay of flow r, U_r – the utility function of flow r. In order to do that, DANUMS prioritizes flows which gain the most utility from being served. A "virtual queue" level, defined as a product of a packet backlog and the value of the first derivative of a given flow utility function calculated for current flow performance parameters [4]. The parameters which influence the utility value include flow's packets delivery delay and throughput measured at destination node. Both the mentioned parameters are attained by active measurement through the use of DRM [10]. In parallel, virtual queue levels are used to perform Max-Weight Scheduling (MWS) [15] on each hop in distributed manner [4]. Additionally, the process is optimised by taking into account the 2-hop collision domain and use of the URM, which serves as more frequently broadcast abbreviation of queue levels information used to grant the right to transmit for a given node.

4.2. WiOptiMo

WiOptiMo provides decoupling between the IP address provided by the Internet Service Provider (ISP) and the IP address used to connect to a service on the Internet [12]. This feature allows a mobile node to change gateway transparently, without suffering service disruption, by means of a broker (SNAPT). However, there is still a limitation due to the architecture of Internet routing. It is not possible to change the SNAPT handling the application until its connections end.

To address the scalability problem, multiple SNAPTs can be deployed, hence allowing clients to use the least loaded available machine. The CNAPT monitors the delay to available SNAPTs, by means of passive and active monitoring of the control connection, and the bandwidth used by applications, in order to make a wise choice of the SNAPT to be used by subsequent applications. In particular, the calculated delay is used to select a SNAPT depending on the delay tolerance of the user application, while the estimated remaining throughput is used to avoid overloading a single SNAPT.

5. DANUMS and WiOptiMo Cooperation

Since DANUMS uses active probes (DRM), it cannot measure the each parameters of flows with endpoints beyond the CARMNET network. In such a case, DANUMS Loadable Kernel Module (LKM) can take into account only local queuing delay up to the Internet sharing node. Moreover, the accuracy of the rate measurement for such flows is also reduced. To remedy these problems, the DANUMS LKM allows injection of measurements from external sources through the procfs interface [12]. This interface is represented in Linux system as a regular file allowing both read and write operations [16]. The DANUM system operates on per-flow basis, i.e., it measures the network performance parameters for each flow separately. Consequently, each line of the procfs file corresponds to a single flow managed by DANUMS (see Fig. 2).

```
no src_address dst_address protocol delay throughput
0: 96FE299C:E624 C1AB8952:0102 06 * 120
1: 96FE299C:* C1AB8952:* * 156 *
2: C1AB8952:* 96FE299C:* * 156 *
```

Fig. 2. DANUMS LKM procfs interface.

However, DANUMS allows supplying information regarding routes as well. WiOptiMo monitors the route between CNAPT and a SNAPT with a separate control traffic flow. Information gathered in this process can be passed to the DANUMS LKM by writing it to the procfs file as flow measurement lines with reduced flow identifier to the source and destination IP addresses only. As a result, WiOptiMo (via the procfs interface) can provide measurements necessary for DANUMS to optimize its operation. In particular, the delay between a gateway and a SNAPT measured by CNAPT can be added to the delay of flows destined to that SNAPT and leaving the CARMNET network through the used gateway, in order to enhance the accuracy of measurements used in utility calculation. The CNAPT on the source node can also overwrite throughput measurements for flows which it is responsible for, in order to take into account the losses experienced outside the CARMNET network.

In standard use cases of wireless mesh network application (i.e., without the specific handover support), only Internet gateways have public IP addresses and use NAT to share this connection settings. If the Internet gateway is changed, the already established connections will break due to the public IP change. The most common practice is to preserve once chosen gateway regardless of changing conditions for routes. This is not applicable to connections established with help of CNAPT, as WiOptiMo solution allows public IP to be changed seamlessly. As a consequence, the best available route to the selected SNAPT can be used always, regardless of the currently selected gateway. This feature can be implemented without modifying the OLSR protocol and software implementation on nodes by simply introducing "virtual" host into Topology Control OLSR messages representing a connection between Internet sharing node and SNAPT as a link state information.

6. Experimentation Scenarios and Setup

The presented experimentation scenarios represent typical uses of the CARMNET network. Both experiments use a simple WMN topology presented in Fig. 3. Internet-sharing nodes B and C allow other nodes applying DANUM-based scheduling policy to access the external network.

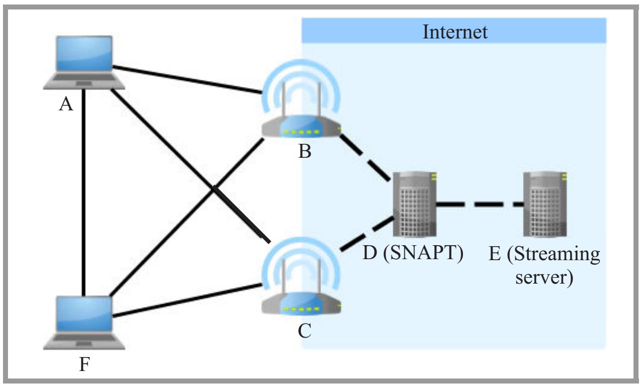

Fig. 3. Basic network topology used in the experiments.

Gateway nodes B and C perform NAT to allow nodes from WMN to connect to external hosts without the need for external IP address. The CNAPT clients, which goal is to enable mobility for chosen flows, are present on every wireless node (A, B, C and F). SNAPT – the server-side implementation of WiOptiMo – is hosted on server D. Server E hosts services, which can be accessed by client nodes A and F. When mobility support is used, client nodes connect to server E through SNAPT hosted on D.

6.1. wnPUT Testbed

The presented experiments are realized with help of physical testbed deployed in Poznan University of Technology campus – the wnPUT testbed [14]. Every node is equipped with wireless and wired network interface cards for serving experimental and out-of-band traffic, respectively. Testbed software consists of management framework and Debian-based Linux distribution allowing for the remote disk-less boot and control of testbed modes. Main task of the software framework is to ease the process of experimentation by allowing conduction of repeatable experiments on the commonly available hardware platforms. With wnPUT testbed each experiment can be described in the already established DES-Cript [17] format and executed without further intervention. For additional features such as remote power-cycling nodes in-house extensions of the experiment description syntax is used, preserving compatibility with DES-Testbed [17] at the same time. For testing in wnPUT testbed the DANUMS has been implemented as LKM (responsible for scheduling and resource allocation) and a plugin for OLSRd – the most widely adopted implementation of OLSR protocol (responsible for routing and measurement support).

For each experiment a physical testbed nodes was used which were interconnected by ad-hoc wireless network. The Internet connection was emulated with help of NetEm, simulating both delay and loss [18] commonly associated with Internet connection. In order to ensure packet backlog accumulation on gateway nodes, the Linux traffic control network subsystem was used to limit the available bandwidth of interfaces connected to the external network. This modification was required to illustrate common case, where Internet connection bandwidth is lower, than the one available in the local, even wireless, network.

6.2. The External Delay Influence Analysis

In the first experimentation scenario nodes A and F connect to the service provider E. Both connections route through SNAPT server hosted on D. SNAPT server is chosen in a way ensuring negligible delay to host E. The flow $f_{\mathrm{A-E}}$ is routed through the gateway B, whereas the flow $f_{\mathrm{F-E}}$ is routed through the gateway C. The gateways use measurements provided by WiOptiMo, which reports approximately 100 ms of delay between B and D, and approximately 20 ms of delay between C and D. Both delays were artificially enforced in order to simulate the fact that the gateways use different ISPs. UDP flows $f_{\text{A-E}}$ and $f_{\text{F-E}}$ experience different delays along the path due to operation of the DANUM scheduling and uncontrolled external network environment. For the topology stability, the OLSR standard-compliant hop-count metric was used. The experiment has been performed in two variants differing in the order of starting the flows. In the DANUM subsystem, the following UDP-based utility function [4] has been assigned to both flows:

$$U_{UDP}(x,d) = \frac{w}{1+e^{b(d-t_d)}}, \tag{2}$$

where $w = 10^6$ controls maximal utility, $b = 0.0375$ controls the slope of sigmoid function, and $t_d = 300$ specifies a delay threshold in which the derivative is minimal, hence the price is maximal. The delay is measured in milliseconds.

6.3. Seamless NAT Gateway Switching

In the second experimentation scenario the WiOptiMo solution is used to reduce influence of gateway switching in WMN. At $t = 0$ a single TCP connection is initiated between node A and service provider E using CNAPT-SNAPT software. This connection uses all available bandwidth and adapts accordingly to TCP agent flow and congestion control. At $t = 0$, C is the chosen Internet gateway of node A due to lower delay. In the 20th second of experiment ($t = 20$), the quality of path decreases by introduction of 5% loss on link A–C simulating node movement or external interference.

For execution of flows both with and without support of WiOptiMo solution the virtual node in OLSR topology could be used. This solution would allow independent gateway choice for both flows served in the standard way as well as for the case of WiOptiMo mobility support. For the sake of simplicity of implementation we have opted to transmit the information about the path to SNAPT by Host and Network Association (HNA) messages – same way that Internet access is broadcast by Internet sharing nodes. To change default gateway switching behavior the *NatThreshold* parameter of OLSRd is disabled on each node, as the gateway-preserving mechanism is not needed when the WiOptiMo solution is used. To detect and take the link quality changes into consideration, the Expected Transmission Count (ETX) route metric is used [19]. Using this metric OLSRd is able to choose paths with overall lower loss rate allowing higher throughput.

7. Experimentation Results

7.1. The External Delay Influence Analysis

Results of the experiment described in Subsection 6.2 show that the flow with lower delay suppresses the flow with higher delay regardless of the order of starting the flows (see Figs. 4 and 5). At the beginning of the experiment, there is only one flow in the network and, because

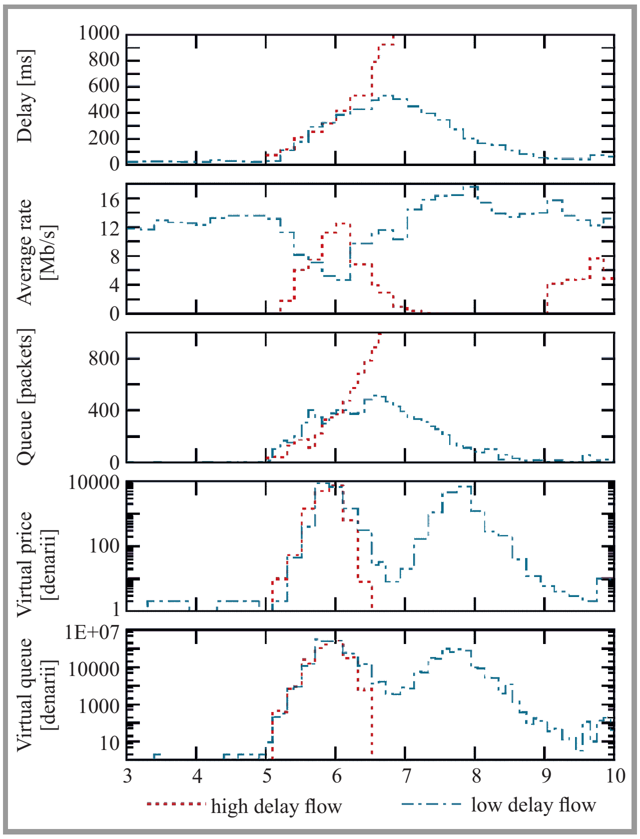

Fig. 4. Influence of external measurements on flow behavior. A flow with lower external delay has been started first.

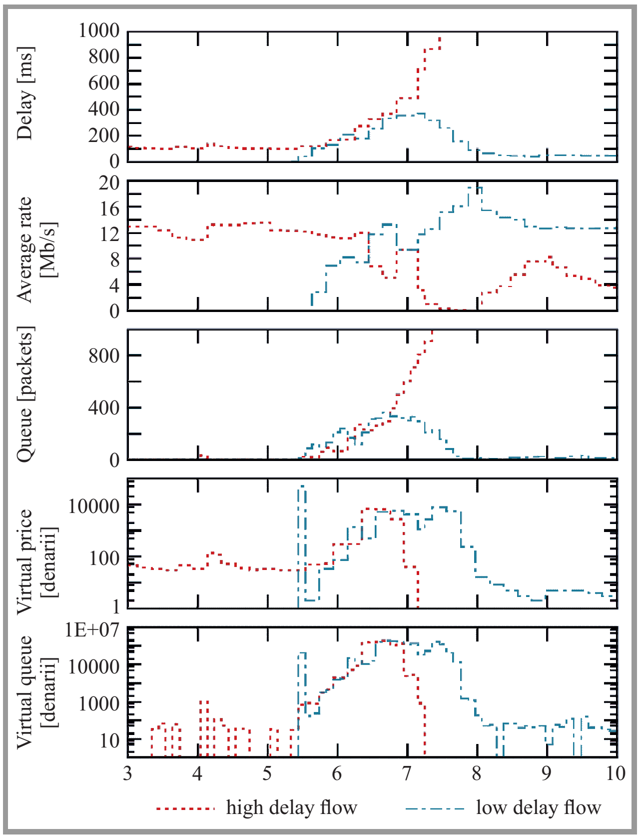

Fig. 5. Influence of external measurements on flow behavior. A flow with lower external delay has been started second.

of lack of competition, it experiences no delay besides the amount caused by external network conditions. When the second flow starts, both flows begin to compete for the bandwidth. The competition in this case (two UDP flows) consists of two phases. In the first phase the delay of both flows does not exceed the threshold of 300 ms. In this phase, the virtual price increases with the grow of delay. The second phase begins when the delay of one of the flows exceeds the threshold. Starting at this moment the price of this flow drops with the rise of its delay. This behavior is a consequence of the application of utility function used for UDP flows (see Eq. 2) for the DANUMS utility calculation.

In the first phase, packets accumulate in both flows' queues, because the sumarized flows rate exceeds the available bandwidth. As a consequence, the flows' delays increase, what causes the rise of the flows' price. Eventually, this situation leads to rising virtual queue levels. Basically, DANUMS tries to equalise virtual queues, which are calculated as a product of packet queue length and virtual price. Packets belonging to a flow with the highest virtual queue are sent first. Sending packets from flow's queue lowers the flow's virtual queue, but also causes the increase of the virtual queue level at the other flow. As a result, packets of both flows are sent alternately at insufficiently high rate. The situation changes when the delay of one of the flows eventually exceeds its threshold and the second phase begins.

At the beginning of the second phase, the price of one of the flows is maximal. This means that in order to achieve the same level of virtual queue, that flow has to accumulate more packets. However, the higher is the length of packet queue, the higher is the queuing delay. Further increase of delay causes the price to drop and, despite accumulating more packets, the flow becomes unable to compete for resources. Eventually, its price and, consequently, its virtual queue level drops to zero and its transmission terminates. In parallel, the second flow "wins" the competition and acquires bandwidth previously used by the "losing" flow. Superfluous bandwidth is used to transmit overdue packets of the winning flow, because it still has non-zero price and virtual queue. When the packet queue of the winning flow drops to zero, the flow rate stabilises and the remaining bandwidth is used to transmit packets of remaining flows.

As explained above, a flow "wins", if it has lower delay than other flows for the same value of virtual queue at the end of the first phase. Since one of the flows has lower delay, it tends to have higher packet queue level in order to compensate the price difference. Having a higher packet queue means that the price can be lower while still maintaining sufficient virtual queue level. In the first phase, the lower price translates to lower delay, therefore flows with lower delay are more likely to suppress other flows. It is worth to note, however, that external network conditions have the strongest influence on flows' delays. Without information about delay outside the wireless network controlled by the CARMNET system, flows would perceive very similar delays and determining the winning flow would be left to chance. This behavior is not desirable and shows importance of external measurements.

7.2. Seamless NAT Gateway Switching Experiment Results

Results of the gateway switching experiment (performed according to the scenario described in Subsection 6.3), are shown in Fig. 7. The topology evolution caused by the change of environment conditions is illustrated in Figs. 6a and 6b.

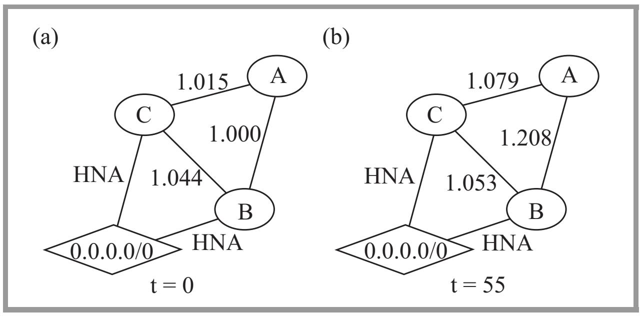

Fig. 6. Network topology with ETX metric values (as perceived by the OLSR daemon) on node *A* during the gateway switching experiment: (a) before and (b) after the change of environment conditions.

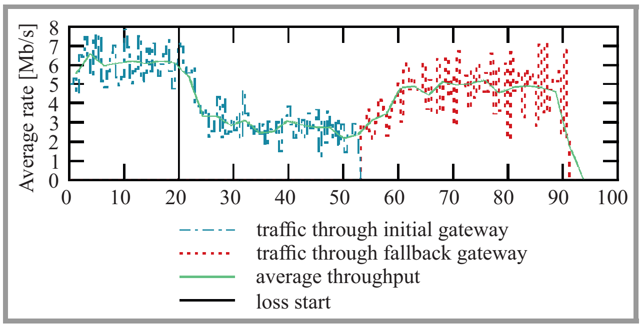

Fig. 7. Seamless gateway switching experiment results.

In the experiment execution, node A initially chooses node B as its gateway, accordingly to lower path cost measured by ETX metric (Fig. 6a). The TCP connection between nodes A and E is established with help of the WiOptiMo subsystem. As can be seen in Fig. 7, at $t = 20$, the degradation of the link quality has caused a decrease in the utility of flows. The higher loss ratio combined with the TCP congestion algorithm operation results with much lower throughput (around 3 Mb/s) then the one observed previously (6 Mb/s). Due to the implementation of the hysteresis-based solution in the OLSRd link quality calculation algorithm, it has taken some time for this drastic change to be reflected in path metric. Eventually, at $t \approx 53$, the OLSR daemon on node A decides to select the alternative path leading through another gateway (node C) to the service provider H (see Fig. 6b).

For the case of service without the mobility support, any connection from A to the external network through NAT-performing gateway would be severed without warning. What is more, the gateway change may occur late or even not at all (e.g., as a result of the *NatThreshold* application).

In contrast, as in this experiment we use WiOptiMo mobility solution, the connections remain intact even when gateway changes. Moreover, a new path allows regaining at least part of previously attained throughput (over 5 Mb/s).

8. Conclusions and Future Work

Integration of DANUMS and WiOptiMo solutions provides better resilience for topology and link-quality changes in multi-gateway mesh networks. Additional flow measurements provided to DANUMS allow improvements in DANUMS operation, in particular help to achieve the DANUMS main goal, i.e., optimization of overall network utility in the case when the flows transmitted to the external network such as the Internet are served.

Future work includes implementation of techniques allowing per-flow and preemptive transitions between routing paths. Loose source routing could be used (at least in the transitional phase) to preemptively change a gateway by WiOptiMo, using information gathered by the OLSR daemon. This task could also be realized by using approach already implemented in existing OLSRd extension – the Multi Smart Gateway – which creates IP-over-IP tunnels between gateways and wireless client [7]. Such a solution could preserve compatibility with routers which treat IP Source Routing header as a second-class citizen due to security and implementation issues.

Acknowledgement

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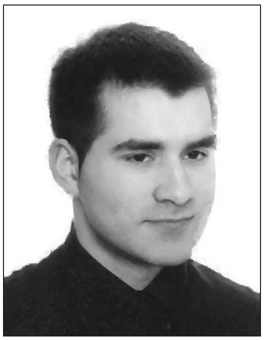

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Paper

New Threats and Innovative Protection Methods in Wireless Transmission Systems

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Abstract—Many improvements in the field of wireless communication can be observed nowadays. Some developments are gradual, others are revolutionary. It is obvious that each innovation in the area may lead to new security threats and vulnerabilities. Such technologies and transmission methods as: Near Field Communication (NFC), Visible Light Communication (VLC), handover, mesh networks, 5G cellular network, mobile IPv6, beamforming, cooperative beamforming, Multiple Input Multiple Output (MIMO), Orthogonal Frequency Division Multiple Access (OFDMA), transmission in Extra High Frequency (EHF) band are very important from the security point of view. In order to preserve high level of security one needs to identify, analyse and classify distinctive sets of threats and vulnerabilities as well as some emerging data protection opportunities related to innovative wireless transmission methods and technologies. This identification, analysis and classification is a main purpose of the paper. It will focus on cryptography in wireless systems, security vs. energy tradeoffs, physical layer security. For example, common problems related to cryptography may be solved with a use of physical layer security. Data confidentiality may be fulfilled with a use of beamforming and jamming, authentication may be performed with a use of out-of-band authentication model.

Keywords—authentication, confidentiality, cryptography, threats, vulnerabilities, wireless transmission.

1. Introduction

1.1. Advances in Wireless Transmission

Wireless, mobile communication systems are commonly used. Number of applications as well as number of technologies are growing. An evolutionary improvements may be observed in such exemplary communication systems as: cell networks, wireless Local Area Networks (LANs), wireless Metropolitan Area Networks (MANs). Some new (often revolutionary) forms/modes of transmission crop up from time to time. The technology progress is a result of greater demand for rich media services and rapid growth of subscriber base.

From the user point of view transmission system should be effective, reliable and secure. Unfortunately, the main drawback of wireless transmission is fundamental lack of data security. There are many intentional as well as non-intentional threats and many vulnerabilities that should be evaluated in wireless environment. Numerous security problems should be solved today, nevertheless pioneering wireless technologies materialize together with new threats and vulnerabilities.

The advances in wireless transmission may be described with a use of three factors:

– universal trends in development,

– constant, evolutionary enhancements in well-established and commonly used systems,

– new forms, modes, techniques (some of them are revolutionary) of transmission.

1.2. Universal Trends

There are some trends observable in the development of different transmission systems. Obviously, the systems progress in time: transmission parameters (throughput, delay, bit error rate, jitter) are improving. Better, wireless communication channel parameters are achieved with advances in many areas, such as:

– silicon technology,

– keying (modulation) techniques,

– spread spectrum techniques,

– signal processing,

– radio bandwidth expansion,

– adaptive arrays and other types of smart antenna,

– multiple access techniques,

– network topology.

Spectral efficiency is growing. Nevertheless it is slowly reaching theoretical limits. Furthermore, hardware (user devices as well as infrastructure equipment) costs are constantly decreasing – this is a result of pushing down the cost of components and leaving the manufacturing to Asian companies, which are good at making products at high volumes and low costs.

1.3. Network Convergence

Next important trend is convergence – different networks and services are merging into single network. Wireless networks management systems are more often based on cloud computing. User mobility becomes a service which is supported by upper layers of protocol stacks (e.g. mobility features in IPv6). It must be noted that all given above progress trends have significant impact on data security.

For example, decreasing cost of hardware is a double-edged sword. Users get cheaper devices. At the same time cost of attacks also decreases and new attack methods materialize. Nowadays, frequently used (and cheap) method of Internet attack (called phishing) is based on false servers connected to Internet and used to attract Web users in order to get their passwords to bank accounts and other confidential data. Decreasing cost of telecommunication infrastructure means that it becomes relatively easy to build and attract users not only to false computer imitating bank server but also to false communication infrastructure, e.g. false LTE femtocells. The fake femtocells delivered by impostor may be used for eavesdropping, denial of service as well as for spoofing purposes.

Another set of security issues is related to convergence. For example, incorporating IP protocol in LTE systems means introducing all security threats and vulnerabilities of Internet to cell phone networks. In the past, voice-dominated, phone networks have been built on proprietary interfaces and protocols (e.g. SS7 signalling protocol) also cell phone networks have been relatively difficult to penetrate, malicious attacks were infrequent in comparison to Internet. Radio Access Network (RAN) and backhaul had complex deployment configurations, specific to operator, location and equipment vendor. Attacks on them required sophisticated preparation and on-site access.

Today, cell phone networks are primarily data networks based on IP with more open architecture and protocols (e.g. diameter open signalling protocol). As a consequence new threats emerge. An example is signalling flood problem, which may be caused either by malicious activity directed at the cell phone network, or accidentally as an indirect effect of upgrades[1].

In such converged network single, modern attack on a mobile device may have impact on [1]:

- cell phone subscriber (owner of the device),
- corporate subscriber network,
- mobile core network,
- Internet.

Today, we are living in transitional phase of Internet. Two versions of Internet Protocol are used side by side: IPv4 and IPv6. IPv4 is old, unprotected, inefficient protocol with some additional drawbacks as limited address space and Network Address Translation (NAT) obstacles. IPv6 is newer, integrated with security tools, more efficient protocol, without address complications, with mobility enhancements. Nevertheless, it must be added that IPv6 is not a matured technology. There are many issues related to IPv6 availability, performance and security. Furthermore, there are some problems related to coexistence of the protocols and to transformation from IPv4 to IPv6 period [2], [3].

1.4. New Transmission Forms, Modes, Techniques

In the last years many innovative technologies utilized in wireless systems are appeared. First of all new and diverse bands of electromagnetic waves are incorporated into transmission, e.g.: NFC (13.5 MHz), EHF (30–300 GHz), VLC (428–750 THz).

Electromagnetic waves are main but not only communication medium. Transmission systems based on acoustic energy are also developed[2].

Another area of progress is related to space radio coverage. A lot of research and implementation work is done in such issues as:

- mesh networking,
- cooperative relaying,
- beamforming and cooperative beamforming,
- IP mobility,
- handover processes.

1.5. Inherent Lack of Security in Wireless Networks

Computer system security is frequently defined (e.g. in ISO standards) with a use of three general factors: confidentiality, integrity and availability (called sometimes CIA). These requirements incorporate such elementary security controls as: authentication, authorization, accounting, replay protection, Man in the Middle (MiTM) protection, non-repudiation. The requirements are independent of technology. They should be fulfilled in wired as well as in wireless transmission systems. Nevertheless, wireless networks distinctive features make it is much harder to satisfy the CIA requirements.

First of all, data confidentiality may be threaten by eavesdropping. In ordinary radio communication system eavesdropping is much easier since wireless communications systems have usually broadcast nature – radio signal is available for everyone in the range of the transmitter. Data integrity is a function of transmission correctness. Transmission systems based on wireless medium have much

[1]In January 2012, NTT DoCoMo in Japan experienced a signalling flood that disrupted network access, caused by a VoIP OTT application running on Android phones [http://www.reuters.com/article/2012/01/27/us-docomo-idUSTRE80Q1YU20120127].

[2]For example, Microsoft is working on Dhwani Project, which uses acoustic waves for short range communication (http://research.microsoft.com/apps/pubs/default.aspx?id=192134).

higher (up to 10^7 times greater) nominal bit error rates than transmission systems based on copper wire or optical fibre (Table 1).

Table 1
Bit Error Rates

Technology	BER (absolute)	BER (relative)
Wire		
Gigabit Ethernet	10^{-12}	1
Fibre Channel		
Wireless		
Satellite (GEO)	10^{-8}	10^4
Satellite (LEO)	10^{-6}	10^6
Point-to-point	10^{-5}	10^7
IEEE 802.11	10^{-6}	10^6
IrDA	10^{-8}	10^4

An attack on availability of communication services is called Denial of Service (DoS). There are many methods used to perform DoS or distributed DoS in computer networks. Exemplary attacks use some features of communication protocols from higher layers of TCP/IP stack, such as: Address Resolution Protocol (ARP), Transmission Control Protocol (TCP) or Domain Name Services (DNS). Such attacks may be executed in all networks without regard to transmission medium. In addition to mentioned above attacks based on higher layers, DoS in wireless network may be performed at physical layer by jamming radio channel [4].
Intentionally generating jamming signal is relatively simple and inexpensive. Furthermore, jamming may also result from non-intentional reasons. Radio bandwidth (especially unlicensed part of it) is limited resource. Growing number of systems using the same unlicensed radio band (especially 2.4–2.5 GHz) leads to increasing interference risk. Unlicensed ISM (Industry Science Medicine) band is utilized by such systems and applications as:

– IEEE 802.11 (Wi-Fi),
– IEEE 802.15.1 (HR WPAN),
– IEEE 802.15.3 (Bluetooth),
– IEEE 802.15.4. (ZigBee),
– HomeRF,
– RFID,
– microwave ovens.

A lot of work is done in search of new, higher and uncrowded EHF (30–300 GHz) frequency bands (e.g. IEEE 802.11ad already uses 60 GHz). But, it must be noted that wave propagation at these frequencies is related to entirely different set of difficulties, e.g. increased free space path loss. Channel models developed for systems using the 1–3 GHz microwave bands are inadequate to characterize wireless systems with 10 or even 100 times greater carrier frequencies. Interference problem is additionally difficult to solve since signal bandwidth of a single channel is also much wider than before (see Tables 2 and 3).

Table 2
Signal bandwidth growth in WLAN with IEEE 802.11

IEEE 802.11 version	Single channel bandwidth
802.11a	20 MHz
802.11b	22 MHz
802.11n	40 MHz
802.11ac	80 MHz
802.11ac (option)	160 MHz
802.11ad	2 GHz

Table 3
Signal bandwidth growth in cellular phone systems

Cell phone system	Single channel bandwidth
Tetra, NMT	25 kHz
GSM	200 kHz
UMTS	5 MHz
LTE	20 MHz
LTE advanced	100 MHz

1.6. Common Security Controls

Some security tools and methods for wireless networks are used for years. The main of them are:

– hidden channels in Frequency Hopping Spread Spectrum (FHSS),
– communication range control methods as sector antenna, transmit power restrictions, different forms of Faraday cages, e.g. shielding paints,
– Medium Access Control (MAC) address filtering,
– data encryption,
– Wireless Intrusion Prevention System (WIPS) and Network Admission Control (NAC).

2. Cryptography in Wireless Networks

Data encryption is very powerful and widely utilized protection against eavesdropping. There are many encryption techniques and protocols used in higher layers of TCP/IP stack: IPSec, Secure Socket Layer/Transport Layer Security (SSL/TLS), Secure Multipurpose Internet Mail Extensions

(SMIME), Pretty Good Privacy (PGP) and so on. Nevertheless, the emergence of large-scale, dynamic and decentralized wireless networks imposes many new challenges on classical cryptography.

In wireless environment there are some inherent impediments related to cryptography applications. Data encryption/decryption is restricted in mobile devices with low processor performance and energy limitations. Furthermore, key management is complex and delicate matter especially in heterogeneous environment with many different protocols and networks, i.e. IEEE 802.11, IEEE 802.16, UMTS, LTE. Cryptography has a negative impact on network throughput. Some protocols devised for wireless networks, e.g., Wired Equivalent Privacy (WEP) proved to be very vulnerable to cryptanalysis.

An example, of decentralized wireless network is a mesh. Some specific challenges related to security are visible here [9]:

– secure multi-hop routing,
– detection of corrupted nodes,
– denial of service attacks,
– fairness factor of the distribution of network resources.

From the cryptography point of view, an important impediment is links heterogeneity and devices organized into a mesh. The protection of the communication between non-neighboring nodes is more complex. It requires the use of integrity and/or encryption on a higher protocol layer than the MAC. Furthermore, different wireless technologies such as IEEE 802.11, IEEE 802.16 used in a mesh may support different algorithms with different cryptography strength [10].

Even when message confidentiality is provided by standards with higher level of security like WPA (Wi-Fi Protected Access) or WPA2, usually only the frame payload is protected and MAC address in the header of the frame is transmitted in unencrypted form. So, traffic analysis by frame sniffing may be used to monitor and track users in the network.

Table 4
Exemplary sizes of data fields in IP packets

Application	Data field size [bytes]
Internet games	40–110
VoIP with LP codec (e.g. G.728)	50–80
Ping	56
World of Worldcraft	74
Skype	84
VoIP with PCM codec (e.g. G.711)	180–260
BitTorrent	377
eMule	1180

Furthermore, complex traffic analysis based on entire frame size may give some hints on application that is used by a given user. The frame size is related to data field, which contains IP packet – different network applications send packets with different number of bytes in data fields (Table 4) [5]–[8]. It has been demonstrated that traffic analysis based on IP packet sizes may be used to infer some information, e.g., the source of a Web page retrieved by a given user or applications run by the user [11].

3. Physical Layer Security

As an alternative to data encryption in higher TCP/IP layers the physical layer characteristics of the wireless channel such as fading or noise may be used to protect confidentiality. Beyond securing wireless transmissions of confidential information, physical layer security solutions have been also exploited to provide or enhance the authentication and privacy of legitimate wireless users [12]. There is a lot of research work in the area of physical layer security. The main research topics are:

– code design for physical layer security,
– advanced signal processing and space-time secure transmission techniques, e.g. secure OFDMA subcarrier allocation [13],
– secure relaying and cooperative transmission techniques,
– advanced physical layer security attacks, e.g., smart eavesdropping or jamming, and their countermeasures,
– physical layer authentication.

It must be noted that the use of physical layer for data protection originates from Shannon's notion of communication channel information capacity and perfect secrecy.

4. Energy vs. Security

4.1. Energy Gap

Energy available in mobile devices is a limited resource. Device performance and its internal complexity are rising. Smartphones, tablets and other mobile devices are equipped with more and more functions demanding energy. On the other hand capacity of batteries produced for the devices is growing but the improvements are very slow. As a result power gap between energy consumption and battery lifetime is not decreasing.

The energy gap between energy demands and availability is widening. The gap has negative impact on data security. First of all the security tools such as encryption/decryption, key management, firewall, antivirus working in mobile devices utilize a lot of energy. Switching on the security tools may double the energy used by a given device. Authentication, Authorization, Accounting (AAA) processes consume energy, especially in device that changes location and has to make frequent disassociations and associations with many access points (APs) or base stations (BS) [14]. Secondly, device energy may be a resource that is intentionally attacked.

Here we have some examples of an old dilemma – functionality vs. security. Higher security level equals to shorter battery lifetime.

4.2. Energy Usage by Security Tools

It has been proved that there is no big difference between wireless protocols in terms of energy consumption per crypto operation. The energy consumption related to cryptography is dependent mainly on key length (independently on the selected algorithm) [15]. Longer battery lifetime (more functionality) means short cryptographic key, short cryptographic key means low level of security.

Additional problem is related to user behavior. In order to increase battery life of the device one has to switch off some or all security tools – for a second time, functionality increases while security decreases.

4.3. Intentional Threats Related to Energy

From energy point of view two forms of intentional threats and malware may be distinguished:

- common malware utilizing available energy as side effect,
- dedicated malware for mobile device energy resources in order to exhaust it.

Each common form of attack and malware (e.g. simple port scanning or ping flooding) utilizes energy of the device that has been attacked. The device is forced by an attacker to perform additional tasks like executing code inserted by intruder or sending data to some unexpected by an owner of device destinations. If an attack is persistent for a long time the energy of the device is slowly vanishing. The experiments [14] demonstrated that port scanning attacks or ping flooding attacks may double the power consumption of an exemplary Android based smartphone.

One of the critical categories of attacks is DoS. As a result of such attack user is unable to use his device, resources or services. There are many methods for DoS attacks – they are routinely performed in Internet. The attacks utilize and drain some resources of attacked system, e.g., memory for communication buffers, processing power, throughput. In the case of wireless, mobile device DoS attack may be completed by quickly exhausting the energy of a device. This may be done by:

- creating unsolicited network traffic,
- forcing erratic and CPU consuming behavior,
- utilizing power consuming services, especially GPS or Bluetooth communication [1].

4.4. New Protection Tools and Methods

Common methods for malware detection and for intrusion detection systems are: signature based scanning and heuristic scanning. In the wireless and mobile environment a new technique for malware and attack detection is possible. Anti malware system may use detailed data on current energy usage in the device to detect malware or attack. Normal energy usage profile may be identified, stored and compared with recent usage. Deviations from this typical profile may indicate an infection or an attack.

Another important research area is related to energy savings. In order to save the device energy consumed by security tools some changes are necessary. The following solutions are considered:

- security controls offloading,
- some modifications to existing malware detection methods, e.g. based on virtual machines,
- security tools with decreased energy usage.

4.5. Offloaded Security

In order to preserve mobile device energy some tasks of the device may be moved from the device to another component of IT system. The security tasks may be offloaded to server or cloud [16]. The scenario of malware or attack detection process may look as follows:

- operation (function call, signal) is logged in user mobile device,
- system log is transmitted from the device to server,
- server performs the same operation in simulated user terminal environment and checks security points,
- result of the checking is transmitted from the server to the device.

Offloading security controls has some drawbacks. Energy for data processing (related to security check) in the device is saved but at the same time extra energy is utilized for device-server and server-device transmissions. Furthermore, malware detection in server has to be done with a use of signature-based method, which is vulnerable to some sophisticated malware attacks using stealth techniques, polymorphism or encrypted code. In order to detect such attacks signature-based method should be supported by heuristic detection implemented directly in the device and obviously draining some energy of the device.

4.6. Energy Limited Security Tools

Another important research area is designing security tools with decreased energy usage. This may be done by decreased number of control points for malware detection, or decreased frequency of control checks.

It must be noted that some research teams are working on more general solutions to energy problem. For example, researchers from Worcester Polytechnic Institute are working on analytic 3-dimensional model of relations between data security and energy consumption. The model takes into account a given attack countermeasure and the level of security-reliability it can provide and relationship between the energy spent in carrying out a countermeasure and the energy level that is potentially lost if a given attack is successful [15].

5. Handover vs. Security

Handover is a process for switching wireless network while mobile device is moving from the range of one access point (or base station) to the range of another access point (or base station). Signaling processes may be executed in many layers. MAC signaling is performed with such protocols as IEEE 802.11i or IEEE 802.11r. IP signaling utilizes mobile IPv6 options.

Handover processes may be performed while the device is moving between two access points utilizing the same protocol and the same radio band or while the device is changing access network e.g. from IEEE 802.11 to UMTS. The processes are supported by IEEE 802.21 Media-Independent Handover Services.

Handover processes perform some functions related to security, like: authentication, key management. For real-time services handover delays should be kept minimal (at the level of several tens of milliseconds). Unfortunately, delays are usually much greater, in some cases up to several seconds.

Solutions to handover delay problems are based on predictions of awaiting handover and performing some processes related to handover in earlier times. Furthermore, time may be saved by eliminating unnecessary IP handovers (e.g., when user is roaming among base stations connected to the same access router) and by combining the mechanisms in the MAC layer with that of the IP layer [17].

An example is Handover Keying (HOKEY) proposed by Internet Engineering Task Force (IETF) Working Group [18]. It is based on modified Extensible Authentication Protocol (EAP), with decreased number of messages sent between parties. Keys from previous sessions are used in order to re-authenticate device with next access point. The process is initiated, while the device is still in the range of former access point antenna.

Modified Kerberos authentication for handover is proposed by Ohba *et al*. [19] for secure key distribution. Mobile node obtains master session keys without communicating with a set of authenticators before handover. Signalling related to key distribution is based on re-keying. The process is separated from EAP re-authentication and AAA signalling similar to initial network access authentication.

6. Secrecy Capacity

6.1. Quantitative Security Measure

Security (confidentiality) level of wireless network may be measured not only qualitatively but also quantitatively. Secrecy capacity is a measure for confidentiality level based on communication channel information capacity. Channel information capacity defined as the tightest upper bound on the rate of information that can be reliably transmitted over a given communication channel.

Secrecy capacity is defined as difference between the channel capacity of the link between sender and legitimate receiver and the channel capacity of the link between the same sender and eavesdropper. Value of the parameter is related to Signal to Noise Ratio (SNR) difference between the legitimate receiver and eavesdropper. It must be noted that some theoretical foundations of wireless security were presented many years ago, e.g. [20].

So, in order to increase confidentiality level this SNR level of the communication link between the sender and eavesdropper should be decreased. In systems with MIMO and antenna arrays this may be accomplished by:

– beamforming,

– transmit antenna selection [21],

– sending the jamming signal in the direction of the eavesdropper.

6.2. Beamforming Solutions

Common beamforming may be used to improve secrecy capacity. SNR difference between the legitimate receiver and the illegitimate receiver may be increased in order to minimize eavesdropping risk. Beamforming for security may be used in some different modes with assumption that the location of the eavesdropper is known, or without such assumption, e.g. [22]–[24].

Common beamforming system uses antenna array integrated with a single node. Cooperative beamforming [25] is a transmission mode in which in randomly distributed nodes antenna array is created with antennas from many nodes. The main purpose of such a system is data transmission on long distances in energy-efficient way.

Cooperative beamforming may be used to increase secrecy capacity. The protection is based on a subset of intermediate nodes which adopt distributed beamforming for sending information to legitimate receiver. At the same time other nodes send jamming signal to eavesdropper. Both tasks are accomplished with preserving individual power constraints of the nodes [26].

6.3. An Intelligent Jamming

Intelligent jamming is a security strategy aimed at the potential eavesdroppers locations. If the locations of eavesdropper and legitimate receiver are different then jamming signal may be directed. Jamming signal is sent (by sector antenna or by antenna array with beamforming) in the direction of the area of eavesdropper. At the same time legitimate receiver is not receiving this jamming signal. The method can effectively raise the noise floor at eavesdropper position, which makes for him difficult to distinguish between wireless signals and normal background noise on the wireless medium.

7. Out-of-Band Authentication

Standard electromagnetic waves communication channel may not be considered as trusted or authentic. Electromagnetic waves may be send from long distances, from different places which may be hidden. In general, user receiving such signals is not able to determine their source and may not be assured that a given signal is transmitted by certain sender. So, reliable procedure for sender authentication becomes a challenge. Unsecured authentication may lead to eavesdropping or spoofing.

The authentication problem (also cryptographic key for symmetric algorithms exchange problem) is usually solved with a use of asymmetric cryptography, e.g. Public Key Infrastructure and certificates. Nevertheless, there are some problems related to cryptography, e.g. the complexity of key management problem, especially in wireless and heterogeneous environment. So, another authentication method becomes crucial.

Out-of-band authentication is an authentication mode that uses additional communication channel/method for authentication processes. It is assumed that the system utilizes common unsecured radio channel (with high bitrate) for normal data transmission and the second out-of-band channel with relatively low bitrate, e.g. acoustic channel or visual channel like IEEE 802.15.7), for authentication purposes only.

The second channel should be authentic: receiver is guaranteed that a message he receives actually was send by a given sender. It is important that human (not the electronic device) is able to verify the authenticity of the sender device. The requirement may be satisfied in some different ways:

– electric contact,

– NFC,

– relative location check with a use of ultrasound impulses for measurement,

– visual markers photographed with camera smartphones, e.g. Seeing is Believing (SiB) method,

– common motion of two devices and accelerometer data analysis,

– acoustic, e.g. Loud&Clear (L&C) method.

Experiments with the given above methods are carried on. An exemplary protocol for creating an out-of-band channel for authentication with visible laser light has been proposed in [27]. The authors assume the laser transmission is not confidential. An attacker is able to either violate the confidentiality of data transmitted by VLC or to violate its authenticity. Proposed protocol, based on off the shelf components (so relatively cheap), establishes a secret, authenticated shared key between the personal trusted device and a remote device.

VLC channel is used also in the method proposed by Mayrhofer *et al*. [27]. The devices are equipped with visible barcodes that encode hashes or public keys. The data from the barcode of the first device are read by taking photo by the second device. User is aware what is photographed by the first device. The same data are transmitted by radio channel. Two sets of data received with a use of two different channels are compared in order to authenticate the second device.

Acoustic channel may be used in a similar way. Authentication data are played by one device and heard by the other device.

8. Conclusion

The author have presented several issues related to latest innovations in wireless communications systems. Each innovation is related to new threats and new vulnerabilities. At the same time the innovations may be used as new opportunities for data protection.

There are many protection methods and tools used widely in wired networks. Many of them are used also in wireless environment. Nevertheless, one has to be careful transferring security controls between different networks with different devices (including mobile ones). The features like limited energy, limited processing power and mobile device memory of wireless networks and mobile devices are unlike wired networks and immobile devices. So, security controls should be cautiously chosen and adjusted according to the features of this different environment.

An important and open, scientific research area is related to physical layer security in wireless networks. Upper layers security controls that are hard to use in mobile environment (e.g. cryptography) may be replaced by or used together with physical layer controls, for example: beamforming, dedicated subcarrier allocation in OFDMA, transmit antenna selection, jamming the eavesdropper, authentication with a use of out-of-band communication channel (e.g. VLC, NFC, SiB, L&C).

A lot of work is to be done in the area related to energy-security trade-offs. Such issues as models and standards for these relations, power management methods, off-loading security mechanisms from mobile devices are the exemplary topics. Research on new security tools and methods with reduced power consumption is necessary.

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Paper

Lessons Learned from WiMAX Deployment at INEA

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Abstract—Home broadband access is continuously demanding more bandwidth fueled by video streaming, entertainment and gaming applications. In 2010, a company INEA decided to roll-out new WiMAX-based services aimed to meet the needs of home users across the Wielkopolska region of western Poland. It was decided to follow the 802.16e standard and the Time Division Duplexing (TDD) mode that offer the ability to adjust the downlink/uplink ratio and thus are well suited for data transmission. After an extensive testing period of equipment from various vendors, engineers at INEA have chosen the Motorola (currently Cambium Networks) PMP320 solution because it is compact and its components are space- and energy-efficient. The company choice was also influenced by its simple operation, management and installation, which ensured low costs of ownership. So far, this deployment has provided fast and affordable connectivity for Internet and telephony services to around 5,500 households across the 30,000 sq. km region. After 3 years of experience, INEA would like to share the lessons learned from this roll-out.

Keywords—interference, radio capacity, radio planning, WiMAX.

1. Introduction

Currently mobile WiMAX, as defined by the IEEE Standard 802.16e-2005 [1], is a well-established wireless technology for providing both fixed and mobile access. In Poland, most of WiMAX deployments use 3.400–3.800 GHz band. This follows a decision to publish a public tender for 3.5 MHz channels from 3.600–3.800 GHz band issued by President of the Office of Electronic Communications (UKE in Polish) in 2007 [2]. The Marshal Office of the Wielkopolska Region decided to participate in this public procurement, and is currently sharing its channels with the local operators: INEA, Promax, and ASTA-NET. Despite the fact that the 802.16e standard supports mobile access, the Polish WiMAX networks provide only fixed or nomadic access. This is due to high cost of achieving coverage that would offer mobile access in 3.600–3.800 GHz band.

Since 2010 INEA has started to roll-out its WiMAX 802.16e network. The network is based on the Motorola (currently Cambium Networks) PMP320 solution and supports mostly data and VoIP transmissions. An example of INEA's base station is presented in Fig. 1.

As it is shown in Table 1, the INEA WiMAX network currently supports 5,500 users. The network consists of 72 WiMAX stations and 252 access points (AP) being in-

Fig. 1. One of INEA's WiMAX base station.

Table 1
INEA WiMAX network in numbers

	In total	In Poznań
Stations/Towers	72	11
Access points/Sectors	252	32
CPEs	5500	1000
Average CPEs per AP	22	30

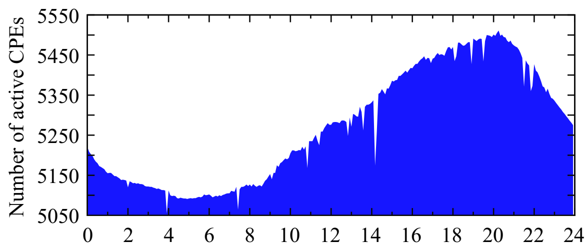

Fig. 2. Number of active CPEs (users) within 24 hours on 26 January 2014.

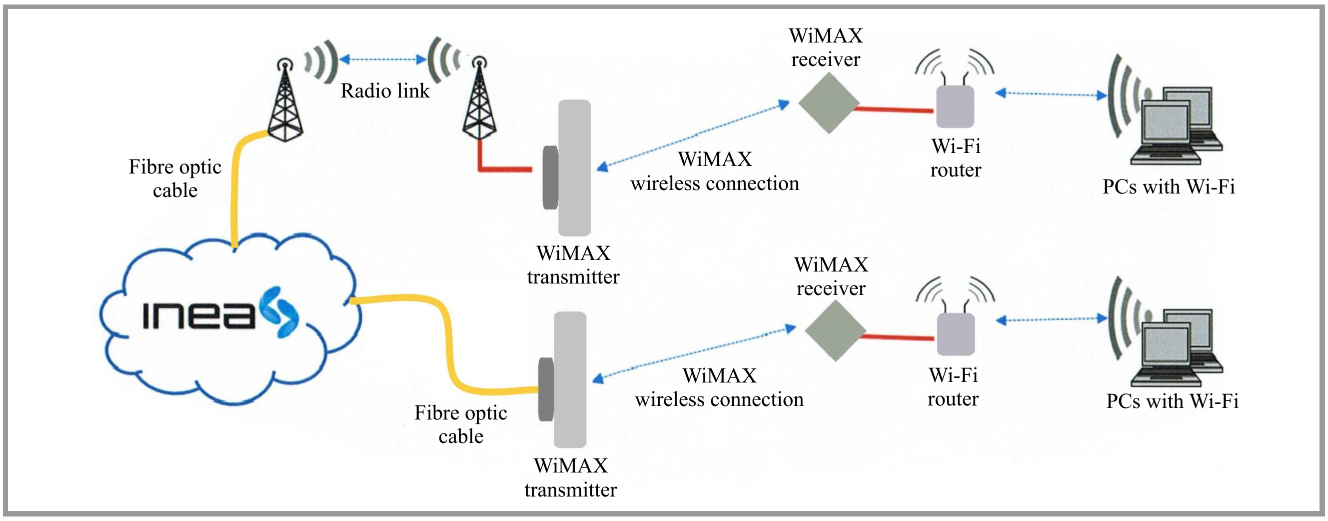

Fig. 3. General topology of INEA WiMAX network.

stalled across the region, which gives the average number of 22 clients per AP. In the city of Poznań it provides services to more than 1,000 clients, using 11 base stations with 32 AP. Figure 2 shows the changes in the number of active users within 24 hours.

The connection between our core network and the WiMAX station is executed either using a microwave link or optical fiber connection. The latter option is most commonly used by INEA, since it is more stable and resilient. The base station realized using Cambium Networks PMP 320 products is composed of: Cluster Management Module 4 (CMM4), GPS antenna, and a number of access points with attached antennas. The CMM4 is a managed L2 (or L3) device that provides power and GPS synchronization for APs. Physically, it consists of two separate devices: EtherWan and Motorola CMM. EtherWan has 12 FE (Fast Ethernet) ports and 2 GE (Gigabit Ethernet) ports.

PMP 320 AP is a Layer 2 (or Layer 3) device and it has a 90 degree antenna attached. The configuration of Transmit/Receive Transition Gap (TTG) and Receive/transmit Transition Gap (RTG) guarantees to achieve a maximum range of 15 km for 7 MHz channel. The definitions of INEA's set of service flows are stored on each and every AP.

Customers are provided with Cambium Networks 3630SM (or 3530SM). This CPE (Customer-Premises Equipment) is a Layer 2 (or Layer 3) device controlled by SNMP and HTTP. The device allows to achieve maximum throughput of 11/5 Mb/s. Customers also have the ability to use VoIP services using dedicated ertPS (Extended Real-Time Polling Service). This service is realized using Linksys SPA2102 VoIP gateways. A diagram of the network is shown in Fig. 3.

In this paper the technical lessons learned from the roll-out are presented. The article does not intend to present any procedural or psychological challenges which the authors needed to overcome in order to allow INEA – cable operator to introduce wireless technology into its portfolio.

Despite quite extensive number of research papers [3]–[11] which evaluate theoretical performance or present system level analysis of WiMAX systems, authors of this paper are not aware of similar deployment analysis being published so far.

2. Radio Planning

Radio planning is an essential aspect of maintaining optimal network coverage and capacity. In real WiMAX networks, especially in high density areas, radio planning is a major challenge for the operator. Before setting up each station, radio planning should at least consist of the following elements:

– a determination of the total coverage of each base station and each of its sectors separately,

– a preparation of the list of addresses in coverage area of each base station (BS),

– channel allocation for (new or changed) sectors of each station,

– a determination of (or a change in) the set of radio parameters for each coverage sector: azimuth, tilt and other constraints if applicable.

The key task is to choose a proper method for a determination of areas to be served by each radio station (access point). This procedure is called a "coverage prediction", but the term is rather meant to denote the use of one of commonly used propagation algorithms and models. Previous studies (e.g. [3], [6], [7], [9]) have show that commonly known algorithms cannot be simply used for WiMAX networks. Summarizing the conclusions of mentioned papers, for gigahertz bands and WiMAX standard some of the algorithms give results close to real performance in urban area while others in suburban or rural area. The key point is modeling the path loss in LOS and NLOS condition:

each of the models under studies (Ericsson 1999, COST, Free Space, Okumura, SUI and other) should be used for well defined signal path conditions – either LOS or NLOS. But in real life scenario both propagation conditions occur simultaneously for each access point sector. Over the last three years even the authors have been continuously working on improving the propagation models (starting from ITU-R P.1546 and ITU-R P.1812, SUI [4]). Despite the three-year effort, the real live conditions didn't match with the theoretical results.

Hence, INEA's radio planning team was forced to create its own unique procedure of finding service areas round the station. The procedure nowadays uses as a basis a combined model composed of a modified free space model and a diffraction term in order to match real-live installation success and failure rates.

Having taken into account signal levels measured by terminals (CPEs) installed in different locations, including different cities, towns and villages, the team were in position to construct a useful algorithm based on an algorithm previously constructed and a selected model parameter list. The basis for the model was provided by one of broadcast models used in Germany [12]. Because of various installation and propagation conditions, such a model obviously does not give accurate results, but thus prepared prediction is good enough to become a basis for acquiring new subscribers in a given area. To minimize the risk of launching a marketing action in a wrong part of the area, it was divided into two sub-areas: the zone near the station, where the probability of good reception (in both directions of transmission) is high, and the outer ring, where the risk of unsuccessful installation is higher, but still there are reasonable grounds for an attempt to acquire new subscribers. In Fig. 4 both areas are depicted in two shades of gray. The described method for a determination of the coverage around each station gives surprisingly good results.

Fig. 4. Example of results of coverage prediction.

Another set of radio planning issues is the channels allocation for each sector. This is relatively simple for rural sites, or generally for sites located far away from other stations using the same channels. In such cases the network may use the widest possible radio channels provided by the equipment (PMP320 AP and CPE), i.e. 10 MHz. A wide channel provides high throughput for subscribers but, at the same time, has some disadvantages as well. The basic quantum of spectrum given by the regulatory office is 3.5 MHz. Thus, for 10 MHz useful spectrum a four such basic neighboring elements were used, which means that $4 \cdot 3.5 - 10 = 4$ MHz (two sections of 2 MHz each) remains unused. In very dense areas the problem is not trivial, because the channel allocation layout may look like the one depicted in Fig. 5 where the same radio channels use the same hue.

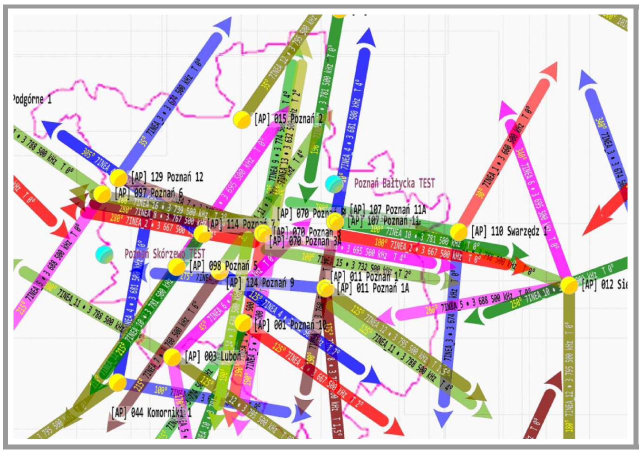

Fig. 5. Example of a dense network in large urban area of Poznań.

In such cases the 4 MHz of spectrum can not be wasted, as described above. Therefore, the authors had to decide to narrow the WiMAX radio channels down to 7 MHz. Fortunately, it still can provide the same throughput for a single subscriber as the 10 MHz channel, and the use twice as many channels as before is possible. Moreover, we have experienced that, thanks to good characteristics of the currently used WiMAX radio equipment, the use two neighboring "halves" of previous 14 MHz block for two sectors of the same station – even if their azimuths are relatively close to each other is still possible.

The introduction of each new station to the network might force some relocation of existing radio channels. Such changes should be performed with care. The issue is not difficult if a new station is created in the area so far unsupported. In some cases, however, a rapidly growing number of subscribers forces us to built a new station (or a new sector on the same location, parallel to the existing one). Sometimes the expansion can be predicted, so particular channels were often reserved for it in advance. In some areas the services offered by INEA are so attractive that the number of subscribers grows much faster than it is expected. It is always good news, but it might necessitate a total reconstruction of the radio channel network in the whole urban agglomeration. Interestingly enough, that such situations occur usually in large centers, such as Poznań and Konin, that already have good wire-line infrastructure in place and thus can hardly be regarded as having no existing access to the Internet (being thus a potential market for wireless services).

Each new WiMAX station in dense population areas, with existing WiMAX stations and subscribers, causes changes of assignments to a service station for many of the CPEs. If a new station is located so that the direction from subscriber location toward the old and the new station is almost the same, such a reassignment is straightforward. However, if both directions differ significantly (for instance, the difference exceeds a change in signal levels of 2 dB, taking into account the antenna patterns and change in the distance between CPE and AP), we should try to rotate the subscriber terminal antenna in a new direction. In some cases it is not possible, or too expensive, so the subscriber must remain with the assignment unchanged. The above describes one of existing limitations of the network optimization.

When a radio network grows, the possibility of co-channel interferences grows up as well. To avoid this as early as at the planning stage of the radio channels assignment, we have to take into account interference range prediction. For this purpose the ITU-R P.1546 algorithm that yields reasonable results is still used. Figure 6 shows an

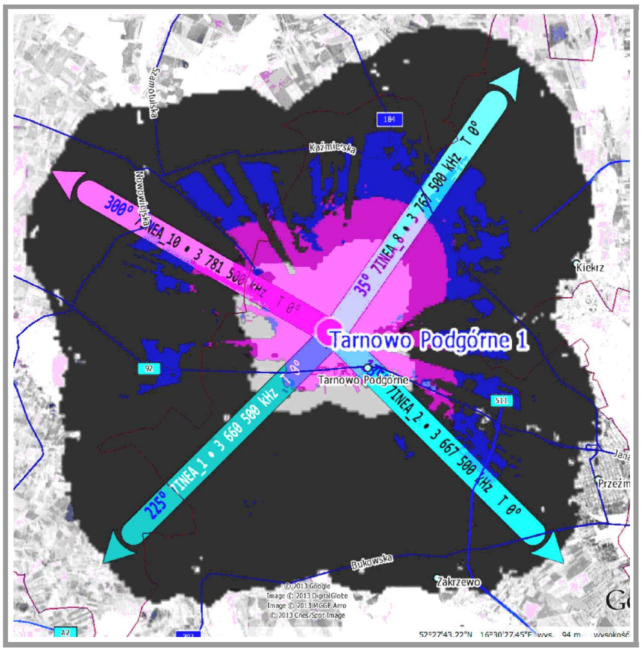

Fig. 6. Predicted interference areas for non-synchronized (other than our own) networks and for the own network of the operator.

example of interference range prediction for a typical four-sector WiMAX station compared with useful coverage of the same station (light gray areas in the center, for explanation see above comments). Light and dark gray regions distinguish between non-synchronized co-channel interference areas and the synchronized ones.

The interference problem is particularly important in high density areas (Poznań, Konin, further expected), not only because of the density of deployed stations. Because of the method of changing CPEs assignment described earlier, after creating a new station the operator should reassign a significant part of subscribers to the new site. As stated earlier, it is possible only for a fraction of existing CPEs. As a result, there still exist terminals associated with a "far away" station using channel interfering with the same channel used in a "near" station. In high density areas neighboring stations are located only a few kilometers apart, so this situation is relatively frequent. For such areas the radio planning team prepares a special "good assignment" map that shows areas of service for each particular sector. The operator should try to force all the terminals located in particular area to use an appropriate sector. It helps the operator keep the interference risk as low as possible. Figure 7 shows an example of the described "good assignment" map.

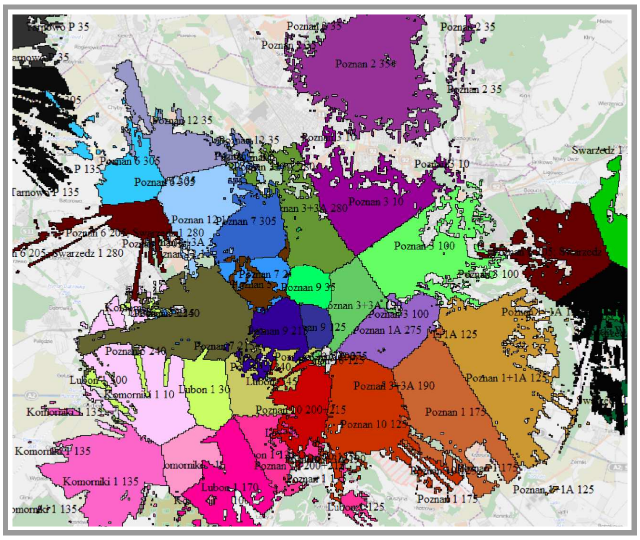

Fig. 7. "Good sector assignment" chart of Poznań urban area.

Lessons learned:

- although for the WiMAX network, where the propagation path looks very different for each CPE and there is no single algorithm for determining coverage, a reasonable method for a prediction of a serving area for each sector and radio station have been found;
- in some areas saturation of a sector occurs faster than expected – we may support the sector with the parallel one, but a better solution is to predict such situation in the initial study phase; otherwise some problems with network optimization and radio channels resources may ensue;
- WiMAX network expansion is favorable for the operator and the subscribers; in dense areas, interference during switching the CPEs into a new station are hard to avoid. The radio planning team should cooperate with network administrators in determining all the possible conflicts.

3. Interference

Interference is often identified as a key cause of performance degradation in wireless networks [5], [8]. In presented network the GPS synchronization to eliminate self-interference between neighboring APs is used. Nevertheless, the network still requires a careful radio planning in order to minimize self-interference. At the time, this was a particularly pronounced problem when the number of base stations in Poznań is increased.
In [13], the vendor of the PMP320 platform provides insights into what needs to be done in order to minimize interference. Among others, the following actions that help reduce interference, are listed [13]:

1. AP down-tilt,
2. Lower AP transmit power,
3. Re-orienting AP sectors,
4. CPE up-tilt to tower on especially short links as required,
5. CPEs registered to correct AP sector,
6. Lowering AP Recv target from –70 to –73 dBm.

In day-to-day operation the first 5 actions is used. We also wanted to use the sixth method. So we were ready to trade uplink CINR and RSSI values for lower interference observed by other APs operating on the same channel. Therefore, after some extensive testing, the AP target receive level was decreased from –70 to –75 dBm. Despite the effort, however, we have never observed avg uplink CINR to increase on other APs.
In summary, a proper radio planning is a key technique of interference mitigation.

4. Capacity

Capacity in a WiMAX network is not fixed [10]. Each CPE operates with spectral efficiency that changes in time and is a product of three parameters: modulation, Forward Error Correction (FEC) coding and Multi Input Multi Output (MIMO) mode. Therefore, the radio capacity of a given AP is a function of the number of CPEs and their spectral efficiency. For example, if one CPE operates with modulation of 64 QAM, then its spectral efficiency is 6 b/s/Hz. If it is using 5/6 FEC, then the spectral efficiency is reduced to 5 b/s/Hz. If it is using MIMO-B, then its spectral efficiency is doubled and reaches about 10 b/s/Hz. The INEA WiMAX network uses 10 MHz channels. When all active CPEs use 64 QAM5/6, then the capacity is about 43.2 Mb/s. Thus, if we have 10 CPEs operating at 64 QAM5/6 and all are fully busy, then each of them is able to achieve 4.3 Mb/s resulting in the total capacity of about 43 Mb/s. However, when two new CPEs connect to give AP with modulation QPSK1/2, then the capacity drops to 21 Mb/s and each CPE is able to achieve about 1.8 Mb/s. This explains why spectrum is a valuable resource and needs to be properly managed.
Because the three parameters: modulation, Forward Error Correction (FEC) coding and MIMO mode are changing quite frequently, hence the capacity of a WiMAX channel and its utilization also changes significantly. This can be observed in Fig. 8 that demonstrates the utilization of OFDMA symbols in the WiMAX frame.

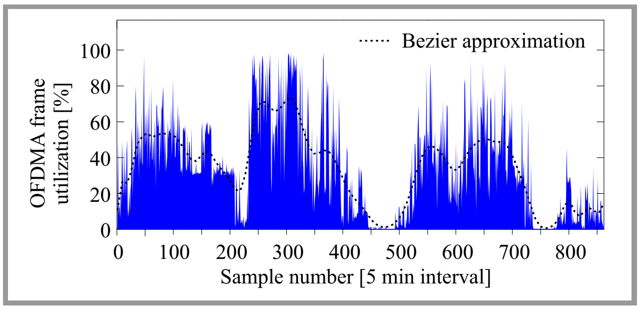

Fig. 8. Example radio utilization of one AP.

In order to maintain high capacity, the network operator needs to control modulation, FEC and MIMO mode. Moreover, WiMAX 802.16e products do not implement load balancing functionalities. Soft handoff of CPE allows the connection to be re-established though does not offer load balancing functionality across APs covering the same area. Due to this limitation, INEA has implemented the following mechanisms that help maintain high capacity:

- fixed assignment of CPEs to AP by fixing frequency or BSID,
- assignment of CPEs to AP, which allows for the highest modulation, FEC and MIMO mode,
- monitoring of radio capacity (utilization of OFDMA symbols).

It is only when the operator uses mechanisms similar to those developed by INEA that it is possible to maintain a high capacity WiMAX network. Moreover, only properly planned and restricted installation conditions allow the operator to fully exploit MIMO multi antenna technology and OFDMA technologies for providing maximum throughput in multi-path environment. Chaotic roll-outs result in networks with unpredictable capacity level.

5. Signal Strength

In the INEA WiMAX network each AP is transmitting at maximum power, but CPEs transmit power is not fixed and it is controlled by Auto Transmit Power Control (ATPC). ATPC is a mechanism implemented by vendor (Cambium Networks) in order to allow each AP to control the output power of all connected CPEs. AP tells each CPE to transmit with such an output power so it can receive it at the target receive level of –70 dBm. It was expected that out of N active CPEs, the k-furthest CPEs (or those that were operating in non-line-of-sight (non LOS) environment) would transmit at the maximum power (+27 dBm), whereas the other remaining $N-k$ CPEs would operate with reduced power, as shown in Fig. 9.

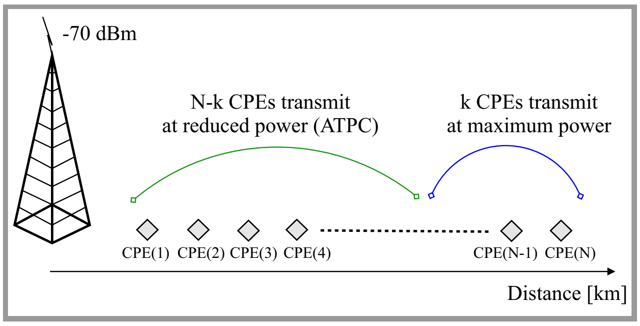

Fig. 9. ATPC – Auto Transmit Power Control.

Received Signal Strength Indication (RSSI) is the measurement of the received power present in a radio signal. These measurements were collected from all CPEs in order to evaluate their performance. Both downlink RSSI measurement (DL RSSI) and uplink RSSI measurement (UL RSSI) can be expressed by the following simple formulas:

$$\begin{aligned} RSSI_{DL} &= TX_{AP} + AntennaG_{AP} - Pathloss + AntennaG_{CPE} \\ RSSI_{UL} &= TX_{CPE} + AntennaG_{CPE} - Pathloss + AntennaG_{AP} \end{aligned} \quad (1)$$

Assuming that the pathloss component in both directions is the same, the difference can be calculated as:

$$RSSI_{DL} - RSSI_{UL} = TX_{AP} - TX_{CPE}\,. \quad (2)$$

When the AP transmits at its maximum power (+25 dBm) and the CPE is transmitting at its maximum power (+27 dBm), we obtain:

$$RSSI_{DL} - RSSI_{UL} = 25\text{ dBm} - 27\text{ dBm} = -2\text{ dB}\,. \quad (3)$$

However, since the CPE TX power is controlled by ATPC, then its TX power is $\leq +25$ dBm, therefore:

$$RSSI_{DL} - RSSI_{UL} <= -2\text{ dB}\,. \quad (4)$$

At the time of the implementation the above formula would hold true for all our measurements, however this turned out not to be true. This is shown in Fig. 10.

Figures 10 and 11 present the measurements collected from all 6000 CPEs during 5 days long experiment. The location of each APs and CPEs was fixed. Therefore, these figures present variance of propagation conditions observed by all 6000 CPEs which have occured during the experiment.

When CPEs are close to the AP, they reduce their output power to meet the receive target level of –70 dBm set by the AP. In such a close proximity Formula 4 is satisfied. However, when the CPEs are far away from the AP, the results of our measurements do no satisfy Form. 4. This happens

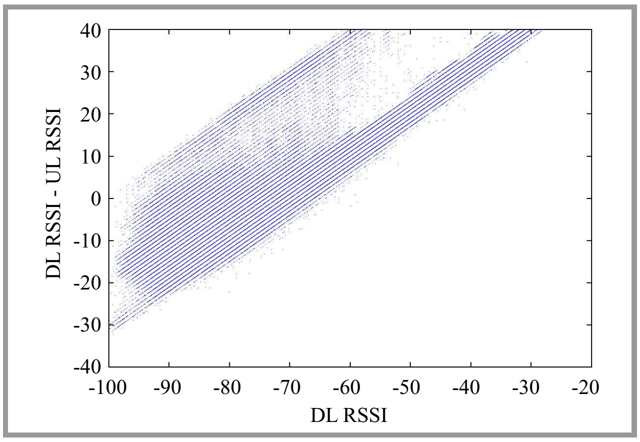

Fig. 10. DL RSSI vs. DL RSSI – UL RSSI.

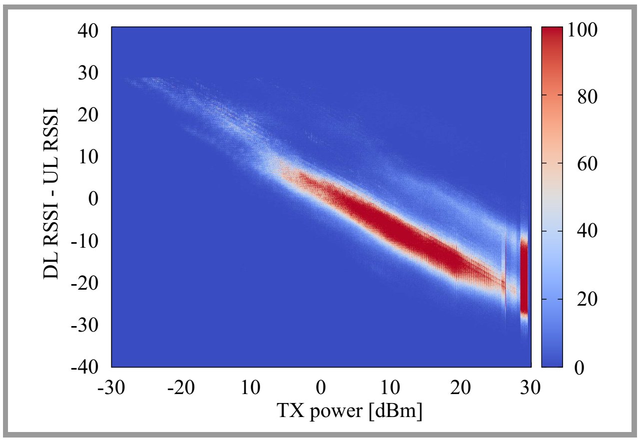

Fig. 11. CPE TX power vs. DL RSSI – UL RSSI.

due to the so-called sub-channelization gain. When CPEs is far away from the AP and has only a small amount of data to send, the power density of its transmission is increased by the use of only a subset of available sub-channels. This allows it to meet the target receive level set by AP but it only happens if CPE has a small amount of data to be transmitted. If that CPE has a large amount of data, then it will use more sub-channels, and hence it will lower the power density. When this is the case, Form. 4 is not valid. Only properly planned and restricted installation conditions allow the operator to fully exploit the MIMO and OFDMA technologies to provide maximum throughput in a multi-path environment. Chaotic roll-outs result in a network with an unpredictable capacity level.

The sub-channelization gain is a helpful mechanism, but when it occurs it also suggests that the CPE has troubles with transmission that would satisfy target receive level.

6. Summary

Over the last three years INEA has made an enormous effort to provide reliable broadband service over a wireless connection. Within this time, INEA wanted to provide the reliability associated with wire-line services with the cost advantage of the wireless technology. In the paper, the technical lessons learned from the WiMAX roll-out are presented.

One of the most important learned lessons is that, despite existing theoretical propagation models that are well described in literature, we need to tune them carefully in order to match real world measurements. In addition, the theoretical coverage is only an approximation of real coverage and thus includes internal uncertainty. The radio capacity is probably the most tricky element that the operator needs to maximize. In order to do so, the Quality of Service features provided by the 802.16e standard (in particular the best effort and ertPS flows) have been utilized, but also we have restricted the installation conditions. Only properly planned and restricted installation scenario allows to exploit the MIMO and OFDMA technology to provide successfully the maximum throughput in multi-path environment.

Lastly, the authors have also found out that some measurements (described in Section 5) may at first glance look invalid. However, they may be explained with the help of a professional support offered by the vendor.

Although operators need to learn how to deal with problems specific to WiMAX and wireless medium, we observe that it allows to reduce installation time and provides great installation flexibility as compared to wire-lines technologies. Therefore, it will continue to co-exist in presented network with GPON, DOCSIS, VDSL networks for a number of years to come.

Acknowledgements

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Paper

DVB-T Channels Measurements for the Deployment of Outdoor REM Databases

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Abstract—In this paper the analysis of the spectrum occupancy in the TV band is provided based on the outdoor measurements campaigns carried out in Poznań, Poland in 2013 and 2014. The goal of this work is to discuss the stability and other important features of the observed spectrum occupancy in the context of outdoor Radio Environment Maps database deployment. Reliable deployment of these databases seems to be one of the critical points in practical utilization of the TV White Spaces for cognitive purposes inside buildings and in densely populated cities. The results obtained for outdoor scenario are briefly compared with the previous measurements conducted indoors in Barcelona, Spain, and in Poznań, Poland.

Keywords—channel measurements, DVB-T, spectrum occupancy, Radio Environment Maps.

1. Introduction

The problem of high spectrum underutilization has attracted the researchers, network operators and various governmental bodies all over the world. Numerous measurement campaigns have been performed in many places on all continents proving that around 20–30% of spectrum in the frequency band up to 3 GHz is actively used for data transmission [1]–[6]. Clearly, this average value will vary depending on the exact frequency subband, location, date and time of the day, yet even in the densely populated cities and during the rush hours the maximum occupancy of the frequencies below 3 GHz did not exceed the tens percents.

Such a situation has motivated researchers to put significant effort on finding the way for novel techniques targeting better spectrum utilization. In consequence, the concept of cognitive radio and dynamic spectrum access appeared to be an effective solution to the aforementioned problems. Indeed, sophisticated cognitive-radio-oriented algorithms developed for the mobile terminals and for the base stations (or the whole network) are widely treated as the technical enabler of better spectrum utilization. After around fifteen years of investigation in that area, there are still various aspects that block the practical application of cognitive radio techniques in real life. However, it is worth noticing that from these investigations several lessons have been learnt [7]–[9]. One of them is the observation that the delivery of wideband wireless Internet on wide areas using dynamic spectrum access is usually very hard. Thus, it is also said that the cognitive technologies (or more specifically white space transmission) can be considered but rather locally and mainly with the use of small-cell devices. In such an approach, low-power and small-range transmitters (such as femto- or picocells) are deployed inside or outside buildings in order to improve data rate and achieving higher spectrum utilization (please see various white papers available at e.g. [10]). One can observe that the deployment of such small base stations or access points would require detailed and reliable assessment of the spectrum occupancy at the considered location.

Thus, in this work the authors concentrate on the analysis of the measurement results obtained during the campaigns performed in 2013 and 2014 in two European cities, i.e. Poznań in Poland and Barcelona in Spain, particularly focusing on the drive-tests conducted in Poland. Some of the results have been already presented in the prior work of the authors [11], [12], and particularly in [13], as this work is an extension of it. In this paper, previous observations have been extended, mainly on the measurements obtained during the drive tests. However, selected new indoor measurement results will be also presented. The indoor and outdoor measurements have been done focusing mainly on the TV band, which has been selected for many reasons.

First, it offers relatively low transmit power due to better wall penetration characteristics compared to other higher frequencies. Moreover, TV band occupancy seems to be rather stable in the sense that the positions and transmit power of the digital terrestrial television (DTT) towers, as well as the TV channel allocation maps for a given country are fixed and do not change in time. One has to also remember that this band can be also used by Program Making and Special Events systems, and that the signals generated by these devices are of relatively narrow bandwidth (around 200 kHz). In consequence, it can be assumed that the occupancy of that spectrum fragment will be rather stable and not change rapidly in time, thus it is predictable and can be utilized by white space devices. The aspect that has to be considered is the decision on the way, how the information of the current spectrum occupancy can be obtained. Due to the unreliability of the currently existing spectrum sensing algorithms, the implementation of the local databases in form of Radio Environmental Maps (REM), appears to be an attractive solution [14]–[18]. The goal of this paper is to present the conclusions that can be drawn based on a detailed analysis of the performed measurements and their implications in the context of a REM-based system implementation.

The rest of the paper is organized as follows. First, the measurement setups used in Poznan University of Technology (PUT) and in Universitat Politècnica de Catalunya (UPC) are presented. In Section 3 first the results obtained during the drive tests are analyzed, and then the indoor and outdoor measurements collected at the campuses in Poznań and Barcelona are compared. The whole work is concluded in Section 4.

2. System Setup

The two street measurement campaigns, described in this paper, have been conducted in Poznań, Poland, whereas the indoor measurements (used in this work for comparison purposes) have been also carried out in Barcelona, Spain. The measurement devices have been setup in two configurations.

2.1. The Indoor/Outdoor Measurements

In the case of indoor/outdoor measurements in both cities the DVB-T signal was captured by an omnidirectional antenna and transferred to a spectrum analyzer, which transfers it to a spectrum analyzer for initial data processing. Then the measured samples have been stored on the portable computer with the use of appropriate Matlab toolbox. In case of Poznań measurements active quad antenna, covering 40–850 MHz (1–69 TV channel), was connected via coaxial cable Lexton 3C2V of length 3 m to the R&S FLS6 spectrum analyzer. In Barcelona scenario a passive discone antenna of type AOR DN753 was used, covering the frequency range from 75 to 3000 MHz, and connected to Anritsu MS2721B device. In both setups the resolution and video bandwidth of the spectrum analyzers were the same and equal to RBW = 30 kHz and VBW = 100 kHz, respectively.

2.2. The Drive-Tests

In the case of street measurements the omnidirectional discone antenna AOR DA753 has been attached to the rooftop of a car. The aerial was connected to Rohde&Schwarz FSL v6 spectrum analyzer via low loss H155 cable. The spectrum analyzer was previously equipped with a card allowing for powering it from direct current (DC) source, i.e. lighter socket. The spectrum analyzer was connected (as in indoor setup) via Ethernet cable to a laptop that runs Matlab with Instrumental Control Toolbox installed. Additionally, GPS receiver was placed on the top of the car and connected via USB cable to the laptop. It allowed us obtaining, for each measured frequency point, the exact geographical location of the measurement. As in previous measurements, RBW and VBW were set to 30 kHz and 100 kHz, respectively. The photography of the car used for measurements is shown in Fig. 1.

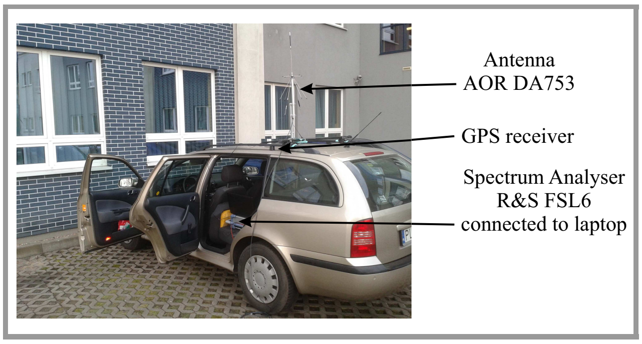

Fig. 1. Car used for street measurements.

The measurements paths were made around Poznań city center in normal traffic conditions during daytime in October 2013. The first path recorded via GPS receiver is presented in Fig. 2, where the distances are highlighted. It took about 1 hour to travel with the total length of more than 8 km. The starting point (0,0) had GPS coordinates 52° 23' 14.661" N 16° 55' 24.795" E. In order to illustrate the changes in the surrounding environmental conditions the route has been also projected on the Google Maps in Fig. 2. One can observe changing scenarios – from loosely populated areas (marked on the map as City center – Residential area), through the city center with high tenement houses (Old market square), finishing on the ducts over the big river (Warta river) and sport areas (Malta lake). In this figure also the preview positions of the closest DVB-T towers are shown, highlighting also the distance from the PUT campus. Finally, let us stress that the measurements have been done in the typical daily traffic conditions, i.e. depending on that traffic and on the switch-on turn-on phases of the lamps at the crossroads, traffic lights, in some places the number of collected samples will be, e.g., higher than in other places due to the travel speed.

2.3. Test Campaign

As the first drive-test measurement campaign has been realized in autumn 2013, the second one was realized in analogous conditions (i.e., normal traffic conditions during daytime) but late winter, precisely in February 2014. The second road, illustrated in Fig. 3, was longer (around 25 km) and started just before the premises of the Faculty of Electronics and Telecommunications (52° 24' 0.66" N, 16° 57' 20.51" E) and finished in the city center. One can observe that in that case the route led around the strict city center, providing different observations as compared to the

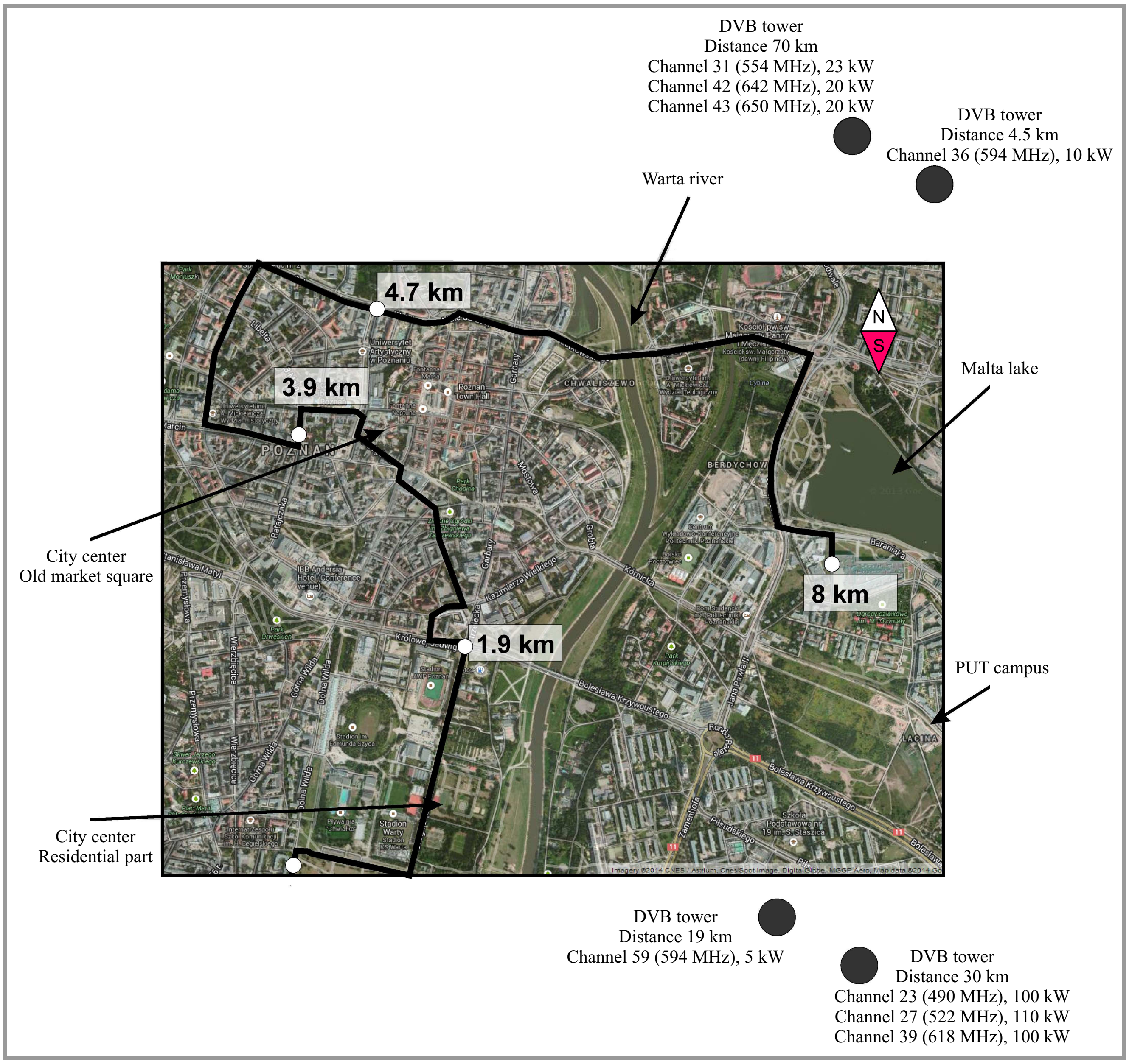

Fig. 2. The first route of car projected on the Google Map of Poznań.

results achieved in the first case. Moreover, one can notice the presence of the new DVB tower (called "Piatkowo"), which has been launched just between the two conducted measurement campaigns, probably in order to strengthen the received power at this channel. The perspective illustration of the location of the five DVB towers visible in the Poznań city center is presented in Fig. 4. It is also worth noticing that in Fig. 3 the position of two specific places has been marked (denoted as Place A and Place B), where the car used in the measurements stayed for longer time, i.e., it was intentionally stopped allowing for stable power measurements in one specific point as a function of time. Finally, for the sake of clarity the first route (Route 1) has been also highlighted in Fig. 3 in form of white dashed line.

3. Measurement Results

3.1. Drive Tests

First, let's analyze the achieved results from the drive test, which are presented in form of the received power in the selected DVB-T channels (23, 27, 36, 39, see Table 1) expressed in dBm per 8 MHz as the function of the distance from the start of the route (Fig. 5 for first route and Fig. 6 for the second route) and as the function of time from the measurement beginning (Fig. 7 for first route and Fig. 8 for the second route).

Let's stress that these channels have been selected to be observed as they are the only channels occupied by the DVB-T signal at Poznań. One can observe quite high variations

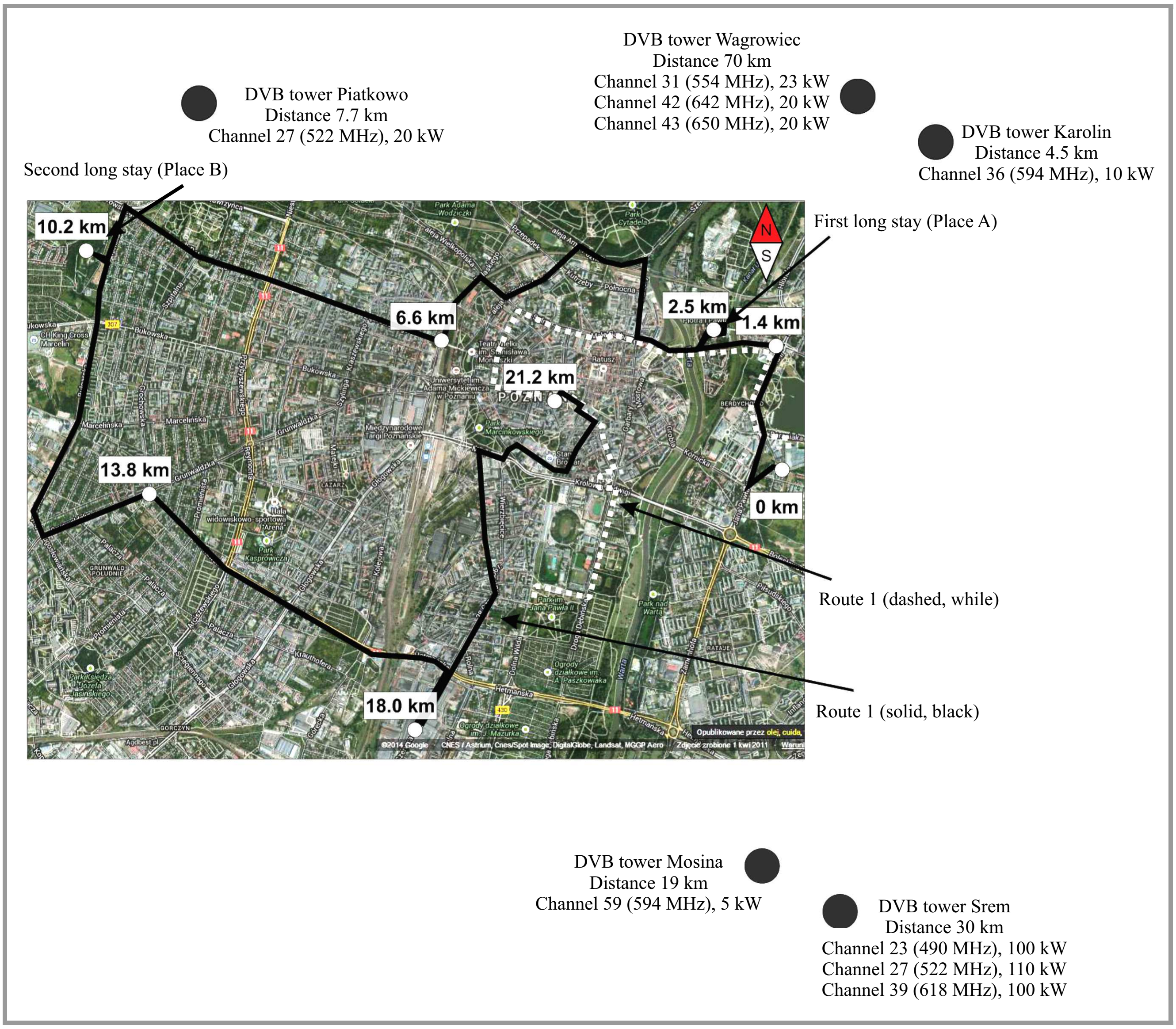

Fig. 3. The second route of car projected on the Google Map of Poznań.

of the received signal power depending on the current position. For example, the black curve (the one at the top of Figs. 5 and 6) represents the changes in the received signal power from the closest DVB-T tower. One can see the correlation between the types of scenario, in which the measurement have been performed (residential area or the surroundings of artificial Malta lake), and the

Table 1
List of scanned TV channels

TV channel	Frequencies [MHz]	Carrier frequency [MHz]
23	486–494	490
27	518–526	522
36	590–598	594
39	614–622	618

value of the received power. The highest values of the received power have been observed near the river and Malta lake (see the Route 1). Surprisingly, the variations of the received signal power in other TV channels were much smaller. In terms of numbers, the standard deviation of the received power in channel 36 was equal to around 7 dB (Route 1) or 6 dB (Route 2), while the variation of the received signal in the other channels was close to 3 dB (Route 1), and 4 dB (Route 2). Moreover, detailed analysis proved that the local, spatial variations of the signal power (observed in the TV channel shown in Fig. 5 to Fig. 6, but also on the other 8 MHz bands) are not so rapid, since no or very limited number of high spikes have been observed. If this is a case, there should exist a practical possibility for deployment of low power White Space Base Stations that will operate in the TV White Spaces with the support of the local Radio Environment Map.

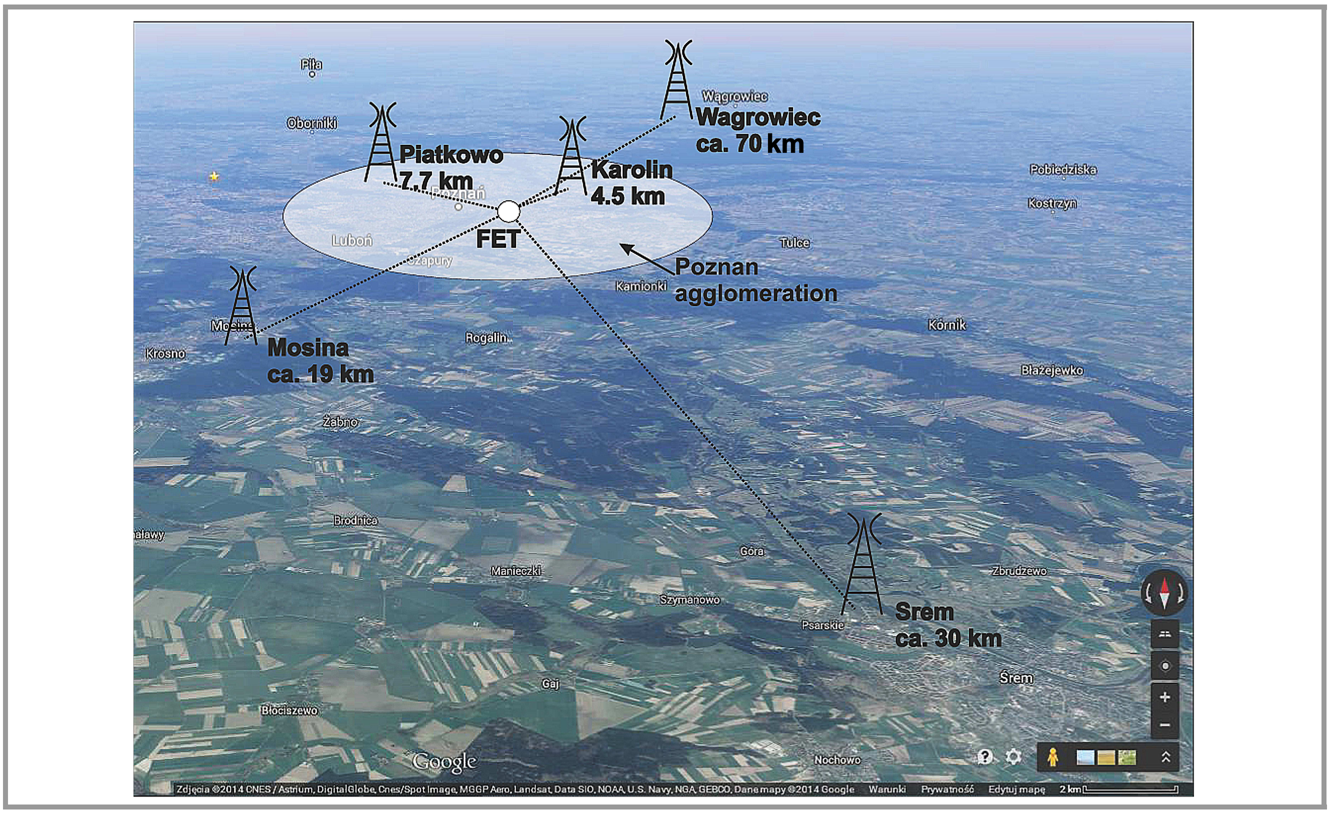

Fig. 4. Location of the DVB towers – perspective (FET – stands for the Faculty of Telecommunications).

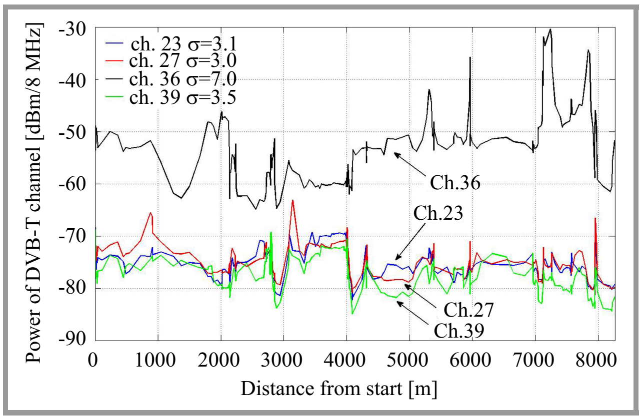

Fig. 5. Received signal power as the function of the distance from the route start – Route 1.

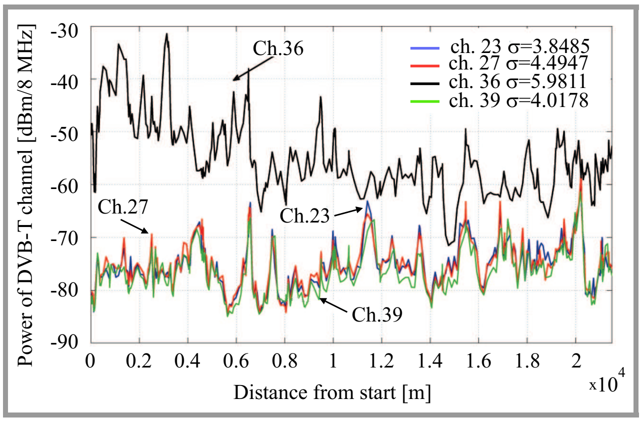

Fig. 6. Received signal power as the function of the distance from the route start – Route 2.

Analogously, similar conclusions can be drawn from the next figure (Figs. 7 and 8), where the signal power is presented as a function of time. The former figure coincides with Fig. 5, however there is one significant difference – for around 400 s (around 6 minutes) the measurements have been performed in one place (the parking place near the castle located in the centre of Poznań). This period can be observed in Fig. 7 in the range of the 1100 to 1500 s. The number of moving objects (including cars, people and animals) in this area is rather big. Nevertheless, the received signal power in all TV channels seems to be stable. It suggests that the time variations of the signal are small and the influence of the moving elements in surrounding environment on the received power is also limited. Such a conclusion is very important since it builds the fundamentals for the deployment of low power base stations that will operate in free TV channels. In order to prove this observation two additional places for long-time measurements during the drive tests have been selected, i.e., Place A (near the Poznań cathedral, close to the city center) and Place B (near military fort – Fort VII, a place where there are no high buildings in the closest vicinity). The measurements from that two specific places have been marked with the two ellipses in Fig. 8. One can observe

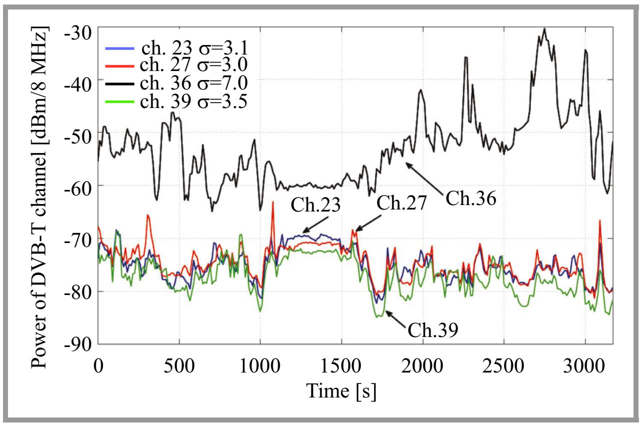

Fig. 7. Received signal power as the function of the time passed from the route start – Route 1.

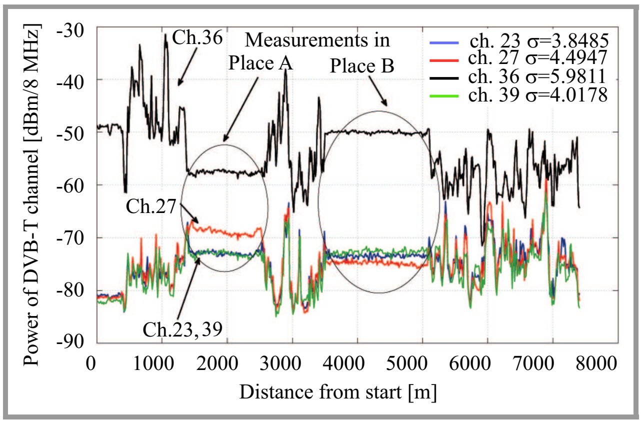

Fig. 8. Received signal power as the function of the time passed from the route start – Route 2.

that the channel variations in these points can be stated as negligible.

It is also worth noticing that very similar behavior of the received signal power is observed in all other TV channels (23, 27 and 39) transmitted from the same TV tower. Although the distance between the TV channels in frequency domain is even up to 120 MHz, the received signal powers as the function of time and distance are highly correlated. All of the above mentioned observations and conclusions are of high importance since the measurements have been done in very typical day-time traffic conditions.

Finally, let's analyze the averaged Power Spectral Density of the received signal (Fig. 9 for the Route 1 and Fig. 10 for the Route 2) and the box plot corresponding to the signal power calculated within the considered TV channels (Fig. 11 for Route 1 and Fig. 12 for Route 2). In these figures the presence of four high-power DVB-T signals are observable at channels 23, 27, 36 and 39. The former figure illustrates three curves: the received power averaged during the whole route (middle plot), minimum observable power during the whole route (bottom plot), and maximum observed power during the whole measurement time (upper plot). Since the raster on the horizontal axes is 30 kHz, one can observe that relatively high number of narrowband

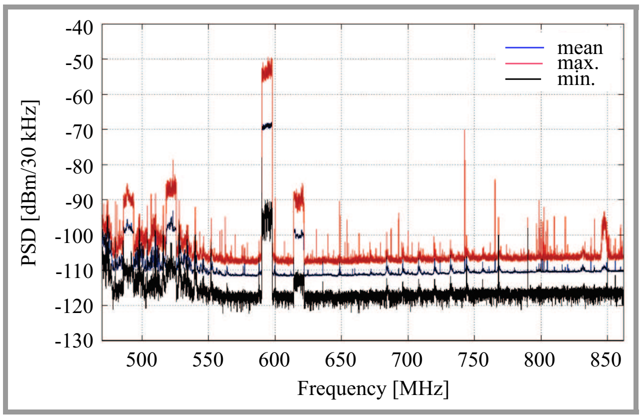

Fig. 9. Power Spectral Density Function of the received signal samples during the whole drive test – Route 1.

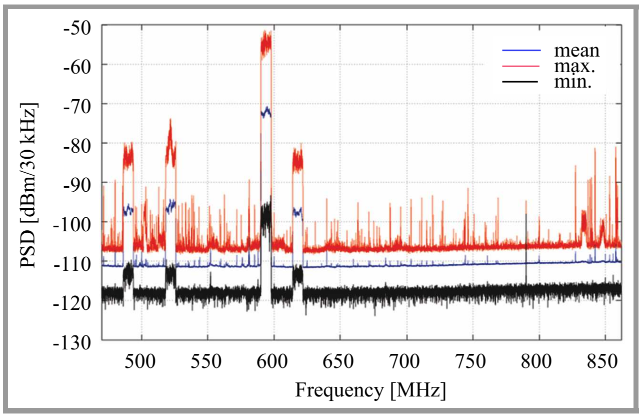

Fig. 10. Power Spectral Density Function of the received signal samples during the whole drive test – Route 2.

peaks are observable in various locations. It is also worth noticing that the two lower channels 23 and 27 in some locations are not detectable or at least severely degraded, since the received power in that band is close to the ambient noise power.

Valuable observations can be drawn from the analysis of the Fig. 11, where the box plot is shown for the frequency raster set to 8 MHz. The signal power received in channel 36 is rather very high regardless of the high values of variance. However, one can see that for the other channels the decision about the presence or absence of DVB-T signal is not so straight-forward. For example the mean measured power in channel 21 is very similar to that from channel 23, while in the former one the channel should be stated as vacant. Furthermore, although the mean value of the received power in channel 39 is less than in other occupied channels, the decision of the presence of the signal is rather easy due to the low ambient noise power observed in the adjacent frequency bands. Hopefully, local information about the received signal strength in the TV channels seems to be stable and weakly dependent on the surrounding environmental conditions. Analogous conclusions can be drawn from the analysis of the box-plot related to the second route, i.e., Fig. 12. However, some differences can

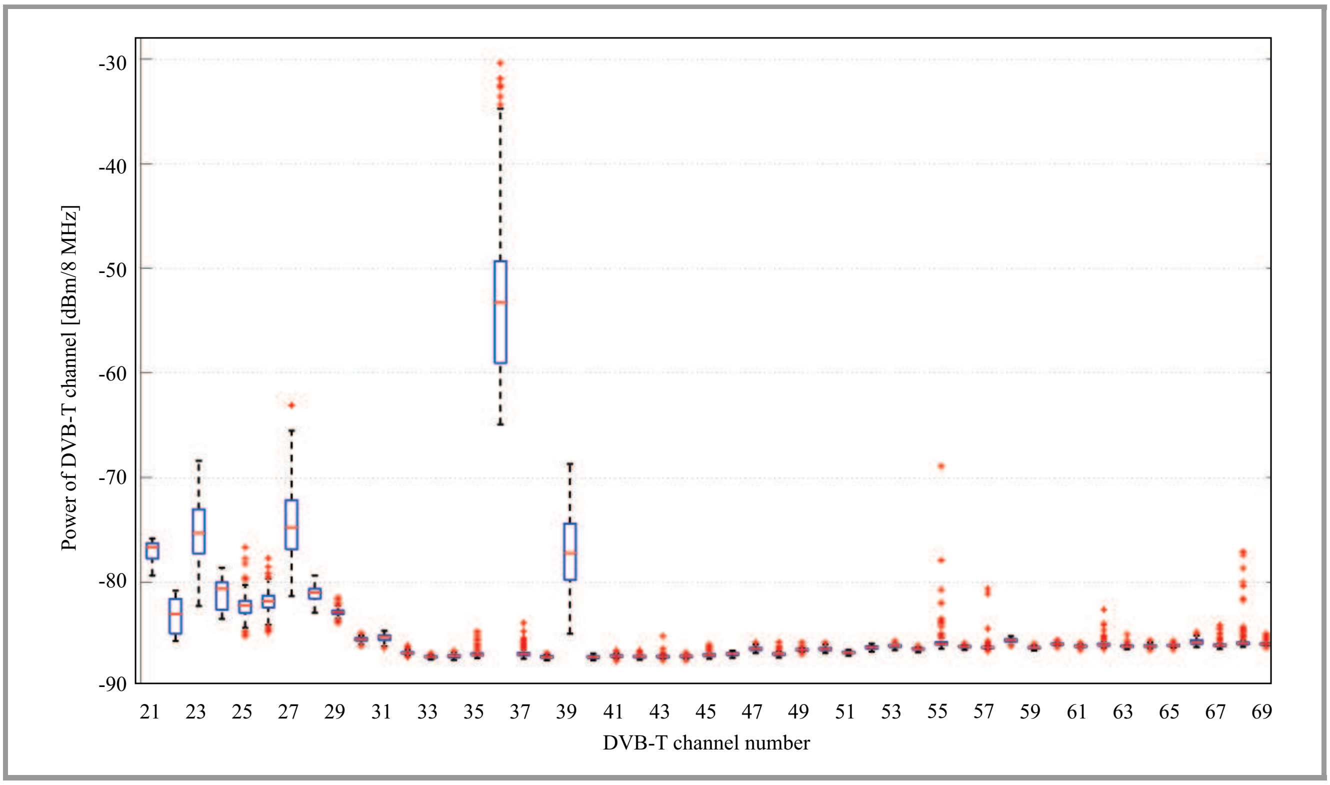

Fig. 11. Box-plot of the received signal power along the whole route – Route 1.

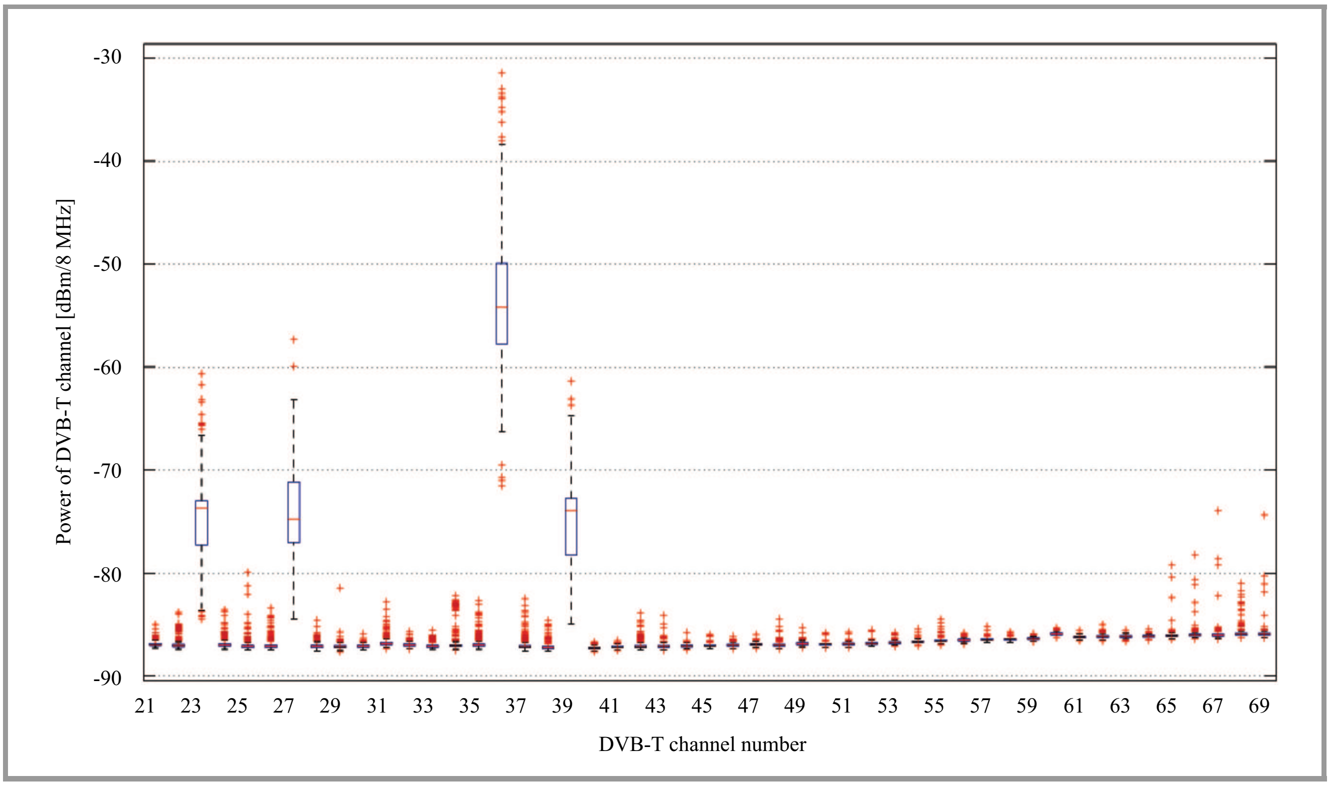

Fig. 12. Box-plot of the received signal power along the whole route – Route 2.

be noticed between these two plots. First, the variance of the received power within channel 27 is higher for the second route. This is due to the fact that the new DTT tower has been launched between the first and second measurement campaigns. On the other hand, the variance observed at channel 39 is also higher for the second route, but this time the reason could be just the fact that in the two measurement campaigns different routes have been used. Finally, there is relatively high power observed at the lower frequencies for the first route (channels up to 31), which is not observed in the second case. It is not straight-forward to decide on the reason for that phenomenon. For sure it is not caused by different system setup, since the same measurement hardware and car have been used in both campaigns. This power in lower TV channels can be some sort of interference coming from "unknown sources". Similarly, some differences can be observed at higher TV channels.

3.2. Indoor-Outdoor Measurements

Detailed analysis of the performed measurements inside buildings of Poznań and Barcelona has been done in [11], [12], and in further discussion will be based on the conclusions presented therein. In a nutshell, the main clue that originated from the measurements discussed in the referred paper is the following:

- the received signal power inside the building is rather stable in time, although it varies depending on the location,
- the influence of the people walking inside the room on the observed signal power is strongly limited,
- the attenuation of the walls is very strong, such that in many cases the reception of the DVB-T signal was not possible and the usage of external (e.g. roof-top) aerial was required,
- a local REM data base seems a feasible option to characterize the spectrum occupation in different positions inside the building.

However, in this paper the authors would like to focus rather on the comparison between the street measurements (drive-tests) and the selected results observed by both stable (non-moving) roof-antennas and the antennas located inside buildings in the UPC campus. Analyzing the conclusions presented above in this subsection with the discussion done in the previous sections one can observe high similarity. It can be stated that both indoor and outdoor measurements prove the high local stability of the received signal power and weak influence of the moving objects on the received signal power. It further means that the deployment of the local white space REMs should be tractable. Now, let's compare the values of the received power in Poznań and in Barcelona and try to generalize the results. Focusing on the results presented in Figs. 3 to 5 it can be stated that the received power in the city center varies from

Table 2
Received signal power observed at channels 26 and 61 expressed in [dBm/8 MHz] – in Barcelona

	Ch. 26	Ch. 61
Outdoor reference	–51,27	–52,18

Table 3
Received signal power at channel 26

Ch. 26	A	B	C	Avg.	Att.
Basement	–68,19	–70,41	–74,39	–69,25	17,98
Ground floor	–69,75	–60,39	–70,77	–63,49	12,22
1st floor	–56,53	–54,34	–65,85	–57,85	6,58
2nd floor	–55,72	–56,02	–61,60	–57,60	6,33

Table 4
Received signal power at channel 61

Ch. 61	A	B	C	Avg.	Att.
Basement	–79,28	–82,47	–82,35	–80,69	28,51
Ground floor	–68,43	–66,01	–73,91	–68,47	16,29
1st floor	–62,26	–60,60	–72,96	–64,08	11,90
2nd floor	–67,79	– 57,89	–73,56	–63,18	11,00

–50 dBm/8 MHz down to –80 dBm/8 MHz. Let's now compare these values with the results obtained in Barcelona using stable (non-moving) measurement equipment. In Table 2 the referenced signal power in two TV channels – 26 and 61 – measured outside the building (at the rooftop) is shown. Then, in Tables 3 and 4, received signal power in the same TV channels is presented but for four other indoor measurements locations. One can see that the received power outside the building is around –51 dBm, what is also around 20 dB higher when comparing with the indoor measurements of the same TV channels signals. Analogous results have been obtained for other buildings and other locations, as presented in [11] and [12]. Clearly, the direct comparison of these two measurement scenarios is not fair, however, one can easily conclude that based on these results it can be stated that in most cases the outdoor signals are high enough to be easily detected, however, inside the buildings or in, e.g., "street valley" the received signal power can be so low that it would be impossible to detect the TV transmission. In other words, the measurements have proved that although the REM can cover wide areas, the granularity of its entries (records) shall be rather high. Moreover, when buildings are considered, the granularity of the REM should include the 3D dimensions since the floor inside the building plays a key role on the received power, as seen in Tables 3 and 4.

4. Conclusions

In this paper the comparison of the measurement results of the two street drive-tests has been presented and compared with the indoor measurements done in two European cities, focusing on the similarities and differences that occur between indoor measurements and drive tests. It has been stated that hopefully in both scenarios the stability of the TV channels is very high, and the influence of the surrounding moving objects is rather limited. Such an observation is crucial, since it is the basis for further work on the deployment of local outdoor and indoor radio environment maps for TV White Space Communications.

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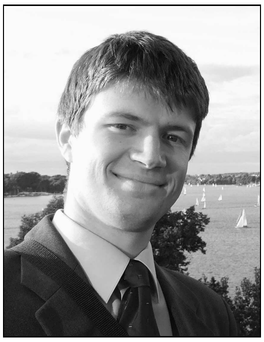

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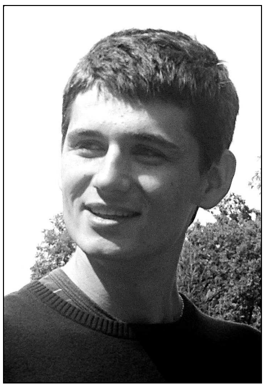

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Paper

Adaptive Algorithms Versus Higher Order Cumulants for Identification and Equalization of MC-CDMA

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Abstract—In this paper, a comparative study between a blind algorithm, based on higher order cumulants, and adaptive algorithms, i.e. Recursive Least Squares (RLS) and Least Mean Squares (LMS) for MultiCarrier Code Division Multiple Access (MC-CDMA) systems equalization is presented. Two practical frequency-selective fading channels, called Broadband Radio Access Network (BRAN A, BRAN B) normalized for MC-CDMA systems are considered. In the part of MC-CDMA equalization, the Zero Forcing (ZF) and the Minimum Mean Square Error (MMSE) equalizer techniques were used. The simulation results in noisy environment and for different signal to noise ratio (SNR) demonstrate that the blind algorithm gives approximately the same results obtained by adaptive algorithms. However, the proposed algorithm presents the advantage to estimate the impulse response of these channels blindly except that the input excitation is non-Gaussian, with the low calculation cost, compared with the adaptive algorithms exploiting the information of input and output for the impulse response channel estimation.

Keywords*—*blind identification and equalization, higher order cumulants, RLS, LMS, MC-CDMA systems.

1. Introduction

Many algorithms have been proposed in the literature for the identification of Finite Impulse Response (FIR) system using cumulants, established that blind identification of FIR Single-Input Single-Output (SISO) communication channels is possible only from the output second order statistics of the observed sequences (Auto Correlation Function and power spectrum) [1]. Moreover, the system to be identified has no minimum phase and is contaminated by a Gaussian noise, where the Auto Correlation Function (ACF) does not allow identifying the system correctly because the cumulants vanishes on order greater than 2 [2]–[4]. To overcome these problems, other approaches was proposed by several authors in [5]–[14]. This paper is focused on channels impulse response estimation with non-minimum phase and selective frequency such as: BRAN A and BRAN B normalized for MC-CDMA systems. In MC-CDMA a single data symbol is transmitted at multiple narrow band subcarriers [15], [16]. Indeed, in MC-CDMA systems, spreading codes are applied in the frequency domain and transmitted over independent subcarriers. In most wireless environments, there are many obstacles in the communication, such as buildings, mountains, and walls between the transmitter and the receiver antennas. The reflections from these obstacles cause many propagation paths. The problem met in communication is the synchronization between the transmitter and the receiver, due to the echoes and reflection between the transmitter and the receiver antennas. Synchronization errors cause loss of orthogonality among sub-carriers and considerably degrade the performance especially when large number of subcarriers is present [17].

This paper describes a blind algorithm which is based only on third order cumulants. In order to test its efficiency, it was compared with the adaptive algorithms such as Recursive Least Square (RLS) and Least Mean Square (LMS) [18], [19]. Two practical frequency-selective fading channels called Broadband Radio Access Network (BRAN A, BRAN B), normalized by the European Telecommunications Standards Institute (ETSI) were considered [20], [21]. In this paper, a novel concept of blind equalization is developed and investigated for downlink MC-CDMA systems. Moreover, the developed method is compared with the adaptive equalization obtained using RLS and LMS algorithms. The bit error rate (BER) performance of the downlink MC-CDMA systems, using blind BRAN A and BRAN B estimation, are shown and compared with the results obtained with the adaptive methods.

2. Problem Statement

The output of a FIR system that is excited by an unobservable input and is corrupted on output by an additive white Gaussian noise (Fig. 1) is described by the following formula

$$y(k) = \sum_{i=0}^{q} x(i)h(k-i). \tag{1}$$

The observed measurable output $r(k)$ is given by

$$r(k) = y(k) + n(k), \tag{2}$$

where $n(k)$ is the noise sequence.

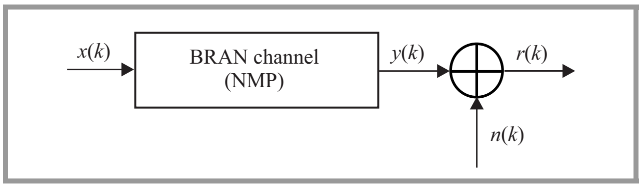

Fig. 1. Channel model.

In order to simplify the algorithm construction, it was assumed that the input sequence $x(k)$ is independent and identically distributed (i.i.d.) zero mean and non-Gaussian. The system is causal and truncated, i.e. $h(k) = 0$ for $k < 0$ and $k > q$, where $h(0) = 1$. The system order q is known. The noise sequence $n(k)$ is i.i.d., Gaussian, independent of $x(k)$ and with unknown variance.

3. Blind Algorithm (Algo-CUM)

The m^{th} order cumulants of $y(k)$ can be expressed as follows [6], [18], [19]:

$$C_{my}(t_1, t_2, \ldots, t_{m-1}) = \xi_{mx} \sum_{i=0}^{q} h(i)h(i+t_1)\ldots h(i+t_{m-1}), \tag{3}$$

where ξ_{mx} is the m^{th} order cumulants of the excitation signal $x(i)$ at origin.
For $m = 2$ in Eq. (3), the second order cumulants is obtained as follows:

$$C_{2y}(t) = \xi_{2x} \sum_{i=0}^{q} h(i)h(i+t). \tag{4}$$

Analogically, if $m = 3$, Eq. (3) yields to

$$C_{3y}(t_1, t_2) = \xi_{3x} \sum_{i=0}^{q} h(i)h(i+t_1)h(i+t_2). \tag{5}$$

The second-order cumulants Z-transform is straightforward and gives Eq. (6)

$$S_{2y}(z) = \xi_{2x} H(z)H(z^{-1}). \tag{6}$$

The Z-transform of Eq. (5) is Eq. (7)

$$S_{3y}(z_1, z_2) = \xi_{3x} H(z_1)H(z_2)H(z_1^{-1})H(z_2^{-1}). \tag{7}$$

If $z = z_1 z_2$, Eq. (6) becomes

$$S_{2y}(z_1 z_2) = \xi_{2x} H(z_1 z_2)H(z_1^{-1} z_2^{-1}). \tag{8}$$

Then, from Eqs. (7) and (8), the following formula is obtained:

$$S_{3y}(z_1, z_2)H(z_1 z_2) = \mu H(z_1)H(z_2)S_{2y}(z_1 z_2), \tag{9}$$

where $\mu = \frac{\xi_{3x}}{\xi_{2x}}$.
The inverse Z-transform of Eq. (9) demonstrates that the 3rd order cumulants, the ACF and the impulse response channel parameters are combined by:

$$\sum_{i=0}^{q} C_{3y}(t_1 - i, t_2 - i)h(i) = \mu \sum_{i=0}^{q} h(i)h(t_2 - t_1 + i)C_{2y}(t_1 - i). \tag{10}$$

Using the ACF property of the stationary process such as $C_{2y}(t) \neq 0$ only for $-q \leq t \leq q$ and vanishes elsewhere if $t_1 = -q$, the Eq. (10) becomes:

$$\sum_{i=0}^{q} C_{3y}(-q - i, t_2 - i)h(i) = \mu h(0)h(t_2 + q)C_{2y}(-q). \tag{11}$$

Using the property of the cumulants, $C_{3y}(t_1, t_2) = C_{3y}(-t_1, t_2 - t_1)$, Eq. (11) is:

$$\sum_{i=0}^{q} C_{3y}(q + i, t_2 + q)h(i) = \mu h(0)h(t_2 + q)C_{2y}(q). \tag{12}$$

The considered system is causal. Therefore, the interval of the t_2 is $t_2 = -q, \ldots 0$. Otherwise, if $t_2 = -q$, Eq. (12) will takes the following form:

$$C_{3y}(q, 0)h(q) = \mu h(0)C_{2y}(q). \tag{13}$$

Thus, based on Eq. (13) and eliminating $C_{2y}(q)$ from Eq. (12), the equation constituted of only the third order cumulants is obtained:

$$\sum_{i=0}^{q} C_{3y}(q + i, t_2 + q)h(i) = C_{3y}(q, 0)h(t_2 + q). \tag{14}$$

The system of Eq. (14) in matrix form is as follows

$$\begin{pmatrix} C_{3y}(q+1, 0) & \ldots & C_{3y}(2q, 0) \\ C_{3y}(q+1, 1) - \alpha & \ldots & C_{3y}(2q, 1) \\ \vdots & \vdots & \vdots \\ C_{3y}(q+1, q) & \ldots & C_{3y}(2q, q) - \alpha \end{pmatrix}$$

$$\times \begin{pmatrix} h(1) \\ \vdots \\ h(i) \\ \vdots \\ h(q) \end{pmatrix} = \begin{pmatrix} 0 \\ -C_{3y}(q, 1) \\ \vdots \\ -C_{3y}(q, q) \end{pmatrix}, \tag{15}$$

where $\alpha = C_{3y}(q, 0)$.

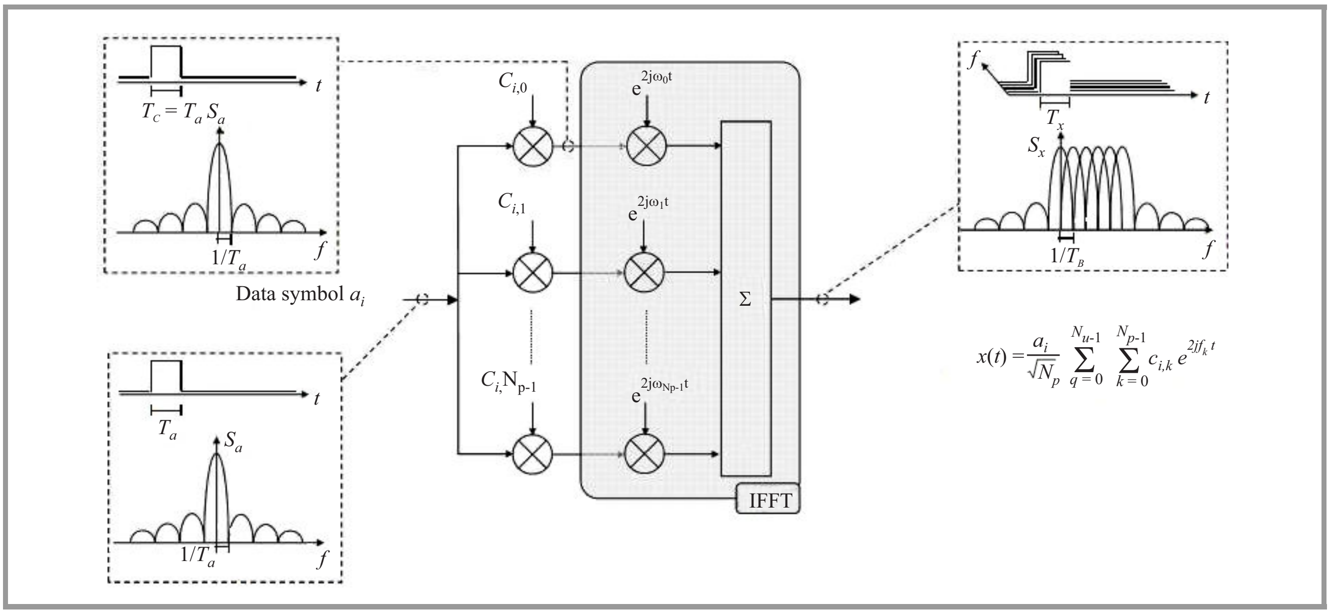

Fig. 2. MC-CDMA modulator principle.

In more compact form, the Eq. (15) is:

$$Mh_e = d, \tag{16}$$

where M is the matrix of size $(q+1)\times(q)$ elements, h_e is a column vector constituted by the unknown impulse response parameters $h(k)$: $k = 1,\ldots,q$ and d is a column vector of size $(q+1)$, as indicated in the Eq. (15).

The least squares solution of the Eq. (16) permits a blindly identification of the parameters $h(k)$. Therefore, the solution can be written as:

$$\widehat{h_e} = (M^T M)^{-1} M^T d. \tag{17}$$

4. RLS Algorithm

The RLS algorithm [19] is described by the following equations (with initialization $h(0) = 0$).

$Q^{-1}(0) = \delta^{-1} I$, δ is a small positive constant value.

$$k(n) = \frac{\lambda^{-1} Q^{-1}(n-1) X(n)}{1 + \lambda^{-1} Q^{-1}(n-1) X(n)}, \tag{18}$$

$$e(n) = r(n) - X^T h(n-1), \tag{19}$$

$$h(n) = h(n-1) - k(n)e(n), \tag{20}$$

$$Q^{-1}(n) = \lambda^{-1} Q^{-1}(n-1) - \lambda^{-1} k(n) X^T(n) Q^{-1}(n-1). \tag{21}$$

5. LMS Algorithm

The LMS algorithm [18] is described by the following equations with the initialization $h(0) = 0$, and computed for $n = 0,1,2,\ldots$

$$e(n) = r(n) - X^T h(n-1), \tag{22}$$

$$h(n) = h(n-1) + \mu e(n) X(n). \tag{23}$$

where μ is the convergence factor.

6. Equalization of MC-CDMA System

The operation principle of MC-CDMA system is described by a symbol a_i of each user i transmitted at multiple narrow band subcarriers [22], [23] (Fig. 2). Indeed, in MC-CDMA systems, spreading codes are applied in the frequency domain and transmitted over independent subcarriers.

6.1. MC-CDMA Transmitter

The symbol a_i of user i is multiplied by each chip $c_{i,k}$ of spreading code and then applied to the modulator. Each subcarrier transmits an information element multiplied by a code chip of that subcarrier. For example, the case, where the length L_c of spreading code is equal to the number N_p of subcarriers is considered. The optimum space between two adjacent subcarriers is equal to inverse of duration T_c of spreading code in order to guarantee the orthogonality between subcarriers. Thus, the MC-CDMA emitted signal is given by [6]:

$$x(t) = \frac{a_i}{\sqrt{N_p}} \sum_{q=0}^{N_u-1} \sum_{k=0}^{N_p-1} c_{i,k} e^{2jf_k t}, \tag{24}$$

where $f_k = f_0 + \frac{k}{T_c}$, N_u is the user number and N_p is the number of subcarriers. Figure 3 explains the trans-

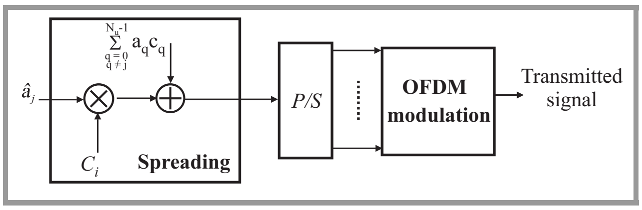

Fig. 3. MC-CDMA transmitter.

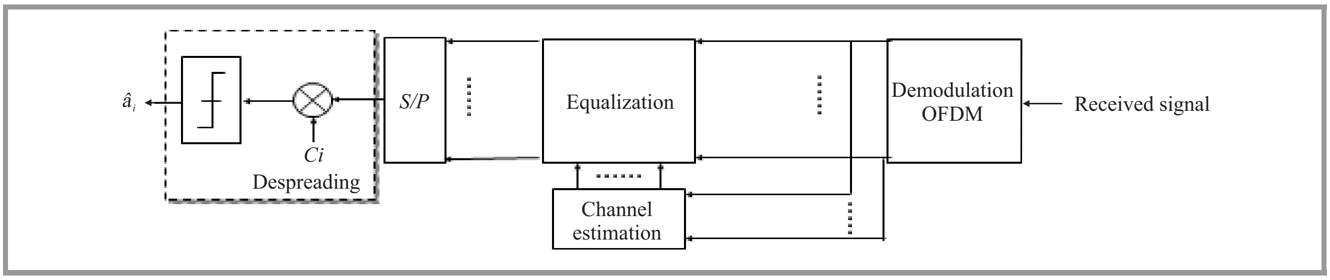

Fig. 4. MC-CDMA receiver block diagram.

mitter operation principle of the for downlink MC-CDMA systems.

Assuming that the channel is time invariant and it is impulse response is characterized by P path of magnitudes β_p and phases θ_p, the impulse response is given by:

$$h(\tau) = \sum_{p=0}^{P-1} \beta_p e^{j\theta_p} \delta(\tau - \tau_p), \tag{25}$$

$$\begin{aligned} r(t) &= \int_{-\infty}^{+\infty} \sum_{p=0}^{P-1} \beta_p e^{j\theta_p} \delta(\tau - \tau_p) x(t-\tau) d\tau + n(t) \\ &= \sum_{p=0}^{P-1} \beta_p e^{j\theta_p} x(t - \tau_p) + n(t), \end{aligned} \tag{26}$$

where $n(t)$ is an additive white Gaussian noise.

6.2. MC-CDMA Receiver

The received signal is given by the following equation [6], [23]:

$$\begin{aligned} r(t) &= \frac{1}{\sqrt{N_p}} \sum_{p=0}^{P-1} \sum_{k=0}^{N_p-1} \sum_{i=0}^{N_u-1} \times \\ &\times \Re\{\beta_p e^{j\theta} a_i c_{i,k} e^{2j\pi(f_0+k/T_c)(t-\tau_p)}\} + n(t). \end{aligned} \tag{27}$$

The main goal, is to obtain a good estimation of the symbol $\widehat{a_i}$. The first operation is the received signal demodulation, according to N_p subcarriers. The second step is the received sequence multiplication by the users code.

The receiver structure for downlink MC-CDMA systems is shown in Fig. 4. The emitted user symbol $\widehat{a_i}$ estimation is given by:

$$\begin{aligned} \widehat{a_i} &= \sum_{q=0}^{N_u-1} \sum_{k=0}^{N_p-1} c_{i,k}(g_k h_k c_{q,k} a_q + g_k n_k) \\ &= \underbrace{\sum_{k=0}^{N_p-1} c_{i,k}^2 g_k h_k a_i}_{\mathrm{I}\ \ (i=q)} + \underbrace{\sum_{q=0}^{N_u-1} \sum_{k=0}^{N_p-1} c_{i,k} c_{q,k} g_k h_k a_q}_{\mathrm{II}\ \ (i \neq q)} \\ &+ \underbrace{\sum_{k=0}^{N_p-1} c_{i,k} g_k n_k}_{\mathrm{III}}, \end{aligned} \tag{28}$$

where the part I, II and III of the formula present respectively: the desired signal (i.e. considered user signal), a multiple access interferences (i.e. others users signals) and the noise, i.e. pondered by the equalization coefficient and by chip spreading code.

6.3. Equalization for MC-CDMA

6.3.1. Zero Forcing

The zero forcing (ZF) technique operation principle is to cancel the distortions brought by the channel. The gain factor of the ZF equalizer principle is

$$g_k = \frac{1}{|h_k|}. \tag{29}$$

Therefore, the estimated received symbol $\widehat{a_i}$ of the user i is given by:

$$\widehat{a_i} = \underbrace{\sum_{k=0}^{N_p-1} c_{i,k}^2 a_i}_{I\ \ (i=q)} + \underbrace{\sum_{q=0}^{N_u-1} \sum_{k=0}^{N_p-1} c_{i,k} c_{q,k} a_q}_{II\ \ (i \neq q)} + \underbrace{\sum_{k=0}^{N_p-1} c_{i,k} \frac{1}{h_k} n_k}_{III}. \tag{30}$$

Using the orthogonality condition, i.e.

$$\sum_{k=0}^{N_p-1} c_{i,k} c_{q,k} = 0 \qquad \forall i \neq q, \tag{31}$$

Eq. (30) becomes:

$$\widehat{a_i} = \sum_{k=0}^{N_p-1} c_{i,k}^2 a_i + \sum_{k=0}^{N_p-1} c_{i,k} \frac{1}{h_k} n_k. \tag{32}$$

6.3.2. Minimum Mean Square Error

The Minimum Mean Square Error (MMSE) technique combine the multiple access interference minimalization and the signal to noise ratio maximization. The MMSE minimize the mean square error for each subcarrier k between the transmitted signal x_k and the output detection $g_k r_k$ [6]

$$E[|\varepsilon|^2] \quad = \quad E[|x_k - g_k r_k|^2]. \tag{33}$$

The $E[|\varepsilon|^2]$ function minimalization gives the optimal equalizer coefficient, under the minimalization of the mean square error criterion for each subcarrier as:

$$g_k = \frac{h_k^*}{|h_k|^2 + \frac{1}{\zeta_k}}, \tag{34}$$

where $\zeta_k = \frac{E[|r_k h_k|^2]}{E[|n_k|^2]}$.

The estimated received symbol $\widehat{a_i}$ of symbol a_i of the user i is described by:

$$\widehat{a_i} = \underbrace{\sum_{k=0}^{N_p-1} c_{i,k}^2 \frac{|h_k|^2}{|h_k|^2 + \frac{1}{\zeta_k}} a_i}_{I \quad (i=q)} + \underbrace{\sum_{q=0}^{N_u-1} \sum_{k=0}^{N_p-1} c_{i,k} c_{q,k} \frac{|h_k|^2}{|h_k|^2 + \frac{1}{\zeta_k}} a_q}_{II \quad (i \neq q)} + \underbrace{\sum_{k=0}^{N_p-1} c_{i,k} \frac{h_k^*}{|h_k|^2 + \frac{1}{\zeta_k}} n_k}_{III}. \tag{35}$$

Assuming that the spreading codes are orthogonal, the Eq. (35) becomes:

$$\widehat{a_i} = \sum_{k=0}^{N_p-1} c_{i,k}^2 \frac{|h_k|^2}{|h_k|^2 + \frac{1}{\zeta_k}} a_i + \sum_{k=0}^{N_p-1} c_{i,k} \frac{h_k^*}{|h_k|^2 + \frac{1}{\zeta_k}} n_k. \tag{36}$$

7. Simulation Results

To evaluate the proposed algorithm performance, the BRAN A and BRAN B models representing the fading radio channels are considered. Their corresponding data are measured for multicarrier code division multiple access (MC-CDMA) systems. The Eq. (37) describes the impulse response $h(k)$ of BRAN radio channel:

$$h(k) = \sum_{i=0}^{N_T} A_i \delta(k - \tau_i), \tag{37}$$

where $\delta(n)$ is Dirac delta, A_i stands for the magnitude of the targets i, $N_T = 18$ is the number of target and τ_i is the time delay (from the origin) of target i.

Although, the BRAN channels are constituted by $N_T = 18$ parameters and seeing that their value are very low, for that the following procedure is taken:

- The BRAN A channel impulse response is decomposed into four sub-channel as:

$$h(k) = \sum_{j=1}^{4} h_j(k). \tag{38}$$

- The parameters of each sub-channel are independently estimated, using the proposed algorithm.
- All sub channel parameters are added, to construct the full BRAN channels impulse response.

Table 1
Delay and magnitudes of 18 targets of BRAN A channel

Delay τ_i [ns]	Mag. A_i [dB]	Delay τ_i [ns]	Mag. A_i [dB]
0	0	90	−7.8
10	−0.9	110	−4.7
20	−1.7	140	−7.3
30	−2.6	170	−9.9
40	−3.5	200	−12.5
50	−4.3	240	−13.7
60	−5.2	290	−18
70	−6.1	340	−22.4
80	−6.9	390	−26.7

7.1. BRAN A Radio Channel

In Table 1, the values corresponding the BRAN A radio channel impulse response are shown [6], [24].

In Fig. 5 the estimation of the impulse response of BRAN A channel is presented using the blind and adaptive algorithms in the case of $SNR = 24$ dB and data length $N = 4096$.

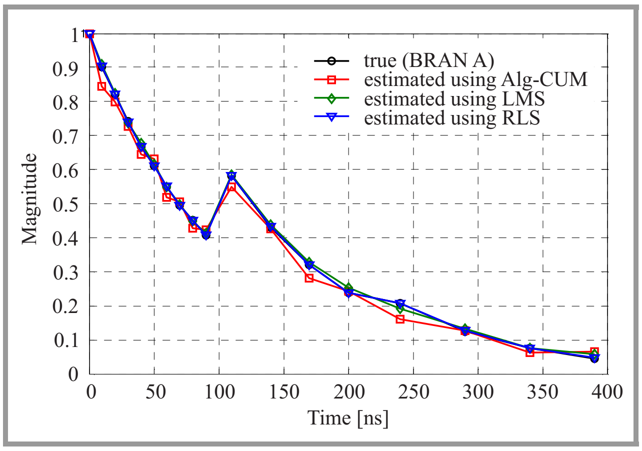

Fig. 5. Estimated of BRAN A channel impulse response, for an $SNR = 24$ dB and a data length $N = 4096$.

The estimated BRAN A channel impulse response using the adaptive algorithms (RLS and LMS) and the Algo-CUM algorithm are much closed to the true type, for data length $N = 4096$ and $SNR = 24$ dB. The robustness of the Algo-CUM proposed algorithm comparatively to the adaptive algorithms allows to act: without information about the input signal, and gives good estimation of the BRAN A channel. It is in opposite to the RLS and LMS versions in which the authors exploit the information of input and output for the estimation of the impulse response channel.

In Fig. 6 the estimated magnitude and phase of the impulse response BRAN A is presented, for $N = 4096$ and $SNR = 24$ dB using the all algorithms.

From the Fig. 6 the authors conclude that the magnitude and phase estimations using blind and adaptive algorithms

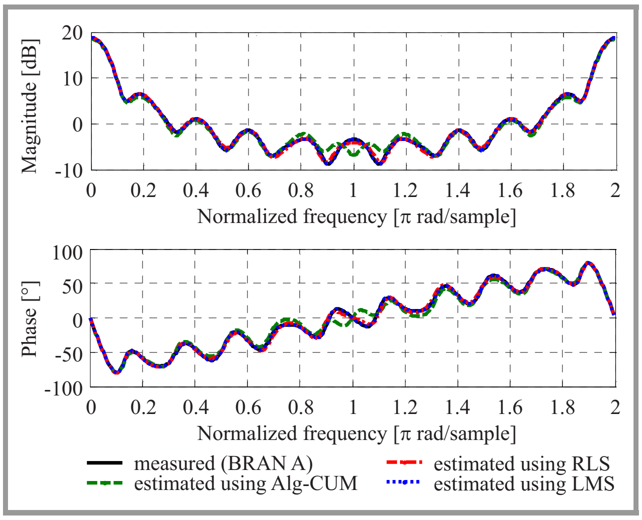

Fig. 6. Estimated magnitude and phase of BRAN A channel impulse response using all target, for $SNR = 24$ dB and $N = 4096$.

have the same allure comparatively to the true ones. However, the algorithm has the advantage of blind estimation of the channel parameters, i.e., without any information about the input.

7.2. Bran B Radio Channel

In Table 2, the values corresponding to the BRAN B radio channel impulse response are presented [24].

Table 2
Delay and magnitudes of 18 targets of BRAN B channel

Delay τ_i [ns]	Mag. A_i [dB]	Delay τ_i [ns]	Mag. A_i [dB]
0	−2.6	230	−5.6
10	−3.0	280	−7.7
20	−3.5	330	−9.9
30	−3.9	380	−12.1
50	0.0	430	−14.3
80	−1.3	490	−15.4
110	−2.6	560	−18.4
140	−3.9	640	−20.7
180	−3.4	730	−24.6

In Fig. 7, the estimation of the impulse response of BRAN B channel is shown using the blind and adaptive algorithms in the case of $SNR = 24$ dB and data length $N = 4096$.

Figure 7 shows that the estimated BRAN B channel impulse response, using the blind and adaptive algorithms, is closed to the true type, for data length $N = 4096$ and $SNR = 24$ dB, but the blind algorithm have the advantage of estimate the impulse response of BRAN B channel blindly with the faible calculate cost, comparing to RLS and LMS implementations.

In Fig. 8, the estimated magnitude and phase of the impulse response BRAN B are presented using all target, for an data length $N = 4096$ and $SNR = 24$ dB, obtained using blind algorithm, compared with the adaptive algorithms (RLS, LMS).

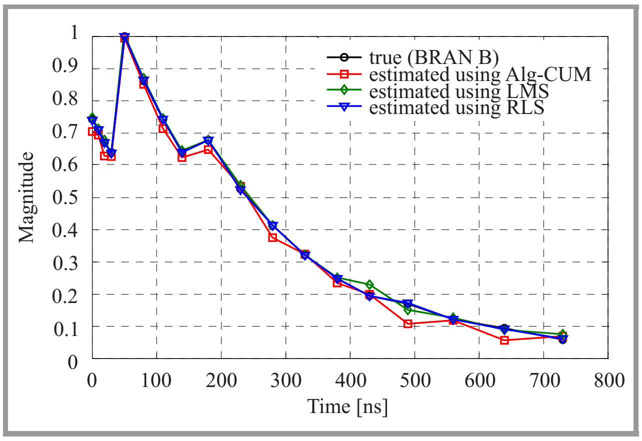

Fig. 7. Estimated of BRAN B channel impulse response, for an $SNR = 24$ dB and a data length $N = 4096$.

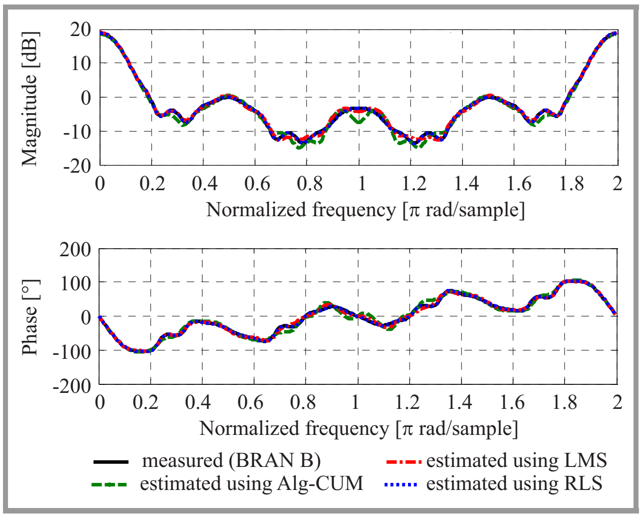

Fig. 8. Estimated magnitude and phase of BRAN B channel impulse response using all target, for an $SNR = 24$ dB and a data length $N = 4096$.

Figure 8 shows, that the estimated magnitude and phase, using blind and adaptive algorithms, have the same form showing no difference between the estimated and the true type version.

8. MC-CDMA System Application

To evaluate the performance of MC-CDMA systems using the blind and adaptive algorithms, the Bit Error Rate, based on two equalizers (ZF and MMSE) and measured and estimated BRAN A and BRAN B channels impulse response are computed. The results are evaluated for different values of SNR.

8.1. ZF and MMSE Equalizers – Case of BRAN A Channel

In Fig. 9, the BER estimation simulation results for the blind and adaptive algorithms using BRAN A channel estimation are shown. The equalization is performed using ZF equalizer.

Figure 10 depicts the BER for different SNR using the blind and adaptive algorithms for BRAN A channel. The equalization is performed using the MMSE equalizer.

The BER results for different SNR values demonstrates that the results obtained by the blind algorithm are similar to those obtained using the adaptive algorithms (RLS and LMS).

With $SNR = 24$ dB, for all algorithms BER = 10^{-4} can be achieved only using ZF equalizer, and using MMSE it lowers near to 10^{-5}. The proposed algorithm is very interesting because it is able to estimate the impulse response of these channels blindly.

8.2. ZF and MMSE Equalizers – Case of BRAN B Channel

Figure 11 shows the BER for different SNR values using the blind and adaptive algorithms for BRAN B channel. The equalization is performed using the ZF equalizer.

It demonstrates clearly that the BER parameter obtained using all algorithms gives good results comparable to

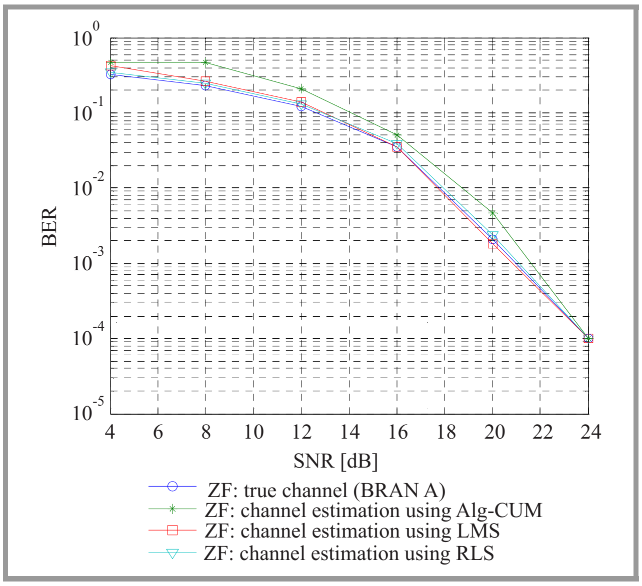

Fig. 9. BER parameter for estimated and measured BRAN A channel using the ZF equalizer.

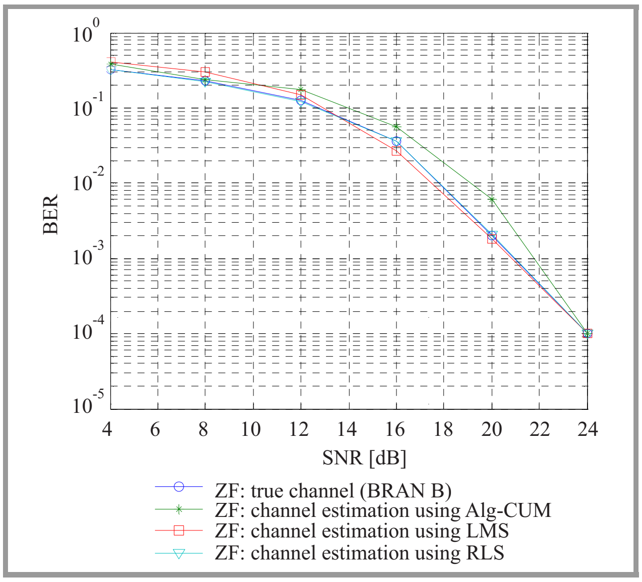

Fig. 11. BER of the estimated and measured BRAN B channel using the ZF equalizer.

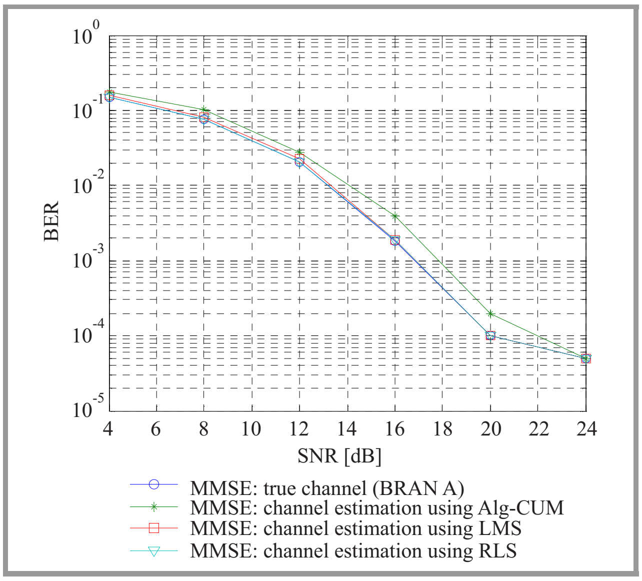

Fig. 10. BER coefficient for an estimated and measured BRAN A channel using the MMSE equalizer.

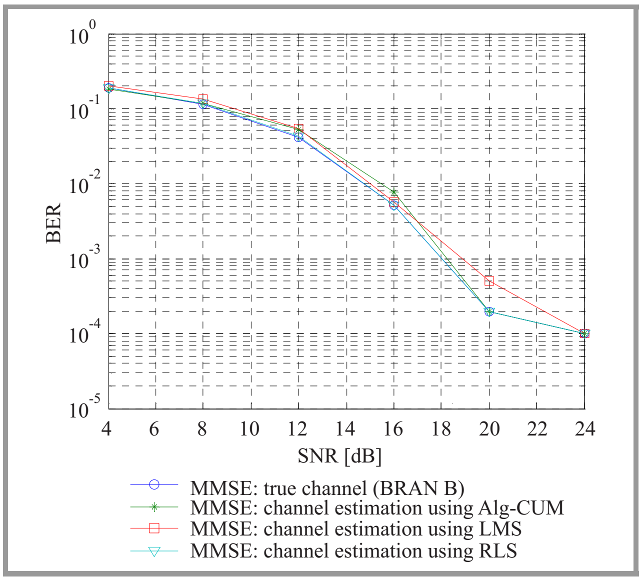

Fig. 12. BER of the estimated and measured BRAN B channel using the MMSE equalizer.

these obtained using measured values for ZF equalization. Therefore, if the $SNR = 24$ dB, for all algorithms BER stays at 10^{-4} level.

Figure 12 illustrates the BER for different SNR values, using the blind and adaptive algorithms for BRAN B channel. The equalization is performed using the MMSE equalizer.

It can be observed that the MMSE equalization, for all algorithms gives the same results as obtained using the measured BRAN B values. Then if the SNR values are superior to 20 dB, BER is 10^{-4} bit. However, if the SNR is superior to 24 dB, there is a BER lower than 10^{-4}.

9. Conclusion

In this paper, a comparative study between the adaptive algorithms (RLS and LMS) and blind algorithm based on third order cumulants was presented. These algorithms are performed in the channel parameters identification, such as the experimental channels, BRAN A and BRAN B. The simulation results show that they are efficient, but the blind algorithm presents the advantage to estimate the impulse response of frequency selective channel blindly with low calculation power required, comparing to RLS and LMS. The magnitude and phase of the impulse response are estimated with a good precision in noisy environment principally for high data record length. In the MC-CDMA equalization part, a good results using the proposed algorithm have been obtained comparatively to LMS and RLS algorithms.

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Paper

QoS Equalization in a W-CDMA Cell Supporting Calls of Infinite or Finite Sources with Interference Cancelation

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Abstract—In this paper, a multirate loss model for the calculation of time and call congestion probabilities in a Wideband Code Division Multiple Access (W-CDMA) cell is considered. It utilizes the Bandwidth Reservation (BR) policy and supports calls generated by an infinite or finite number of users. The BR policy achieves QoS equalization by equalizing congestion probabilities among calls of different service-classes. In the proposed models a multiple access interference is considered, and the notion of local blocking, user's activity and interference cancelation. Although the analysis of the proposed models reveals that the steady state probabilities do not have a product form solution, the authors show that the calculation of time and call congestion probabilities can be based on approximate but recursive formulas, whose accuracy is verified through simulation and found to be quite satisfactory.

Keywords—bandwidth reservation, infinite/finite sources, recursive formula, time-call congestion probabilities, W-CDMA.

1. Introduction

Wideband Code Division Multiple Access (W-CDMA) networks support calls from different service-classes with heterogeneous Quality of Service (QoS) requirements. The existence of own-cell and other-cell interference in these networks, increases the complexity of the call-level analysis both in the uplink (UL) and downlink directions.

We study the UL direction of a W-CDMA cell that has fixed capacity and supports K service-classes whose calls are generated by an infinite and finite number of sources. In the first case, the call arrival process is the Poisson process while in the second case the call arrival process is a quasi-random process [1]. Calls of service-class k $(k = 1,\dots,K)$ have a fixed bandwidth requirement and an exponentially distributed service time. According to the CDMA principle of W-CDMA networks a call is noise for all in-service calls. Therefore, a new call is accepted in the cell if its bandwidth requirement is available and the noise of all in-service calls remains below a tolerable level.

To take into account the interference increase caused by the acceptance of the new call, the notion of Local Blocking (LB) was adopted in analysis. The latter means that a new call can be blocked in any system state, if its acceptance results in the increase of noise of all in-service calls above a threshold.

We model a reference W-CDMA cell as a multirate teletraffic loss system, and aim at calculating Time and Call Congestion Probabilities (TC and CC probabilities, respectively) via recursive formulas [2]–[4]. In [2]–[4], the calculation of congestion probabilities is based on the classical Kaufman-Roberts recursive formula used in the Erlang Multirate Loss Model (EMLM). The Kaufman-Roberts formula determines, in an efficient way, the link occupancy distribution for a single link that accommodates, under the Complete Sharing (CS) policy, Poisson calls of K service-classes with different bandwidth requirements and generally distributed service time [5], [6]. In [2], an extension of the EMLM is proposed, based on the Delbrouck's model (where a Bernoulli/Poisson/Pascal call arrival process is considered) [7]. This model allows new calls to have different peakedness factors. In [3], new calls are generated by an infinite number of sources, i.e., calls follow a Poisson process. In [4], calls come from a finite number of sources, a rather realistic case since cells have limited coverage area. As far as the LB modeling is concerned, two approaches exist in the literature. The first ensures reversibility in the underlying Markov state transition diagram but is complex [2]. We adopt the second approach, proposed in [3], since it is simpler and more realistic for W-CDMA systems. The interested reader may resort to [4] for a comparison of these approaches.

In this paper, a research from [3] and [4] is extended by applying the Bandwidth Reservation (BR) policy to guarantee call-level QoS for each service-class. In particular, an equalization of TC or CC probabilities is achieved among different service-classes by reserving bandwidth in favor of service-classes whose calls have high bandwidth requirements. Applications of the BR policy in wired (e.g., [8]–[13]), wireless (e.g., [14]–[17]) and optical networks (e.g., [18],[19]) show the importance of the policy in teletraffic engineering. In addition, the authors study the effect of Interference Cancelation (IC) on congestion probabilities and provide recursive formulas for their calculation. Note that IC receivers reduce own-cell interference and thus decrease congestion [4].

This paper is organized as follows. In Section 2, the basic formulas in the UL of a W-CDMA cell are reviewed. In Section 3, random (Poisson) arrivals are considered and recursive formulas for the calculation of TC and CC probabilities under the BR policy are proposed. In Section 4, the case of quasi-random arrivals is considered. In Section 5, numerical results are presented and evaluated by simulation. The paper is concluded in Section 6.

2. Basic Formulas in the UL of a W-CDMA Cell

Consider the UL direction of a W-CDMA reference cell which is controlled by a Base Station (BS) and surrounded by other cells. This cell is modeled as a multirate loss system that supports K different service-classes. A service-class k $(k = 1,\ldots,K)$ call, when accepted in the system, alternates between transmission (active) and non-transmission (passive) periods. The ratio of "active" over "active + passive" periods is the activity factor of a service-class k call, v_k.

In the W-CDMA cell, a user "sees" the signals generated by other users as interference. Thus, the BS's capacity is limited by the own-cell interference, P_{own}, caused by the users of the reference cell and the other-cell interference, P_{other}, caused by the interference power received from users of the neighboring cells. Due to the stochastic nature of interference, we consider the interference limited capacity of the radio interface. Thermal noise is also considered, P_{noise}, which corresponds to the interference of an empty W-CDMA system. The values of P_{own} are reduced by the application of IC, whose efficiency, β, can be determined by [20]:

$$\beta = \frac{P_{\text{own}}^{\text{No IC}} - P_{\text{own}}}{P_{\text{own}}^{\text{No IC}}} \Rightarrow P_{\text{own}} = P_{\text{own}}^{\text{No IC}}(1-\beta), \qquad (1)$$

where $P_{\text{own}}^{\text{No IC}}$ is the own-cell interference without IC.

Let P_k be the total received power from a service-class k user. Then, the power control equation is [20]:

$$(E_b/N_0)_k = \frac{G_k P_k}{(P_{\text{own}} - P_k)(1-\beta) + P_{\text{other}} + P_{\text{noise}}}, \qquad (2)$$

where $(E_b/N_0)_k$ is the signal energy per bit divided by the noise spectral density, $G_k = W/v_k R_k$ is the processing gain of service-class k in the UL with data rate R_k and W the chip rate of 3840 kcps. Based on Eq. (2), the values of P_k are given by:

$$P_k = (P_{\text{own}}(1-\beta) + P_{\text{other}} + P_{\text{noise}})/(1-\beta+G_k/(E_b/N_0)_k). \qquad (3)$$

Assuming that $P_{\text{own}} = P_k N_k$, where N_k is the maximum number of service-class k calls in the cell, we have [4]:

$$P_{\text{own}} = \frac{N_k(P_{\text{other}}+P_{\text{noise}})}{1-\beta-N_k(1-\beta)+G_k/(E_b/N_0)_k}. \qquad (4)$$

Consider now the Noise Rise (NR), defined as [21]:

$$NR = P_{total}/P_{noise} = (P_{own}+P_{other}+P_{noise})/P_{noise}, \qquad (5)$$

where $P_{\text{total}} = P_{\text{own}}+P_{\text{other}}+P_{\text{noise}}$ is the total received power at the BS.

The relation between the NR and the total UL cell load, η_{UL}, is given by [22]:

$$NR = 1/(1-\eta_{UL}), \qquad \eta_{UL} = (P_{\text{own}}+P_{\text{other}})/P_{\text{total}}. \qquad (6)$$

Based on Eqs. (4)–(6) it is proved that [4]:

$$N_k = \left[(1-\beta) + G_k/\left(E_b/N_0\right)_k\right] \frac{[\eta_{UL}(\delta+1)-\delta]}{[1-\beta(\eta_{UL}(\delta+1)-\delta)]}, \quad \delta \equiv P_{\text{other}}/P_{\text{noise}}. \qquad (7)$$

Based on Eq. (7), the spread data rate $R_{s,k}$ of service-class k, as a proportion of W is determined:

$$R_{s,k} = W/N_k. \qquad (8)$$

Now, we transform W and $R_{s,k}$ to the capacity C and the bandwidth b_k, of each service-class k, respectively. This is achieved by considering a basic bandwidth unit (*bbu*) as the greatest common divisor of the bandwidth of all service-classes, or as an arbitrarily chosen small value. So, $C = \lceil W/bbu \rceil$ and $b_k = \lceil R_{s,k}/bbu \rceil$ channels.

3. Congestion Probabilities Under the BR Policy – the Case of Random Arrivals

Consider a new service-class k call that arrives in the cell according to a Poisson process with mean arrival rate λ_k and requires b_k channels in order to be accepted in the system. Let j be the number of occupied cell's channels at the time of arrival, $j = 0,1,\ldots,C$. Also, let t_k be the BR parameter that expresses the reserved channels to benefit calls of all service-classes other than service-class k. Due to the BR policy, the service-class k call is not allowed to enter the states $j = C-t_k+1,\ldots,C$. These states form the so-called reservation space of service-class k. So, after the acceptance of the call, $j \le C-t_k$, i.e., the available capacity upon the call arrival is $C-t_k-j$. Now two types of blocking states j are considered: hard blocking states due to bandwidth unavailability and soft (local) blocking states with a probability $0 < L_{j,k} < 1$ (LB states) due to the other-cell interference. The latter is approximated by an independent, lognormally distributed random variable, with parameters μ and σ [4]:

$$\mu = \frac{P_{\text{other}}+P_{\text{noise}}}{P_{\text{own}}+P_{\text{other}}+P_{\text{noise}}} C \Rightarrow \mu = \frac{i+i/\delta}{1+i+i/\delta} C, \quad \sigma = \mu, \qquad (9)$$

where $i = P_{\text{other}}/P_{\text{own}}$.

The LB probability (LBP) in state j, L_j, is the probability that the other-cell interference is greater than the available cell's capacity $(C-t_k-j)$ [4]:

$$L_j = 1 - P\left(j' < C-t_k-j\right) = 1 - CDF(C-t_k-j), \qquad (10)$$

where j' refers to the occupied channels due to the other cell interference and $CDF(x)$ is the cumulative distribution function of the lognormal distribution.

The values of $CDF(x)$ are given by:

$$CDF(x) = \frac{1}{2}\left(1 + erf\left(\frac{\ln(x) - M}{S\sqrt{2}}\right)\right), \quad (11)$$

where erf is the error function, while M and S refer to the parameters of the normal distribution:

$$M = \ln\left(\mu^2 \Big/ \sqrt{\mu^2 + \sigma^2}\right), \quad S = \sqrt{\ln\left(1 + \left(\sigma^2/\mu^2\right)\right)}. \quad (12)$$

The service-class k call is accepted in the cell if all b_k channels are assigned to the call simultaneously. Thus, it is assumed that P_{other} and LBP do not alter during this allocation process. We express the passage factor $1 - L_{j,b_k}$, i.e., the probability that the call is not blocked due to the other-cell interference as a function of j and b_k:

$$1 - L_{j,b_k} = 1 - L_{j+b_k+1} = CDF(C - t_k - j - b_k + 1). \quad (13)$$

Thus, the transition rate from $(j - b_k)$ to (j) equals $\left(1 - L_{j-b_k,b_k}\right)\lambda_k = \left(1 - L_{j-1}\right)\lambda_k$. Figure 1 presents an excerpt of the system's state transition diagram, which is depicted by a one-dimensional Markov chain. Note that μ_k is the mean service rate of service-class k calls, while $y_k(j)$ is the average number of service-class k calls in state j.

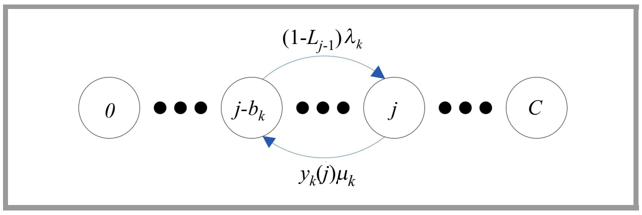

Fig. 1. State transition diagram for Poisson arriving service-class k calls with LB between states $j - b_k$ and j.

To calculate the un-normalized values of the system state probabilities, $q(j)$, the following approximate but recursive formula is proposed:

$$q_{\inf}(j) = \begin{cases} 1, & \text{for } j = 0 \\ \frac{1}{j}\sum\limits_{k=1}^{K} a_k D_k(j-b_k) q_{\inf}(j-b_k)(1-L_{j-b_k,b_k}) & \text{for } j = 1,\ldots,C \\ 0, & \text{otherwise} \end{cases}, \quad (14)$$

$$D_k(j - b_k) = \begin{cases} b_k & \text{for } j \le C - t_k \\ 0 & \text{for } j > C - t_k \end{cases}, \quad (15)$$

where: $\alpha_k = \lambda_k/\mu_k$ is the offered traffic-load of service-class k calls (in erl), t_k is the BR parameter, while the values of $\left(1 - L_{j-b_k,b_k}\right)$ are determined by:

$$\left(1 - L_{j-b_k,b_k}\right) = 1 - L_{j-1} = CDF(C - t_k - j + 1). \quad (16)$$

Note that Eq. (15) facilitates the introduction of the BR policy in the model. The underlying assumption of Eq. (15) is that the population of service k calls, which require b_k channels while $t_k > 0$, is negligible inside the reservation space of service-class k, i.e., when $j = C - t_k + 1, \ldots, C$. In the case of the CS policy, Eq. (14) takes the form [3], [4]:

$$q_{\inf}(j) = \begin{cases} 1, & \text{for } j = 0 \\ \frac{1}{j}\sum\limits_{k=1}^{K} a_k b_k q_{\inf}(j-b_k)(1-L_{j-b_k,b_k}), & \text{for } j = 1,\ldots,C \\ 0, & \text{otherwise} \end{cases}. \quad (17)$$

If we do not consider the existence of LB, then the classical Roberts' formula for the EMLM under the BR policy arises [8]:

$$q_{\inf}(j) = \begin{cases} 1, & \text{for } j = 0 \\ \frac{1}{j}\sum\limits_{k=1}^{K} a_k D_k(j-b_k) q_{\inf}(j-b_k), & \text{for } j = 1,\ldots,C \\ 0, & \text{otherwise} \end{cases}, \quad (18)$$

where the values of $D_k(j - b_k)$ are given by Eq. (15).

Having determined $q_{inf}(j)$'s according to Eq. (14), TC probabilities of service-class k, P_{b_k}, can be calculated as follows:

$$P_{b_k} = \sum_{j=0}^{C} G^{-1} L_{j,\, j+b_k} q_{\inf}(j), \quad (19)$$

where $G = \sum_{j=0}^{C} q_{\inf}(j)$ is the normalization constant and the values of $L_{j,\, j+b_k} = 1 - CDF(C - t_k - j - b_k + 1)$.

Note that TC probabilities refer to the proportion of time the system is congested, while CC probabilities refer to the proportion of arriving calls that find the system congested. TC and CC probabilities coincide in the case of Poisson arrivals due to the PASTA property [1].

4. Congestion Probabilities Under the BR Policy – the Case of Quasi-Random Arrivals

In the case of quasi-random arrivals, this part follows again the analysis of Section 3 up to Eq. (13). At this point, the transition rate from $(j - b_k)$ to (j), becomes: $\left(1 - L_{j-b_k,b_k}\right)(S_k - \overline{n_k}(j-b_k))\gamma_k = \left(1 - L_{j-1}\right)(S_k - \overline{n_k}(j-b_k))\gamma_k$, where S_k is the finite number of service-class k traffic sources, γ_k is the arrival rate from an idle source of service-class k and $\overline{n_k}(j)$ is the average number of service-class k

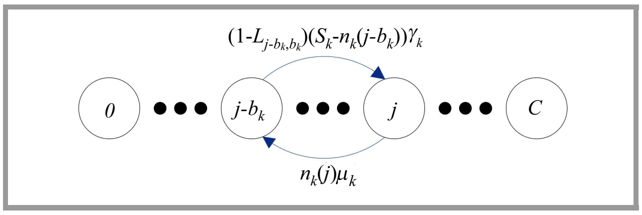

Fig. 2. State transition diagram for quasi-random arriving service-class k calls with LB between states $j - b_k$ and j.

calls in state j. Figure 2 shows the corresponding one-dimensional Markov chain.

To determine the un-normalized values of $q(j)$'s we propose the following recursive formula, for $j = 1, \ldots, C$:

$$q(j)=\frac{1}{j}\sum_{k=1}^{K}(S_k-n_k(j)+1)a_{k,\text{fin}}D_k(j-b_k)q(j-b_k)(1-L_{j-b_k,b_k}), \tag{20}$$

where: $q(0)$=1, $q(x)$=0 for $x < 0$, $a_{k,\text{fin}} = \gamma_k/\mu_k$ is the offered traffic-load per idle source of service-class k (in erl), $\overline{n_k}(j-b_k) = n_k(j)-1$, $n_k(j)$ refers to the number of in-service calls of service-class k in state j, while the values of $D_k(j-b_k)$ and $\left(1-L_{j-b_k,b_k}\right)$ are given by Eqs. (15) and (16), respectively.

In the case of the CS policy, Eq. (20) takes the form [4]:

$$q(j) = \frac{1}{j}\sum_{k=1}^{K}(S_k-n_k(j)+1)a_{k,fin}b_k q(j-b_k)(1-L_{j-b_k,b_k}), \tag{21}$$

Note that if $S_k \to \infty$ for $k = 1, \ldots, K$, and the total offered traffic-load remains constant, then the call arrival process is Poisson. In that case, Eqs. (20) and (21) become Eqs. (14) and (17), respectively.

The determination of $q(j)$'s in Eqs. (20) or (21) requires the values of $n_k(j)$ in each state j. These values are unknown and difficult to be determined. In other finite multirate loss models (e.g., [23]–[26]) there exist methods for the determination of $n_k(j)$ through an equivalent stochastic system, with the same traffic description parameters and exactly the same set of states. However, the state space determination of the equivalent system is complex, especially for large capacity systems that serve many service-classes. Thus, $n_k(j)$ is approximated, as the mean number of service-class k calls in state j, $y_k(j)$, when Poisson arrivals are considered, i.e., $n_k(j) \approx y_k(j)$ and consequently $n_k(j) - 1 \approx y_k(j-b_k)$. Such approximations induce little error (e.g., [27]–[33]).

Based on the abovementioned approximation, Eqs. (20) and (21) take the form of (22) and (23), respectively:

$$q(j)=\frac{1}{j}\sum_{k=1}^{K}(S_k-y_k(j-b_k))a_{k,\text{fin}}D_k(j-b_k)q(j-b_k)(1-L_{j-b_k,b_k}), \tag{22}$$

$$q(j)=\frac{1}{j}\sum_{k=1}^{K}(S_k-y_k(j-b_k))a_{k,\text{fin}}b_k q(j-b_k)(1-L_{j-b_k,b_k}), \tag{23}$$

where the values of $y_k(j)$, for Poisson arrivals, are given by:

$$y_k(j) = a_k(1-L_{j-b_k,b_k})q_{\text{inf}}(j-b_k)/q_{\text{inf}}(j). \tag{24}$$

Having determined LBP by Eq. (16) and $q(j)$'s by Eq. (22) TC probabilities of service-class k calls is calculated, P_{b_k}, based on Eq. (19). Equation (19) can also be used for the determination of CC probabilities of service-class k, but $q(j)$'s should be calculated by Eq. (22) assuming $S_k - 1$ traffic sources.

5. Numerical Examples – Evaluation

The authors compare the analytical and simulation TC probabilities results obtained by the proposed models for different values of the IC efficiency β. For further comparison, the corresponding analytical results obtained in the case of the CS policy, for both Poisson and quasi-random arrivals [4] are also shown. Simulations are based on the SIMSCRIPT III language [34] and are mean values of 7 runs.

Consider a W-CDMA reference cell that accommodates quasi-random arriving calls of K=3 different service-classes. Accepted calls remain in the system for an exponentially distributed service time with mean value $\mu_1^{-1} = \mu_2^{-1} = \mu_3^{-1} = 1$. Table 1 presents the traffic characteristics of all service-classes. In addition, the following assumptions were made: $\eta_{UL} = 0.75$, $i = 0.35$, $\delta = 2$, $bbu = 13.5$ kcps, while the IC efficiency β takes the values 0.0 and 0.8. When $\beta = 0$, the bandwidth requirements and the corresponding BR parameters of all service-classes are: $b_1 = 4$, $b_2 = 7$, $b_3 = 64$ and $t_1 = 60$, $t_2 = 57$, $t_3 = 0$. The values of the BR parameters are chosen according to the rule: $b_1+t_1 = b_2+t_2 = b_3$, to achieve equalization of congestion probabilities. Similarly, when $\beta = 0.8$ then $b_1 = 4$, $b_2 = 5$, $b_3 = 54$ and $t_1 = 50$, $t_2 = 49$, $t_3 = 0$.

Table 1
Traffic parameters of all service-classes

Serv.-class k	R_k [kb/s]	v_k	$\left(\frac{E_b}{N_0}\right)_k$ [dB]	$\left(\frac{E_b}{N_0}\right)_k$	S_k	$a_{k,\text{fin}}$ [erl]	a_k [erl]
1	7.95	0.67	4.0	2.51	20	0.15	3.0
2	12.20	0.67	4.0	2.51	10	0.20	2.0
3	144.00	1.00	2.0	1.58	5	0.01	0.05

In the x-axis of Figs. 3–8 the offered traffic load of the 1st, 2nd and 3rd service-class increase in steps of 0.05, 0.10 and 0.002 erl, respectively. So, point 1 refers to: $(a_{1,\text{fin}}, a_{2,\text{fin}}, a_{3,\text{fin}} = (0.15, 0.20, 0.01)$ while point 6 to: $(a_{1,\text{fin}}, a_{2,\text{fin}}, a_{3,\text{fin}} = (0.40, 0.70, 0.02)$. Figures 3–4 present the analytical and simulation results of the 1st service-class for $\beta = 0$ and 0.8, respectively. Similarly, in Figs. 5–6 and 7–8, the corresponding results of the 2nd and 3rd service-class are presented, respectively. The proposed formulas for the calculation of the occupancy distribution and consequently TC probabilities in the case of the BR policy give quite accurate results in comparison with the simulation results. The increase of β results in the TC probabilities decrease, since the IC reduces the own-cell interference. The TC probabilities obtained by considering the CS policy fail to approximate the corresponding TC probabilities in the case of the BR policy. The application of the BR policy results in a slight decrease of the TC probabilities of the 3rd service-class compared to the increase of the TC probabilities of the other two service-classes. This is expected since the bandwidth per call requirement of the 3rd service-class is much higher (64 b.u.) than the requirements of the other service-classes (7 and 4 b.u.).

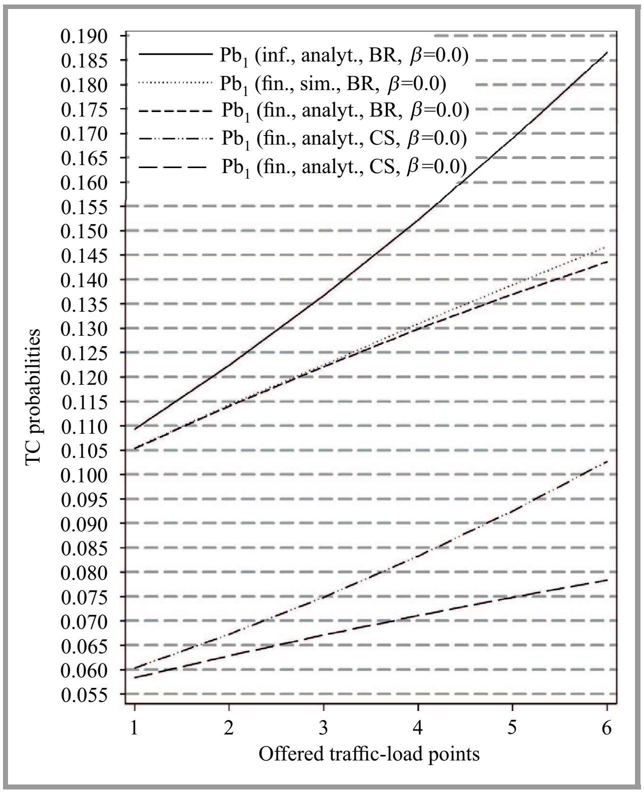

Fig. 3. TC probabilities – 1st service-class ($\beta = 0$).

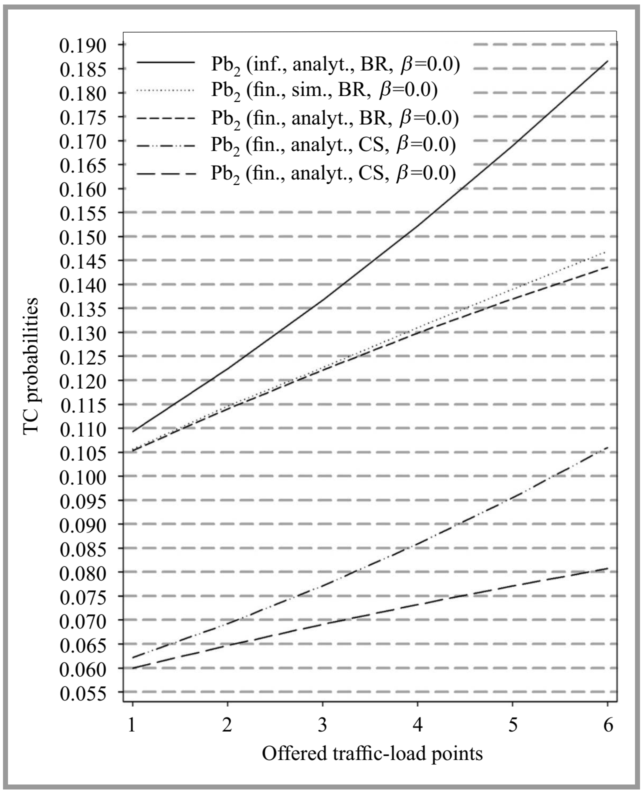

Fig. 5. TC probabilities – 2nd service-class ($\beta = 0$).

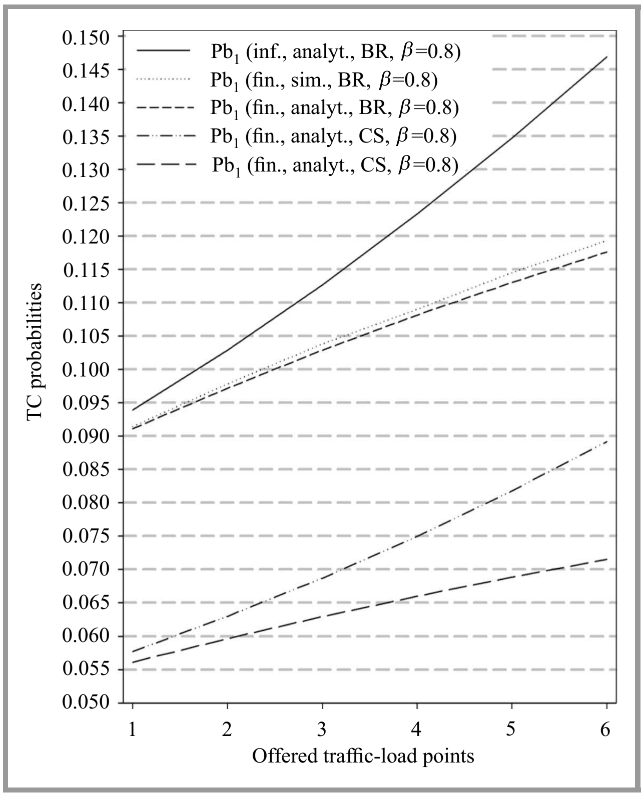

Fig. 4. TC probabilities – 1st service-class ($\beta = 0.8$).

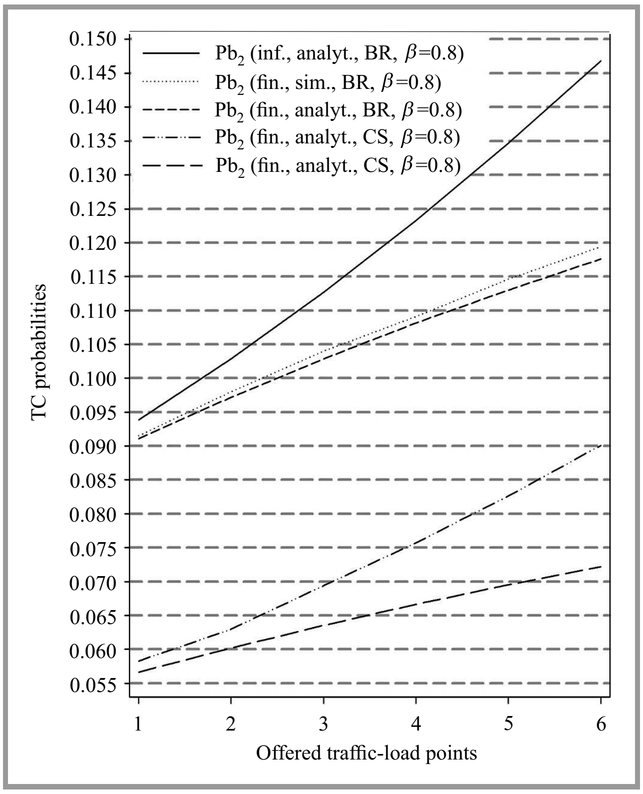

Fig. 6. TC probabilities – 2nd service-class ($\beta = 0.8$).

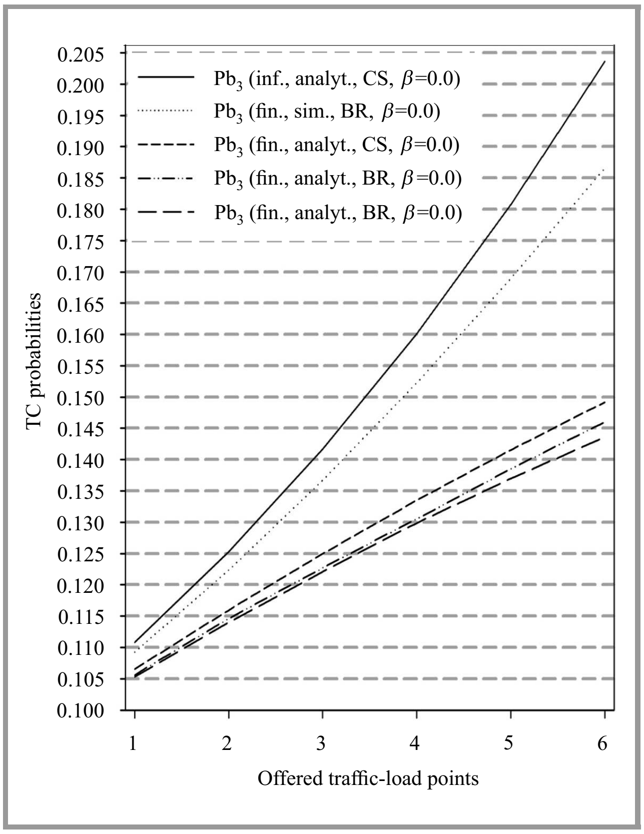

Fig. 7. TC probabilities – 3[rd] service-class ($\beta = 0$).

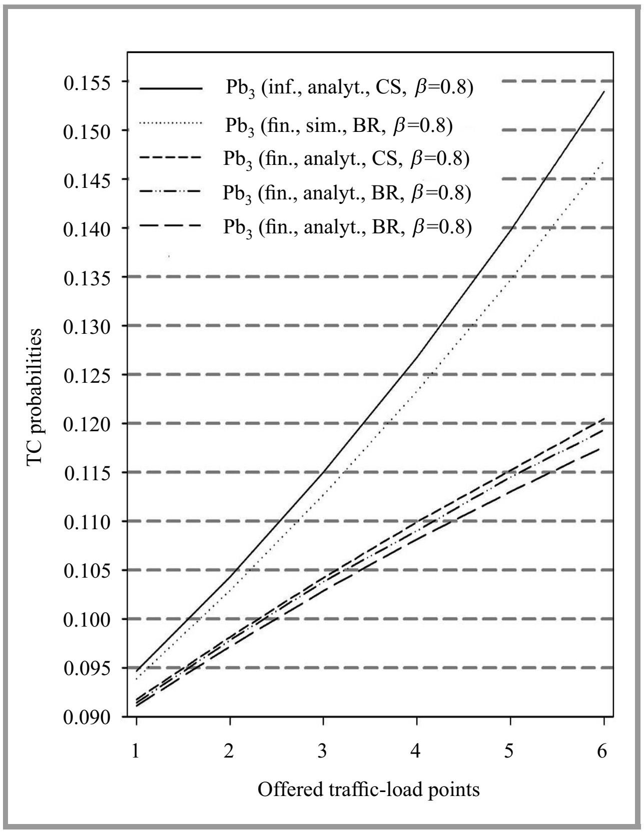

Fig. 8. TC probabilities – 3[rd] service-class ($\beta = 0.8$).

6. Conclusion

In this paper authors propose multirate loss models for the call-level analysis of a W-CDMA reference cell that supports calls from different service-classes with different bandwidth requirements. New call arrivals follow a Poisson arrival process, or a quasi-random arrival process. The proposed models take into account important peculiarities of wireless networks, such as multiple access interference, the notion of local blocking, user's activity, interference cancelation and the BR policy. The latter is used to achieve equalization of congestion probabilities among calls of different service-classes. Due to the existence of local blocking and the BR policy in the proposed models, the calculation of the occupancy distribution (and consequently of time and cal congestion probabilities) is based on an approximate but recursive formula. Simulation results verify the proposed model accuracy.

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Paper

On IPv6 Experimentation in Wireless Mobile Ad Hoc Networks

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Abstract—With the growing interest towards the Internet of Things IPv6-based mobile ad hoc networks (MANETs) become a key enabling technology offering the possibility of automated, unsupervised network configuration and operation. Such a functionality calls for an accurate and reliable testing of the newly proposed solutions, which is challenging due to the dynamic, decentralized and ad hoc nature of MANETs. In this work selected topics are presented on performing IPv6 protocols experimentation in wireless, IPv6-only mobile ad hoc networks – including both simulation – and testbed-based evaluation. Based on the authors experience with the evaluation of the extended IPv6 Neighbor Discovery protocol (ND++) proposed during the course of research, the selection of an open-source simulation environment is presented and a comparison between simulation and emulation experimentation methods is provided. Benefits and drawbacks of both these methodologies for testing IPv6 solutions are depicted. Moreover, the important aspects of topology and mobility considerations are considered. Finally the authors propose a testing approach that would allow for a detailed and accurate evaluation by means of open-source, easily accessible and low-cost methodologies.

Keywords*—*IPv6 simulation, IPv6 wireless testbed, MANET experimentation, MANET protocols evaluation, Neighbor Discovery ++ (ND++), NS-3.

1. Introduction

The features of mobile ad hoc networks (MANETs), which allow them to adapt, dynamically follow the changing networking environment and perform well without a pre-established infrastructure, make them an ideal basis for the Internet of Things. Accompanied with the IPv6 protocol stack, IPv6-based MANETs constitute a perfect solution for bringing Future Internet into the world of connected devices [1], [2]. Internet of Things, though, calls for an automated, unsupervised network configuration and operation [2], [3], since it is expected to ensure sustainable network functionality with minimal external supervision. This implies the need for a detailed, accurate and reliable testing of the newly proposed solutions. Especially stateless address autoconfiguration, as a means of a "plug and play" network set-up, is among the key IPv6 mechanisms that require thorough testing in many realistic and demanding MANET scenarios. This allows to ensure their performance at the level corresponding to the high users and network maintenance expectations.

Testing the newly proposed IPv6 solutions usually is a two-step approach: at the first stage the research idea is evaluated in the course of simulations, secondly the real-world evaluation at the testbed platforms and field trials are performed. Simulations of MANET networks can be executed by means of several available network simulators, including commercial (e.g. OPNET/Riverbed [4] or QualNet [5]) and open-source ones (e.g. OMNET++ [6] or NS-3 [7]), as well as the in-house simulators created to address particular needs of the conducted research. The second step, requiring close to real world conditions, is very often having a pre-commercial character. In case of testing IPv6 networks it can be performed on the big testbed platforms, including those certified with IPv6Ready logo [8], [9]. These are, however, usually designed as the fixed networks environment. Due to the very specific nature of mobile ad hoc networks these testbeds are not suitable for testing most of the MANET-dedicated IPv6 solutions, which leaves MANET researchers and developers at the difficult position. There are few available test sites that allow for testing significant network sizes (tens or hundreds of nodes), like e.g. Open-Access Research Testbed for Next-Generation Wireless Networks (ORBIT) [10], [11]. However, the access to them is in most cases limited and very costly. As such many researchers tend to evaluate their solutions on the in-house testbed platforms comprising of several laptops or other machines [12]–[15]. The drawback of such an approach is very often that the test suite, due to its characteristic, is limited to very few fixed scenarios [12]–[14], which may not be enough to obtain the wide range of accurate results and is rather useful for a "proof-of-concept" type of experiments. In general, the authors tend to observe that many MANET researchers present their results on the small-scale, simplified models [12], [16]–[18] – both in simulation and emulation environments. Whereas in the case of emulation the limitations may be a result of the hardware capabilities and particular testbed characteristics, for simulations they are usually reflecting the lack of adequate topology and mobility considerations.

In this article the authors present the lessons learned in the course of evaluating the newly proposed IPv6 autoconfiguration solution for MANET networks – the Neighbor Discovery++ (ND++) protocol [19], [20]. The aim was to come up with a high-quality testing approach that would allow for a detailed and accurate evaluation by means of open-source, easily accessible and low-cost methodologies. The authors will depict issues related to the selection of

most suitable simulation environment and present the aspects of topology and mobility selection for the reliable simulation set-ups. Referring to the second evaluation stage the simulation-based performance measures will be compared to the real-world testbed experimentation performed at the dedicated wireless testbed platform designed especially to enable creation of multiple MANET scenarios. Both benefits and drawbacks of the two evaluation methods – simulation and emulation are presented, and reveal their complementary nature.
The structure of this article is organized as follows. First the research goal driving the methodology selection is presented in Section 2 and an overview of experimentation objectives and requirements are shown in Section 3. The wireless network simulators for IPv6 experimentation are depicted in Section 4 accompanied with topology considerations in Section 5. Finally Section 6 presents a comparison between simulation- and emulation-based experimentation of IPv6 solutions and a proposed IPv6 testing methodology, whereas Section 7 describes experiences with both of these evaluation methods in IPv6-based MANET networks. Section 8 concludes the article.

2. Research Goal Driving the Methodology Selection

In the course of the authors research the extension to a key IPv6 stateless address autoconfiguration protocol has been proposed [19], [20] – the IPv6 Neighbor Discovery (ND) [21], [22]. The extension – ND++ – is aimed to address the needs of IPv6-based MANET networks and overcome the basic solution limitations, which cannot ensure proper configuration in mobile, ad hoc environments. The proposed ND++ solution has introduced several changes to the basic protocol design depicted in [21], [22]. They are described briefly below, since the character of changes to the common IPv6 stack influences the evaluation methodology. Figure 1 presents the modifications incorporated to the IPv6 packet at different levels. At the ICMP level ND++ uses new message types distinguished by the unique Code field in the ICMP header as well as new option types. There is also a new option proposed for the Hop-by-hop extension header at the IP header level. In addition to these changes, ND++ brings algorithmic modifications of the Neighbor Discovery protocol behavior.

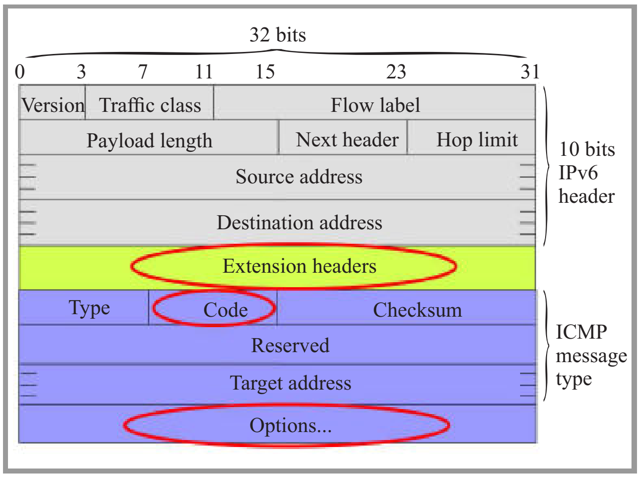

Fig. 1. IPv6 packet overview with marked fields, where changes were introduced by the ND++ protocol.

Considering the characteristic of the ND++ solution presented above, the protocol evaluation and its implementation as a part of an existing IPv6 stack requires code development at IP and ICMP level (mainly in the ND part, but also Extension Headers). Whereas very often dedicated APIs exist for the inclusion of new Extension Header options, changes within ICMP level together with algorithmic changes in the protocol behavior usually require modifications to the IPv6 stack directly. Especially in case of ND++, socket-level programming (used e.g. for many DHCPv6 modifications) cannot be used to implement the whole solution, since basic ND is too tightly coupled with the core IPv6 functionality and cannot be turned on/off or controlled externally. This makes implementation challenging, especially in case of testbed experimentation with the real kernel code modifications.
The proposed ND++ solution evaluation was aiming at improving protocol design and features on one hand as well as evaluating its scalability, performance, and behavior on the other one. To address these needs the protocol evaluation by means of both simulation and real-world testbed emulation was performed.

3. Experimentation Objectives and Requirements

Taking into consideration the particular demands towards experimental evaluation specified by the nature of ND++ research, the authors have identified their objectives and requirements. Thus, the following experimentation environment to be simulated/emulated is envisioned:

- MANET network, where the nodes create ad hoc topologies, there is no centralized server and the network is autoconfigured by means of stateless address autoconfiguration within one network domain; routing is not necessary (including the typical MANET routing protocols), since auto-configuration is performed before routing comes into play during network set-up;
- IPv6 only network – the authors were not interested in dual stack nodes, since ND++ is a purely IPv6 solution;
- Network size from few to hundreds of nodes – a number of nodes depends on the experiment goal; for most simulation scenarios high network node count would be envisioned, however, for testbed-based evaluation an environment with significant network size probably would not be accessible, therefore smaller number of nodes is assumed in this case;

- Possibility to create (simulate/emulate) different topologies – random and pre-selected ones, depending on the particular scenario under test;
- Node mobility can be emulated if necessary.

An exemplary scenario to be simulated and/or emulated, conforming to the environment specified above, is presented in Fig. 2. In this scenario a new node is joining MANET network and performs Duplicate Address Detection (DAD) [20]–[22] as a part of the ND++ based stateless configuration of its newly assigned IPv6 address.

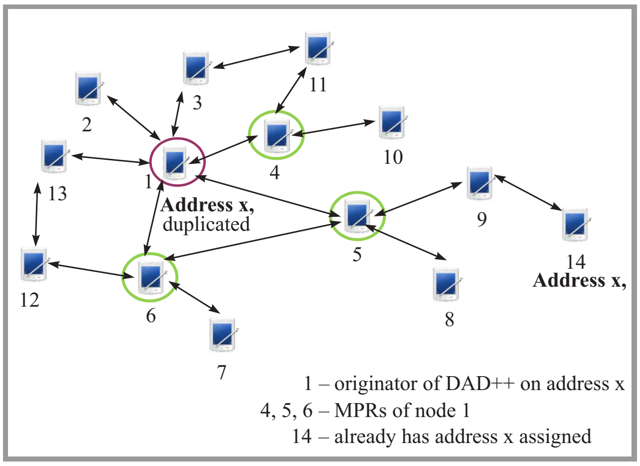

Fig. 2. Exemplary scenario for the ND++ protocol evaluation [20].

Based on the objectives and the research goal defined above a set of requirements towards the experimentation environment is driven. They are particularly specified for testing ND++ solution, however they are representative to most IPv6-based MANET experimentation. Hence, the desired IPv6 MANET test environment should:

- be open-source based;
- have an IPv6 protocol stack incorporated in each node – it must be possible to make modifications to the IPv6 stack implementation directly, since in many cases socket-level changes are not enough; moreover, the IPv6 implementation must be compliant with current Internet Engineering Task Force (IETF) RFCs [23];
- support 802.11 a/b/g network set-ups;
- support large network sizes up to hundreds of nodes (in case of simulators);
- be still actively developed (in case of simulators);
- be used by the research community – this feature ensures obtaining comparative results and their reliability (in case of simulators);
- incorporate visualization tools (not mandatory).

Some of the identified requirements, i.e. the network size, community support and utilization level, refer in fact to the simulation environments – for testbed-based experimentation usually the limitations are imposed with regard to these factors resulting from the availability and access to the hardware experimentation platforms.

4. Wireless Network Simulators for IPv6 Experiments

4.1. Overview

Based on the specified requirements, an overview of the wireless network simulators is presented below, that were investigated as candidate open-source environments for testing IPv6 in MANET networks. The selection contains popular and less-known simulators depicted in MANET-related research papers. Moreover, some simulators for Wireless Sensor Networks (WSNs) were considered as well, since the capabilities of some of them could also be of interest to MANET experimenters. Below the brief overview of the investigated network simulators and their main features is presented:

- NS-2 (Network Simulator 2) – it is a C++ based discrete event simulator [24], which used to be one of the most widely exploited simulators in MANET research. It targets TCP, routing and multicast protocols simulations in wireless and wired networks;
- NS-3 (Network Simulator 3) – it is presented as a discrete-event network simulator for Internet systems [7]. NS-3 is the successor of NS-2 that is gaining an increasing attention of MANET researchers. Similarly to NS-2, it is based on the C++ programming language, however Python API is also available;
- GloMoSim (Global Mobile Information System Simulation Library) – developed by the University of California, Los Angeles, USA (UCLA) [25] using Parsec programming language, became the basis for the commercial simulator QualNet [5]. It used to be popular among the MANET community several years ago;
- OMNET++ – it is an extensible, modular, component based simulation library with several side-projects complementing the core simulator framework [6], [26]. For MANET experimentation especially the INETMANET [6] and OppBSD [27] frameworks are interesting – the first one adds ad hoc functionality and protocols, the second one enables simulations with a FreeBSD operating system (OS) ported to each node, which may be useful in case of the need for an evaluation in a real-world OS. OMNET++ has a wide spectrum of functionality and its popularity has grown significantly in the recent years [6];
- SWANS (Scalable Wireless Ad hoc Network Simulator) – SWANS is a Java-based tool developed at the Cornell University, USA [28]. It leverages the Java

in Simulation Time (JiST) framework [29] to achieve high simulation throughput, good memory utilization and efficient signal propagation computation. Moreover, it allows to run standard Java network applications over simulated networks. Due to its particular concern on simulation performance in terms of resource usage, it enables to simulate large network sizes, exceeding those practically available in NS-2 or GloMoSim [28];

- GTNetS (Georgia Tech Network Simulator) – this tool [30] gathered some attention of MANET researchers several years ago. It became a basis for the MobiREAL project [31], which was proposing a realistic network simulator for MANET networks focusing on the accurate design of mobility models and patterns;
- Sinalgo (Simulator for Network Algorithms) – this software [32], developed by the ETH Zurich, is focused on message exchange, mobility management and topology set-up. These are crucial aspects of MANET networks, however the simulator does not consider inside-node logic and thus does not have TCP/IP stack implementation;
- WSN simulators – the simulators potentially applicable to MANET NETWORKS:
  - AlgoSenSim – framework for simulating distributed algorithms [33], similarly to Sinalgo, it is not protocol stack oriented but algorithm oriented. This framework focuses on network specific algorithms like localization, distributed routing, flooding;
  - NetTopo – designed to test and validate algorithms for WSNs [34], therefore it is algorithm oriented similarly to the previous one;
  - SENSE (Sensor Network Simulator and Emulator) – it has very limited module list, but is interesting due to its emulation capabilities. The simulator focuses mainly on routing in network layer [35];
  - TOSSIM (TinyOS Simulator) – probably one of the most complex WSN simulators. It targets a simulation of Tiny OS nodes and simulates entire TinyOS applications. It works by replacing components with simulation implementations.

4.2. Initial Evaluation

Many of the initially identified MANET simulators depicted above have been discarded at an early selection stage, since some of their features turned out to violate one of the key identified requirements. NS-2 simulator is not developed anymore and is practically superseded by NS-3, therefore the authors have focused on this one in the course of further evaluation. The GloMoSim project finished and the latest release is dated at the year 2000. Since this simulator became the basis for a commercial product, the open-source version has not been further developed since then. There is also a significantly large group of simulators that do not have IPv6 stack – SWANS, GTNetS, Sinalgo, most of the WSN simulators. This is either because it has not been developed so far (SWANS) or as a result of being algorithm oriented (inside node logic is not considered for Sinalgo and WSN simulators like NetTopo and AlgoSenSim). Moreover, SWANS is considered depreciated, since it has not been developed since 2004–2005. There was a SWANS++ effort proposed later on [36], however it reached an Alpha version only, which does not meet our requirements regarding simulator utilization by the community and being under active development. WSN simulators turned out not to be practically useful for simulating IP-level solutions for MANETs, since they are based on the protocol stacks specific for WSNs, not applicable to ad hoc networks. Moreover, they are very often limited to the simulations of only selected networking functionalities, e.g. routing. Interestingly, TOSSIM WSN simulator has its own 6LoWPAN-based IPv6 implementation called Berkeley Low-power IP stack (BLIP). However, as reported in [37], it is currently not completely standards compliant.

4.3. Final Simulator Selection

Having investigated the candidate simulators presented above versus the identified requirements, it turned out that practically only NS-3 and OMNET++ can be an interesting option for testing IPv6 in MANET networks. Hence, the final selection was made between the two of them. Table 1 presents the features of these simulators according to the key requirements.

Table 1
NS-3 and OMNET++ comparison vs. identified requirements

Requirement	NS-3	OMNET++
Can modify IPv6?	Yes	
Implementation up to date?	Yes, but not perfect	Yes, minor bugs
Support for 802.11a/b/g?	Yes	
Support for large network sizes?	Yes, MANET protocols available	
Actively developed?	Very active, support, constant bugfixes	Yes, increasing activity and importance
Other remarks	C++	C++, domain-specific functionality developed as separate projects

Both NS-3 and OMNET++ fulfill the requirements and are capable of making IPv6-based simulations in wireless networks, including MANET-specific protocols. They are currently under constant development with a large support community, which results in high level of their utilization in the research works performed nowadays [6], [7].

The differences influencing the final decision on the simulation environment selection reveal themselves while comparing more detailed simulators features. NS-3 seemed to have better, more bug-free IPv6 implementation. However, it is important to notice that both simulators are actively developed, so the implementations are constantly updated and IPv6 updates are of interest to both NS-3 and OMNET++ teams. Therefore IPv6 code is being improved with each release. There was also a lot of effort put by NS-3 developers to include wide range of accurate mobility and radio propagation models. This enabled to supply it with the capabilities that were often exposed as a weak part of its precursor – NS-2. It is worth noticing, that, apart from outdoor mobility and propagation models, NS-3 contains also an indoor models selection. For these it is possible to position the nodes within the building, specify for it the number of rooms, floors, material from which walls are made, etc. [7]. Moreover, NS-3 is very popular in the community and has a visualization support. As for the OMNET++, its main strength lays in the side-frameworks accompanying the core simulator environment. In the investigated case especially the INETMANET [6] and OppBSD [27] frameworks are of primary importance. INETMANET contains experimental features and protocols dedicated for MANET networks [7]. OppBSD enables to make simulations with FreeBSD ported to each node, which is interesting since FreeBSD is a good target environment for the implementations of IPv6 modifications in the existing kernel code. Unfortunately FreeBSD release ported to the framework is fixed and it is very hard to import the whole release with own modifications. This feature limits practical OppBSD usage, since it is very likely that the FreeBSD version to be used in the real system complementing the simulation work will be different and thus incompatible.

For the purpose of presented research for the evaluation of a new IPv6 solution in MANET networks finally the NS-3 simulator was selected. The most convincing was its strong IPv6 implementation, active developers community and high research community interest which maximizes obtained results credibility.

5. Topology Considerations

The selection of the network topology and mobility patterns properly reflecting the situation in MANET networks is a key aspect in the organization of both simulation and testbed-based experimental evaluation. This issue should be considered on two layers – first one is the position of nodes on the selected area and their interconnections, the second one is the mobility pattern that is applied to such a created scenario. In the mobile ad hoc network mobility influences the topology, so the two are constantly combined together. However, for the evaluation of MANET solutions it is useful to perform part of the evaluation in the static scenarios in order to be able to observe the solution properties before additional factors come into play. The authors will below give an overview of how MANET researchers usually approach topology considerations and propose the strategy that is aiming at maximizing experimentation credibility with regard to this aspect. Referring to the other works the article authors reflect those that present the core MANET protocols evaluation, like e.g. OLSR [38] or OSPF MANET extension [39], or the solutions of a similar nature to the one being investigated.

Topology considerations are treated differently depending on the experiment type. They usually are very limited in case of emulation and testbed evaluation and more sophisticated in case of simulations.

MANET researchers tend to set-up testbed-based experiments with topology generated by hand or from some a priori network settings. As an example, such an approach was applied also in the evaluation of OLSR routing protocol [13]. It enables to obtain proof-of-concept results type with very limited observations possible. Mobility is very often not considered, since in many set-ups (especially in the simple in-house laptop-based testbeds) it is hard to emulate it.

A common approach to the topology creation for the need of simulations is random nodes positioning on the area of a square, a disc or inside a 3D box. Another possibility, similar to the one used for the OSPF MANET evaluation [40], is to place the nodes on a square grid and introduce mobility pattern, e.g. Random Walk, for a specified period of time. The topology "screenshot" after a given time constitutes the node positioning for the experimentation. In the aforementioned OLSR evaluation [13], an automatic scenario generator was proposed to accompany the NS-2 core simulation environment. It allowed for the scenario parameters selection such as mobility, number of nodes, communication parameters, etc. to create a set of random, but in a sense also similar experiments, which could be averaged to obtain final results. With regard to mobility patterns, probably the most popular one for MANET networks is the Random Waypoint (RWP) mobility model. It was used in both OLSR and OSPF MANET evaluation case [13], [40], [41].

NS-3 has a wide range of both position allocators and mobility models available. They not only allow to create the most popular MANET scenarios, but also go a step beyond and create pre-defined indoor scenarios. This, accompanied with realistic radio propagation models, constitutes a powerful tool for MANET research. Moreover, NS-3 allows to provide own topology descriptions created by external topology generators and enables to port the output of the key widely known Internet topology generators – Inet [42], ORBIS [43], Rocketfuel [44] and BRITE [42] – directly to the simulation scripts.

For the extensive simulation experiments it is beneficial to include scenarios based on a realistic, close to real-world topologies. Unfortunately, the topologies generated by the random distributions on a pre-defined areas are not always conforming to the real MANET structures and require high node degree to assure full network connectivity [45]. Therefore several network topology generators can be considered to provide realistic MANET topologies. It would have seemed that the above mentioned Inet, ORBIS, Rocketfuel and BRITE are a good candidate generators. Despite their popularity, they are, however, not suitable for MANET research, since they reflect the Internet fixed network topologies, usually at the autonomous systems level. Moreover, they mostly represent hierarchical router-level topologies, which would correspond to boarder gateways/routers in MANET networks, whereas in MANET research the protocols under investigation usually depict interactions between non-hierarchical MANET nodes being hosts with router functionality inbuilt. An interesting topology generator is the NPART [45], [46]. Although targeted for Wireless Mesh Networks (WMNs), it is based on the assumptions [45] that remain valid also for MANETs and therefore can be applicable to this network type as well. NPART generates topologies similar in nature to WMNs deployed in Berlin and Leipzig. Moreover, it generates connected graph topologies while keeping node degree reasonably low, thus solving an important issue which may occur while using random node placement algorithms. Another solution is the Network Topology Generator (NTG) [47] developed as the module to SciLab open-source numerical computation software [48]. The tool allows not only to generate topologies but also provides a toolchain that enables routing-related analysis on the generated network and provides basic statistics. The authors particularly depict that NTG can be used for MANET simulations. However the tool is in fact oriented towards the design of fixed Internet topologies, whereas MANET simulations are enabled by means of random node distribution accompanied with RWP mobility model. This approach is in fact similar to the standard one described earlier in this section. As such among all the investigated topology generators the NPART tool seams to best fulfill the needs of MANET research.

In order to address the topology considerations in MANET experimentation in a proper manner the authors propose to perform experiments with two different groups of underlying topologies. The first one should be a set of deterministic topologies selected to reflect the particular kind of graphs recognized in graph theory or to expose some particular features of the investigated solution, e.g. circular topology, linear, grid or tree-based topology. The second group consists of random topologies generated in three different manners:

- as the random, most likely uniform, node distribution at a selected area (square, disc, rectangle),
- as the "snapshot" of a topology in the network initiated as a square grid with RWP mobility introduced for a specified amount of time,
- obtained by the network topology generator – the NPART is recommended.

Whereas deterministic topologies allow to verify protocol behavior in some particular, often demanding situations, random topologies are particularly recommended for the simulations aiming at scalability and performance testing with high number of nodes in the network. Generating random topologies by all three methodologies mentioned above ensures maximum accuracy of the obtained results.

For some experiments the network topology selection should be accompanied by the mobility model, which would reflect the MANET environment. Probably the most commonly used by MANET researchers is the aforementioned RWP model [13], [40], [41]. Therefore, this model as a basic scenario is used and recommended. However, it would be beneficial to include also indoor mobility models and more realistic mobility models where the nodes move along the streets, buildings, etc. Unfortunately practical usage of such models in many simulation/emulation environments may not be feasible. There are, however, interesting initiatives like e.g. the MobiREAL project [31], [49] which aims at modeling realistic mobility of humans and automobiles, but their results inclusion into other simulation environments may be in practice impossible.

6. Simulation versus Emulation

Although network simulators are powerful tools enabling simulation of IPv6-based MANET solutions in many diversified environments, there are also several limitations of simulation, which can be effectively addressed by the MANET emulators and testbeds.

Simulators offer the ability to create scenarios with hundreds of nodes in diversified, easily-created network setups. Recreation of such scenarios in the real-world testbeds and emulators is usually impossible, since in many cases the hardware constraints limit them to the few nodes with very limited mobility and/or topology emulation characteristics. However, even if in the emulation-based experiments topology and mobility patterns have to be created artificially and do not result from the real hardware interactions, emulators ensure node behavior closer to reality. This manifests itself especially in the IPv6 experimentation, where emulators based on the real kernel implementations of the IPv6 stack offer better, more bug-free networking stack implementation. At the opposite side the weak point of the currently available simulation platforms is that their IPv6 implementation is not tested thoroughly. As such an unsure behavior may occur, whose underlying source can be hard to identify. This holds true not only for the IPv6 stack implementation, but also for the other elements implementation of the communication stack. The authors will give examples of such a situation in the next section. For emulators, where the real hardware is being used, almost 100% accuracy of bug-free implementation of Wi-Fi drivers and networking protocols can be ensured.

Hence, simulation-based methodology is best suited towards performance and scalability testing, whereas testbed-based experimentation offers not only final verification of the solution in the real-world conditions and a proof-of-concept but also behavior and protocol design evaluation capabilities. These aspects especially reveal themselves in case of testing IPv6 solutions. Therefore, it may be beneficial to exploit emulation-based methodology not only at the testing process final stage, but also at the initial stage, where the emulation environment can help to come up with a well-designed IPv6 solution, capable of operating in the real kernel and integrating with a real hardware. At the next stage such a solution can be evaluated for its performance and scalability – simulation would best address the experiments needs.

Comparing the simulation- and emulation-based experimentation capabilities for IPv6 MANET networks, a testing approach is proposed that would allow for a detailed and accurate evaluation. The goal was to focus on the open-source, easily accessible and possibly low-cost methodologies. Therefore, the usage of both simulation and testbed-based (emulation) techniques interchangeably during the protocol design and evaluation process would be proposed. For the simulation environment an open-source solutions of NS-3 or OMNET++ would be recommended, whereas for the testbed the authors would propose to use possibly high number of machines governed from the central server controlling the experiments and allowing proper topology and mobility emulation. One possible approach to building such a server is depicted in [20] and revealed shortly in Subsection 7.2. The number of minimum 8–10 nodes should allow for obtaining reasonable and quite representative MANET topologies. The more nodes can be afforded, the more diversified results can be achieved. The nodes can run Linux or FreeBSD OS and can be low-cost, even diskless machines equipped with wireless cards for Wi-Fi network set-up. The authors propose to build initial model of the solution in the simulator environment. After tuning the protocol design based on the results obtained with carefully selected topologies and mobility models, following the approach presented in Section 5, testbed trials can be performed which would allow to expose protocol design features that should be improved and would constitute a proof-of-concept. At the last stage the authors would recommend to use the simulation environment with the updated solution description in order to perform scalability testing. In an ideal case it would ease the IPv6 solution development process, if the simulations could be performed by means of virtual machines with real OS representing simulated nodes – such an approach would allow to have only one solution implementation. However, although theoretically possible, such methodology could have practical limitations resulting from difficulties in porting OS versions to the simulation environment.

7. IPv6-based MANET Experimentation – the Authors Experience

In this section the authors will depict their experiences with both – simulation and emulation experimentation of the IPv6 ND++ protocol extension, which was proposed by them for MANET networks. Simulation experiments were conducted in the NS-3 simulator, release 3.11. Emulation experiments were performed on the dedicated hardware testbed platform comprising of 15 nodes based on the FreeBSD operating system in the emulated wireless ad hoc environment.

7.1. Simulating IPv6 with NS-3

Simulations were aiming at the performance and scalability evaluation of the proposed ND++ MANET solution. The implementation in NS-3 environment covered not only the simulation control scripts, but also required to modify IPv6 Neighbor Discovery code of the simulator's IPv6 engine. Therefore a lot of work has been performed with the IPv6 C++ code. In general the implementation was of good quality, as have been expected. However, the authors have noticed that the IETF RFCs, defining IPv6 and its related protocols, were not reflected in detail. Very often basic functionality was ensured, which enabled proper node interactions in most of the standard cases, but more sophisticated features were not implemented or were resolved in a simplified manner. Moreover, throughout the code there were parts of functionality left blank and marked as "to do's" for later NS-3 releases. However, the biggest difference in comparison to the FreeBSD IPv6 networking implementation in the OS kernel was the lack of packet consistency checks. Some packets would have been discarded by the real-world kernel as not conformable to the rules specified in the RFCs, whereas in NS-3 it would be possible that they would have been further processed. Moreover, the researchers have observed that for some issues "own", simplified, solutions have been implemented instead of a very detailed RFC specification. An example of such an approach is the IPv6 address selection procedure in NS-3 release 3.11. However, it has to be underlined that NS-3 is still under active development and in the most current releases IPv6 implementation is being constantly improved. Comparing NS-3 3.11 with the latest release (3.18) several bugs identified previously in the IPv6 implementation were already fixed.

In the performed experiments the NS-3 IPv6-only MANET network was based on the 802.11a/b/g wireless access with nodes creating different topologies (Fig. 3) – pre-defined grid topologies (with and without diagonal connections) and several random node distributions. The investigated scenarios were reflecting a situation, when a new node is joining MANET network and performs ND++ DAD procedure [19], [20] in order to obtain a valid IPv6 address (Fig. 2). In such scenarios some issues with Wi-Fi modules and random numbers at the simulator side affecting simulation experiments were observed. Even for the very basic scenarios, with a grid topology without mobility and small network sizes of less then 30 nodes, the results obtained for different random number seeds were in some cases very distinctive, at the level exceeding expected deviations. After turning off all randomness in proposed solution the issue remained the same and a detailed investigation has shown that the problem is related to the channel access in 802.11-based MAC layer. In the latest release random numbers engine has been significantly improved and underwent major revision, therefore the authors hope that this problem will be solved. Since the experimentation with NS-3 is still a work in progress, the researchers will investigate it further in the course of their future research.

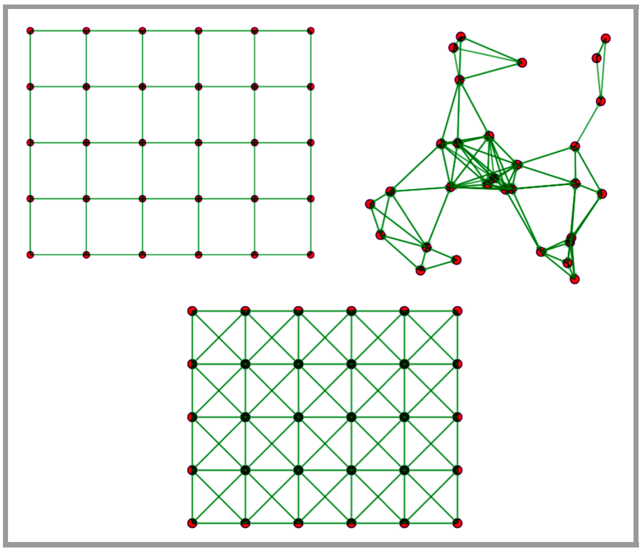

Fig. 3. Selected network topologies used in the NS-3 simulation experiments of ND++.

A consideration of experiences with NS-3 leads to a conclusion that it is more oriented towards TCP-level simulations and one could expect less bugs and issues with such experiments. However, the IPv6 implementation is in general good and would perform well in most of the use cases. Moreover, it is of high community interest to constantly improve the NS-3 simulator, hence the authors would recommend it for IPv6 MANET simulations. An additional benefit of this simulator is also the incorporated tracing and data collection mechanism, which eases processing of diversified result types.

7.2. IPv6 Solution Evaluation on the Testbed Platform

The goal of the testbed-based experimentation was not only to provide proof-of-concept implementation, but also to evaluate the behavior and algorithmic design of the proposed solution. The testbed is a dedicated solution enabling a 15-node MANET network emulation [20]. Its structure is depicted in Fig. 4. Fifteen hardware components connected to the server station and are acting as diskless workstations, which run the kernel version obtained from the server. MANET network is emulated on the Wi-Fi interface of each node by means of an IP firewall (IPFW) packet filtering.

The testbed is based on the FreeBSD operating system with kernel version 7.0. FreeBSD was chosen since it has open IPv6 kernel code implementation (very similar to the one from Linux) with a detailed description of IPv6 networking stack available [50]. One disadvantage of FreeBSD is, however, that updates to the system or its parts are usually not fully compliant among different versions, which causes dependency problems being often hard to handle.

The ND++ solution was implemented as direct modifications to the kernel code. The FreeBSD kernel contains very detailed IPv6 networking stack implementation. As a part of the operating system, it is thoroughly tested and corresponds to RFCs one-to-one with very detailed packet consistency checks and handling all necessary details. This feature is very important while testing protocol behaviors and designs, because it can be assumed that the probability of significant bugs influencing IPv6 protocols behavior is close to zero.

During experimentation none Wi-Fi-level issues have been observed, even though packet-level filtering was used (IPFW rules). At the physical layer all nodes were having direct connection with each other – as such the number of transmissions handled by the Wi-Fi cards in reality was much higher than it was seen at the IPv6 level after filtering. These difficult conditions did not result in problems similar to the ones observed in the simulation environment, which was not that demanding from the physical and MAC layer perspective.

Due to the nature of the ND++ solution, it required modifications to the kernel code directly. Unfortunately socket level mechanism was not enough in this case, however, the ability to use socket-level API would significantly improve the deployment time. When modifying kernel directly kernel recompilation is necessary after each change, which is a very time consuming task. The researchers have managed to reduce recompilation time from over an hour to about 10 minutes, however it still does not ensure comfortable programming. Moreover, bugs not detected during compilation time usually manifest themselves as the kernel panic in a working system. This not only makes debugging process difficult, but also can be hard to handle when the entire operating system crashes. However, the diskless workstations concept introduced in the testbed allowed to deal with such situations easily. Possibly kernel modifications could also be introduced as the kernel modules, which would enable to diminish the recompilation issues.

Although these implementation issues make the code development process difficult, the biggest advantage of the testbed experimentation is that once the solution is implemented and tested the authors can be almost 100% sure about the results and their performance in the real system. Also in the contrary – at the development stage, if something is not working it is almost surely the problem with the modifications, not the kernel implementation. With simulator there is always the risk that some of the models at the simulator core (e.g. Wi-Fi, physical layer, propagation models) were having bugs which may have affected the final results.

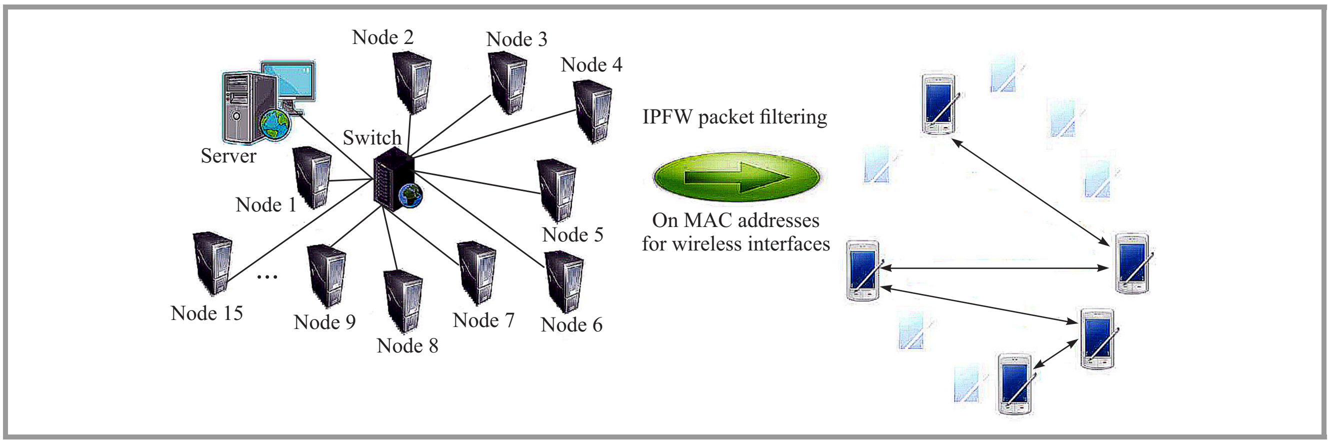

Fig. 4. Overview of the testbed platform set-up [20].

8. Conclusions

With the growing demand towards unsupervised network configuration and operation in the IPv6-based Internet of Things the thorough network testing importance increases. During the course of the authors research the techniques that would provide best test suite for the evaluation of the new IPv6-based solutions in MANET networks were investigated. The findings have shown that relying on simulation or emulation techniques only is not always enough – the variety of complementary techniques is necessary in order to be able to perform both performance and behavioral evaluation. Especially for MANETs, simulation should be complemented by emulation/real-field testing, due to the changing characteristics of such networks and difficulties in reproducing realistic mobility models and network topologies in the artificial environment. The open-source simulation environments investigations reveal, that among quite many MANET simulators only a few can really support IPv6 experimentation. NS-3 simulator best fulfils identified criteria and is probably the most developed and the biggest from the available open-source simulators. This builds its reliability and credibility reflected in the growing attention of MANET researchers. The experiences, though, have shown both the advantages and drawbacks of using this simulator as well as the testbed platform dedicated to IPv6 MANET experiments. As such the authors would formulate the conclusion that simulation and emulation are complementary evaluation methodologies. Therefore, in the proposed testing methodology, accurate testing of IPv6 MANET solutions should exploit both these techniques to maximize credibility and accuracy of the obtained results.

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Paper

Review of Simulators for Wireless Mesh Networks

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Abstract—The research of computer networks construction of models that reflect the current audited environment to carry out practical research is extremely difficult and often involves significant costs. Hence, the popularity of simulation tools that help developers to determine as early as at the stage of the simulation whether a given solution can be deployed in a real network. However, over time many different simulation tools have been developed, each with different characteristics, different uses, different strengths and weaknesses. It is the task of the researcher then to select, before starting the actual research, one of the available simulators in accordance with the needs and adopted criteria of evaluation. In the article the authors present issues related to the simulation tools and the main advantages of simulation as well as their drawbacks. To help researchers select an appropriate simulation environment, the authors present statistical information gathered during a literature survey of a number of research articles from the most popular publishers in which the selected simulators were used in initial system design.

Keywords—simulators, wireless mesh networks.

1. Introduction

Wireless Mesh Networks (WMNs) is one of the key technologies for current Wireless Networks. It has become more popular over the past years. WMN can be an answer for the last mile problem, a solution for rugged terrains, developing regions and countries [1]. The networks equally behave well in education, hospitality management, car industries, etc. [2]. They are also promising technologies for military forces to quickly build wireless networks over, for example, the battlefield. Just as in the case of any computer networks, also in the case of WMN we need to develop routing metrics and protocols which would suit all needs of potential applications [3], [4]. Every such solution, before it can be used in real environment, should be checked by researchers in some test environments. It is not trivial to choose a proper tool for testing and simulating different network behavior [5], therefore in this article the authors attempt to facilitate a choice of them.

The paper is organized as follows. In Section 2, main methods used in the evaluation process of routing protocols and metrics and main advantages and drawbacks of using simulators are presented. Section 3 surveys network simulators indicating their advantages and disadvantages. In Section 4, the authors present the results of a survey based on published articles carried out in the area of WMN network and aim to determine the popularity of individual simulators. Section 5 concludes the paper containing both an attempt to answer the question of what criteria to follow when choosing a simulation environment.

2. Simulation Tools

2.1. Main Evaluation Methods of Routing Protocols

Typically, the development process is divided into two phases: the evaluation by means of quality tools and the subsequent prototype testing in a close-to-real environment test beds.

In the case of Wireless Mesh Networks, as compared to traditional wireless networks, there is an additional challenge, due to the structure of network, stationary nodes roles as well as clients mobility and roles.

It is also worthwhile to mention that there are some specific characteristics of Wireless Mesh Networks that provide additional conditions for simulating them, such as [6]:

- wireless – what implicit limited transmission rates and high loss rate;
- multi-hop – means that traffic is forwarded through nodes that are not in direct range of the node that generates it;
- redundancy – the nature of WMN implies redundant links in the wireless backbone of network;
- mobility – while backbone nodes are mostly stationary, clients of the network should be treated in simulation models as mobile;
- dynamics – because of the self-configuring and self-healing ability of WMN one should consider smooth changes in the structure of the network; the network is established in a very spontaneous way;
- infrastructure – dual type of nodes in network should be considered - mobile clients versus stationary nodes;
- integration – the duality of structure also in roles that nodes play in network - lightweight clients can join the WMN network without serving any routing services.

Researchers that need to evaluate a routing protocol or routing metric for WMN have to choose a evaluation model. We can choose from different types of evaluation processes [6], [7]:

- **Theoretical analysis** – in that process a mathematical models to evaluate network performance is used. The most commonly used mechanism is queuing theory. It is a very difficult means of development, mathematical formulas can get very complex and, thus, can consume a large amount of time. What is more, there are no dedicated mathematical tools to provide such analyses. Nevertheless, a mathematical analysis is often the first step of the development process.

- **Simulations** – with special tools the researcher is capable of modeling virtual environment to help verify the general idea, detailed parameters and solutions, or to compare proposed solutions. Simulations are particularly useful for studying highly distributed networks such as Wireless Mesh Networks or Wireless Sensor Networks. In this way one can discover behavior in such networks under a change in some parameters, while others remain fixed. Additionally, simulation base studies are very flexible with low cost.

- **Emulation** – it is a hybrid study environment that consists of two parts - real and simulated. It depends on the researcher's goal which element is real and which simulated. Emulation has one important advantage – any results from such tests are more realistic as any experiment part because it is a real working part.

- **Virtualization** – general idea of virtualization is to provide virtual environment in which hosts to conduct experiments are run. Nowadays, virtualization is becoming quite simple and inexpensive, so it becomes more and more widely used. It is actually rather easier to use existing hosts and install virtual hosts on it than to build a quite new infrastructure that consists of many physical machines. It can vary to what degree virtualization can be used – it can be full with virtual hosts, virtual operating systems and all network equipment or as virtual instances or virtualized only as a part (for example only client hosts). Virtualization can offer good tools for evaluating communication protocols – it is possible to provide multiple virtual hosts on a single physical machine, thus the experiments cost can be minimized.

- **Real test-beds** – it is a development process based on a prototype implementation that should produce the most realistic results. By using it, the researcher can simply transfer their ideas to the real world, though the influence from environment should be also considered as it can significantly affect conducted experiments.

2.2. Advantages and Drawbacks of the Use of Simulators

Using simulation tools for conducting processes of testing network routing protocols, or any other researcher's ideas connected with networks, has many advantages. The two most important are the low cost of the whole process and the ease of maintaining of a simulation [8].

In the case of testing a new idea, there is almost always a need for rebuilding a number of modules, redesigning the model, etc. While using a real test-bed or a prototype that is part of the process can be expensive both in financial terms and the time involved. Simulations take the building/rebuilding phase out of the loop by using the model already created in the design phase. Most of the time, the simulation is cheaper and faster than performing multiple tests of the design each time in real test-bed [5].

The other important advantage is the ease of maintaining a simulation. A simulation can be repeated as often as it is needed with repeatable results probability close to certainty. Additionally, researchers have full control over the simulation process at any of its stage. A scenario preparation for simulation purposes is easy to create and collecting of results is also easy to perform.

There are also simulations disadvantages as well [9]. It is worth mentioning lack of existing standards in that area (i.e., no standardized tools which would generate results in a way that would be easy to compare with others), such a dependence results from the implementation of simulation tools or the fact that instead of the actual physical layer there is an abstract software layer, which can lead to differences between simulation results and those obtained in real world.

Table 1 presents the main advantages and drawbacks of the use of simulation tools.

Table 1
Advantages and drawbacks of the use of simulators

Advantages
• Easy to expand network topologies due to simulation applications high scalability • Simulation process is easy to maintain • It is the most common way of developing and testing new routing protocols • Testing cost relatively small • Results have high repeatability • Full control of simulation process • Easy process of scenario preparing and data collecting
Drawbacks
• There is no standardized simulation tool that would allow to compare simulation results between different projects • Results can differ from real world because of abstracted PHY layer modeling • Results can depend on a particular implementation of simulation software

3. Simulation Tools for WMN

3.1. Description of Selected Simulation Tools for WMN

Network Simulator 2 (NS-2) is an open source, discrete-event network simulator [10] that provides support for a simulation of main protocols, routing, multicast protocols for wired and wireless networks. NS-2 is the most popular simulator tool among researchers and becomes de-facto a standard for simulators. It was developed in 1989 as a variant of REAL Network Simulator. Based on C++ and OTcl, NS-2 was continuously developed till the end of 2011. The simulation environment can be run on a number of operating systems, i.e., Linux, Windows, OS X, Solaris, etc. As a module for supporting Wireless Mesh Networks, there is a library called WiMsh, additionally there is also a framework for NS-2, called Multi-routing-protocol Simulating Framework, proposed by researchers from Southeast University of China in which WMN is also adopted. NS-2 provides support for OSI Layers except the presentation and session layers. The simulator has a complex structure that makes writing new modules a hard task for researchers because it requires a simulator good knowledge. Additionally, because of the application of two different languages (C++ and object oriented OTcl), a creation of even a simple scenario can become a complicated job. Thanks to the NS2 users community, there are many additional resources like modules for specific scenarios, topology generators, or GUI tools. The most popular tools are: topology generators (i.e., Inet Topology Generator [11], GT-ITM [12], or Tiers Topology Generator [13]), tools to visualise results of simulations (i.e. Nam [14], Fig. 1), or applications for graphic creation of NS-2 scripts – Extended NamEditor [15]. The user interface implemented in NS-2 is based on the command line tool and operation on source files. There is no integrated GUI, but as an addition one can use many tools proposed by the large community that provides such a functionality, for example, Visual Network Simulator (Fig. 2).

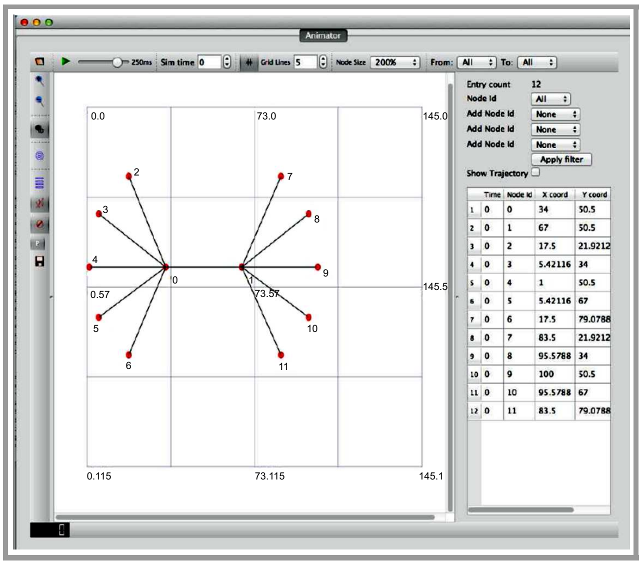

Fig. 1. NS-2 NetAnimator example.

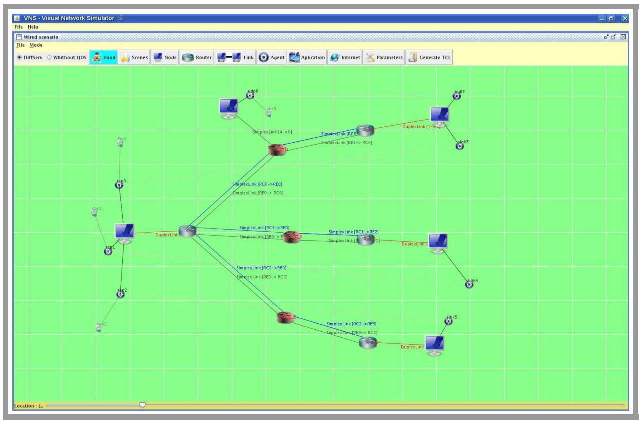

Fig. 2. Visual Network Simulator.

Network Simulator 3 (NS-3) is also an open source (licensed under GNU GPL), discrete-event network simulator, released in 2006 [16] that is still under development. NS-3 should be rather considered as a replacement than an extension to the previously described NS-2 [17]. The simulation environment is based on C++ and Phyton and can run under most of modern operating systems. NS-3 has a possibility to generate pcap traces of simulated models, so researchers can easily debug output with standard tools such a Wireshark [18]. NS-3 includes a radio energy model for simulating energy consumption and has a set of classes for simulating 802.11s mesh networks [19]. Additionally, there is also a number of external tools provided by the NS-3 community. The user interface for NS-3 is command line based, but there are also some additional tools for NS-3

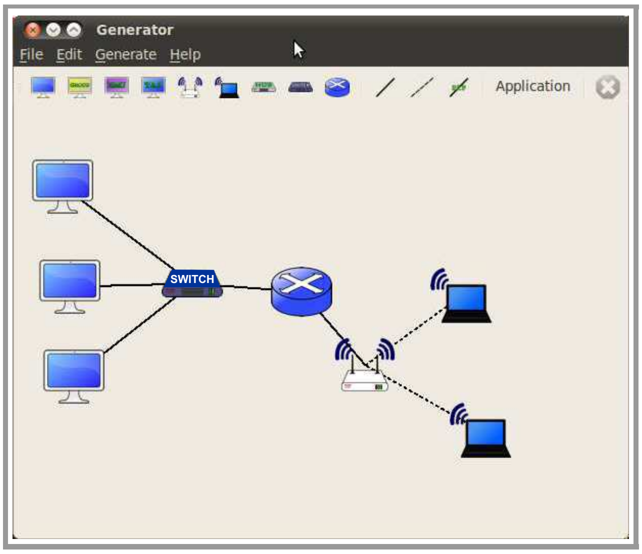

Fig. 3. Example of NS3 GUI.

available that provide GUI (Fig. 3), such as NetAnim [15] for tracing results of tests or PyViz [20] (Fig. 4) for live simulation visualization.

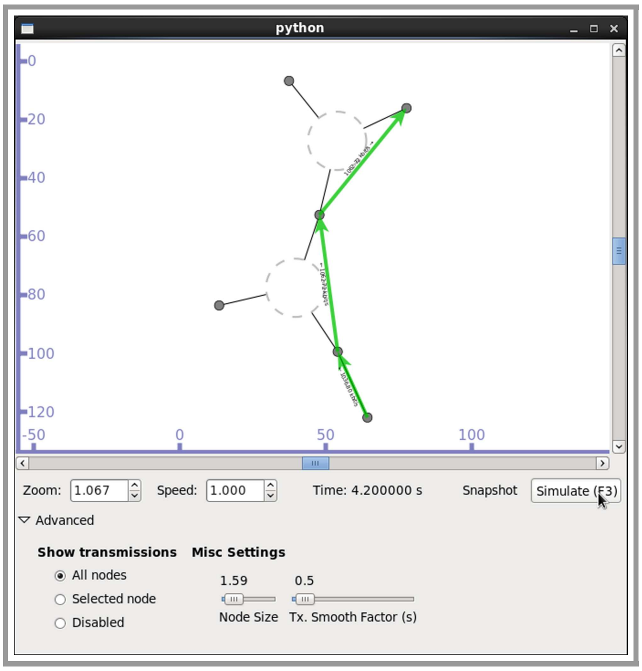

Fig. 4. PyViz application example.

OMNET++ is a component based, modular, open architecture event simulator [21], licensed under the modified GNU Public License called Academic Public License, which means that it is free to use for educational and non-commerce purposes. The interface of OMNET++ is based on C++ equipped with a GUI based on the Eclipse environ-

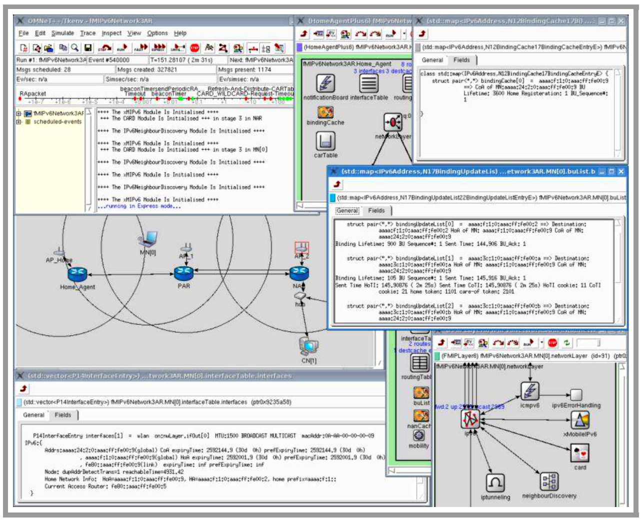

Fig. 5. Example of OMNET GUI.

ment (Fig. 5). The simulator runs on Linux, MacOS and Windows systems. It supports the simulating process of Wireless Mesh Networks – there is a library named Virtual Mesh [22] that can also be used as an emulation framework. Additionally, for simulating mobile and fixed wireless networks, there has been developed a whole modeling framework called MiXiM [23]. Many tools and utilities have been developed by OMNET programmers and community members to maintain simulations results or scenario generation. There are also available extensions for writing code in different programming languages – for example in Java, most of them are available to download for free from OMNET++ Web pages. The documentation and community forum provides a good level of support, which makes this tool quite user-friendly even for beginners.

OPNET is a commercial discrete-event simulator, first proposed in 1986 and developed by MIT in 1987 [24]. The interface of OPNET is C++ based. The dual-purpose simulator provides an environment for: designing protocols and testing scenarios in realistic environments. OPNET Modeler can use topologies created manually as well as those imported or selected from the pool of predefined ones. There is a vast number of protocol models available in the program suite. For a wireless networks simulation, OMNET uses an extension called OPNET Modeler Wireless Suite [25]. Modeler has an advanced GUI interface used for creating models, simulation execution and data

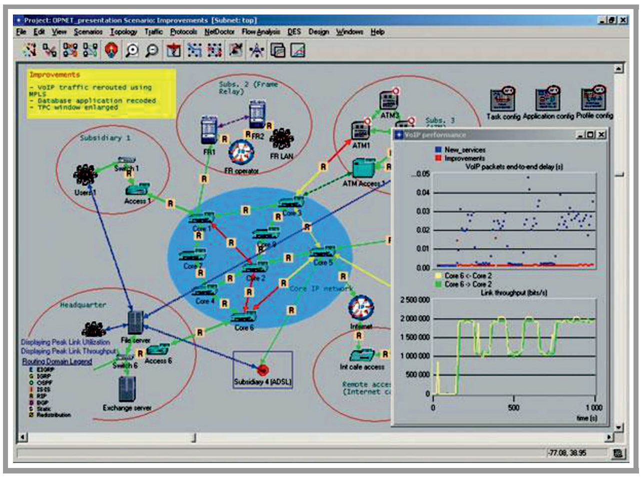

Fig. 6. OPNET GUI example.

analysis (Fig. 6). OPNET Modeler provides a good manual, there is also a dedicated technical support for a commercial use of simulator. There are also specialized training sessions provided my the manufacturer to help you use the software. The system can be run under Windows or Linux.

Global Mobile Information System Simulator (GloMoSim) is a simulator framework used for large scale wireless networks [26], it provides a functionality similar to QualNet that is however truncated from some additional features. There is no GUI provided by GloMoSim and the available documentation is not so in-depth. GloMoSim is also a C++ based simulator (with Parsec for maintaining parallel operations), distributed under Open Source License. Support for Wireless Mesh Networks is not in such a wide range as it is in the case of QualNet, though the simulator is capable of simulating networks that contain thousands of nodes. The simulator offers a possibility to install external GUI tools for the visualiza-

Table 2
Comparison of network simulation tools

	NS2	NS3	OPNET	OMNET++	QualNet	GloMoSim	JSim
Interface	C++/OTcl	C++/Python	C/C++	C++	Parsec	Parsec (C)	Java
Graphical Support	No	Limited	Yes	Yes	Yes	Limited	Yes
Parallelism	No	Yes	Yes	Yes	Yes	Yes	Yes
Scalability	Small	Large	Medium	Large	Very large	Large	Small
Documentation and user support	Excellent	Excellent	Excellent	Good	Good	Poor	Poor
Extendibility	Excellent	Excellent	Excellent	Excellent	Excellent	Excellent	Excellent
Emulation	Limited	Yes	Not direct	Limited	Yes	Not direct	Yes

tion of results, but no tool for designing tests. Despite these drawbacks, GloMoSim is quite a popular tool among researchers.

QualNet is a network simulator developed by Scalable Network Technologies [27]. In fact, it is GloMoSim simulator commercial version. It is a powerful, ultra high fidelity network simulator. Its interface is based on C++. Almost all functions of that tool are available from the GUI level, which makes learning the application process quite easy and lets the researcher focus on simulation process goals (Fig. 7). QualNet also has the support for Wireless Mesh Networks, there are some external models for these type of networks available for download. QualNet runs under Linux and on Windows.

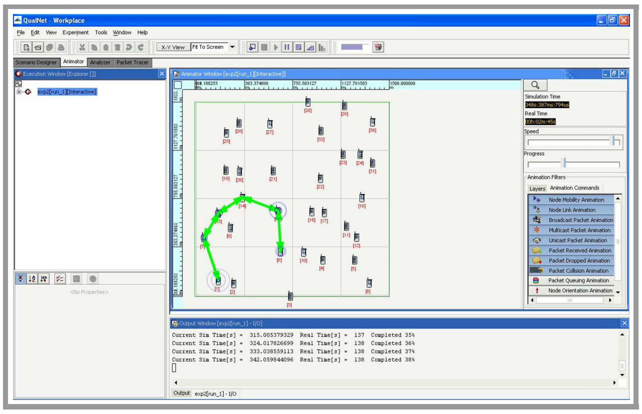

Fig. 7. Example of QualNet GUI.

J-Sim, formerly known as JavaSim, is a Java based simulation system for building and analysing numeric models [28]. It is distributed under an OpenSource License and can be installed either on Linux or Windows machines. With the external tool called gEditor (Fig. 8) the user is provided with the Graphical User Interface that makes it very easy to build network models and conduct simulations. For a wireless networks simulation, J-Sim has a wireless extension but support for Wireless Mesh Networks is not available from the developers of J-Sim, though can be found in the community resources through additional

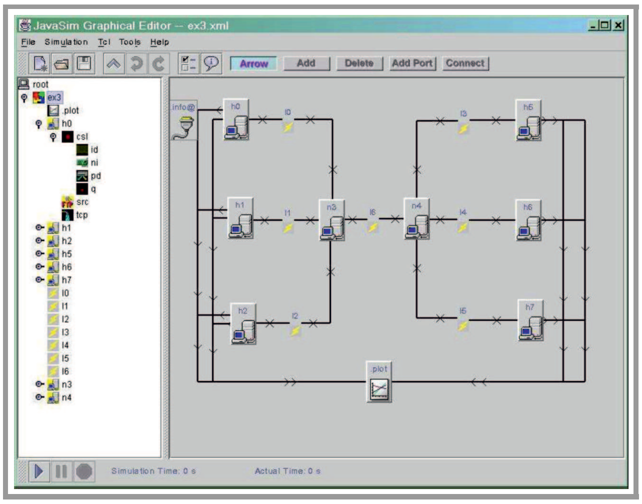

Fig. 8. J-Sim gEditor application example.

libraries. The simulator has a good documentation with examples for some small scenarios though the details level is quite low.

3.2. A Network Simulation Tools Comparison

It is obvious that there is no universal simulation tool that would fit all needs. It is also true that modeling everything in a complete simulation mode is simply unattainable. There are different features that distinguish a particular simulation software for specific applications (Table 2). There are many literature studies that compare different aspects of the simulator software, i.e. [29], [30], but at the end it is the researcher's choice which one they would use.

3.3. Main Criteria for the Simulation Tools Selection

There is a number of criteria that are used to determine whether or not a particular simulation tool is to be selected and is most appropriate for a given purpose. It can distinguish the following exemplary criteria:

- general capabilities: flexibility, available models, reusability, devoted to specific problem or class of problems, orders of magnitude for simulation size;

Table 3
The use of selected simulators in publications related to wireless networks

	2000–2013											
	IEEE			Springer			Wiley			Elsevier		
	Mesh	Ad-hoc	Sensor	Mesh	Ad-hoc	Sensor	Mesh	Ad-hoc	Sensor	Mesh	Ad-hoc	Sensor
ns-2	135	307	300	76	579	128	14	70	65	105	716	382
ns-3	15	16	9	7	23	4	4	1	2	8	50	34
OPNET	43	103	86	36	122	61	9	21	10	56	186	128
OMNET++	4	1	8	17	53	48	7	5	18	15	82	123
QualNet	17	50	32	5	68	16	6	17	6	17	103	42
GloMoSim	5	46	13	1	115	9	1	16	10	7	125	30
j-sim	2	6	14	0	4	18	0	0	7	1	9	11

- hardware/software considerations, i.e., operating system, compilers, specific hardware needs;
- graphical facilities;
- statistical features;
- ease of use, documentation, support;
- output reports and plots;
- popularity.

4. Popularity of Simulation Tools

Choosing a simulation tool that would meet a project's design and functional requirements can be a tough work. It is also the issue which the authors have to solve. To form an opinion on the available simulators popularity, the authors conducted a literature survey of simulators most frequently used by researchers. The survey was based on all articles available through search engines of IEEE, Elsevier, Wiley and Springer publishing houses. The authors searched for articles published between the years 2000–2013 in the field of Wireless Sensor Networks (WSN), Wireless Ad-hoc Networks and Wireless Mesh Networks (Table 3).

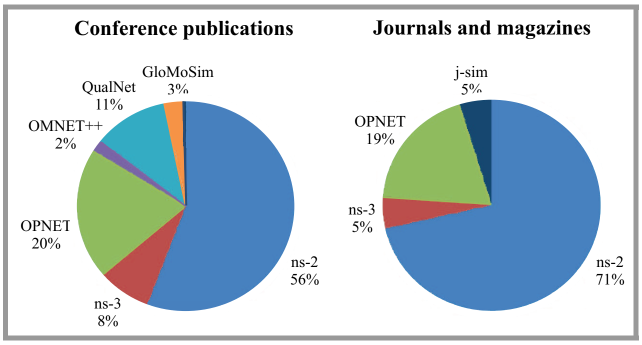

Fig. 9. Popularity of WMN simulators by IEEE.

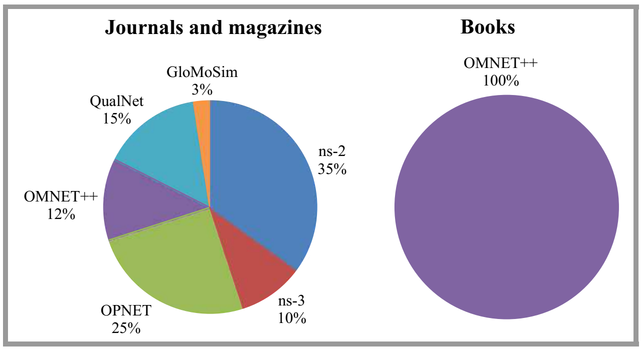

Fig. 10. Popularity of WMN simulators by Wiley.

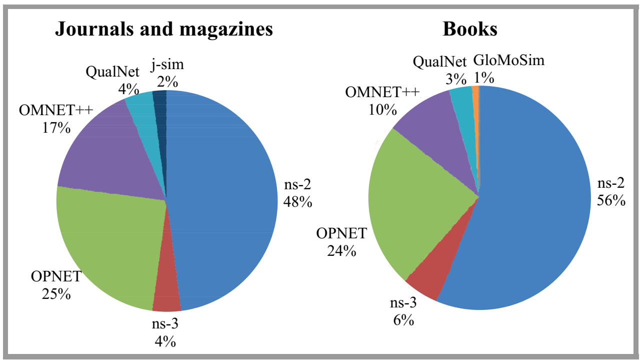

Fig. 11. Popularity of WMN simulators by Springer.

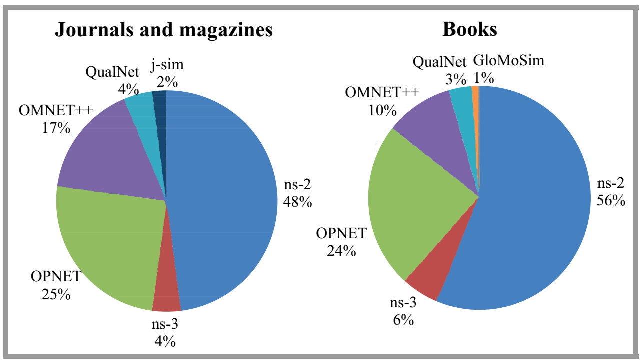

Fig. 12. Popularity of WMN simulators by Elsevier.

The results analysis show that NS-2 is still the most popular simulation tool. Further, in terms of popularity, are OPNET, GloMoSim, QualNet, and OMNeT ++. Figures 9–12 present analysis results.

5. Conclusions

In this article the authors provide a survey of the most popular network simulators: NS-2, NS-3, OMNET++, OPNET, GloMoSim, QualNet and J-Sim. The main properties of these tools are presented followed by a simple comparison. The authors also include some background information on the popularity of particular simulation tools in the form of a survey based on available sources of articles from a number of leading publishers (IEEE, Springer, Wiley and Elsevier). The survey results are also presented in the article.

Based on the conducted investigations it can be concluded that for academic researchers the best choice will be NS-3. It provides suitable libraries to address the particular applications needs. Another good choice is OMNET++, since it is also free for academic use, its users community is quite large, it is popular and one can find many additional modules or libraries for specific needs. In authors' opinion the latter choice is better for researchers that do not want to put too much effort into learning the application as it is more intuitive and, thanks to the well-designed GUI, easier to use. In terms of popularity, NS-2 appears, however, to be the best choice. Since the software is no longer being developed, the authors believe that its replacement – NS-3 – will be soon as popular as NS-2. OPNET Modeler and QualNet are also very good choices but since they are commercial applications they will not be as easily accessible to every researcher. Another good choice is GloMoSim, especially when one considers large scale networks, though, because of the lack of support and documentation, this simulator is still not as popular as those discussed above.

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Paper

Priority Based Routing for Forest Fire Monitoring in Wireless Sensor Network

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Abstract—Recently, forest fire monitoring system in wireless sensor networks has received much attention. The conventional scheme receives fire alert data quickly to inform about fire forest event. However, since two or more nodes may detect a fire, high priority fire detection data frequently collide. In this paper, a new forest fire monitoring system is proposed in order to reduce high priority fire detection data dropped rate, by specifying a high priority received data immediately after fire detection and just before the destruction by fire. Furthermore, the node only transmits high priority data to a node, which has a low possibility of destruction by fire for low end-to-end delay of high priority fire detection data. The simulation results show that proposed scheme can reduce high priority data dropped ratio and the end-to-end delay, and have less effect of wind direction compared with the conventional scheme.

Keywords—event detection, forest fire, priority, routing, wind.

1. Introduction

The forest fire is a serious problem in the world: it is reported that as many as 66,343 wildfires occurred and burned 6,319,586 acres in the USA in 2013 [1]. Currently a satellite-based monitoring is a popular method to detect forest fire [2]. A satellite sends short or medium infrared wavelength images with 500 m resolution per day. These images are analyzed of reflectance and brightness corresponding to burning and non-burning pixels. However, because of long scan period and poor resolution the initial phase of fire forest can be missed [3]. Although there are other fire forest detection schemes, e.g., using a digital camera [4], a long-wave infrared camera [5] or sensors with four propellers [6], they cannot be used in a large areas due to the high cost.

Nowadays, it is expected to use Wireless Sensor Networks (WSNs) for the forest fire event detection by periodically sensing the temperature, humidity and light in whole forest area. In WSN, the sink node (data collector) gathers information from many sensor nodes [7]. They are expected to work with limited energy for a long time period because they are small and lightweight. After obtaining environmental data, sensors apply the processing algorithm such as neural network to detect and forecast fire [8]. In forest fire monitoring, the fact that nodes might burn down when the fire breaks out have to be considered. Although there are many routing protocols, e.g., Leach [9], PEGASIS [10], Teen [11], PEQ [12], none of them considers the case when some nodes are burned down. As a consequence of the fire event, the path between sensor nodes and data collector may be unavailable. In order to overcome this path failure, the unrecoverable path to the data collector causes unnecessary delay. It is also necessary to the utilize power energy from nodes which will be destroyed by fire. Ansar *et al.* propose Maximise Unsafe Path routing protocol (MUP) [13] that maximizes the utilization of nodes that are going to fail sooner, in order to save power in the others. Although MUP selects nodes that must be in a dangerous area, many data are buffered. Thus, superfluous data concentration causes possibility of its loss by node burn before sending whole information. Moreover, MUP loses the high priority data, e.g., when a fire event is first detected reference, because MUP does not manage the priority of each fire alert data. Thus, it causes significant packets loss.

In this paper, two methods to achieve a lower dropped data packets ratio and smaller end-to-end delay is proposed. The first one is to limit attaching the highest priority only to truly urgent events, e.g., when a node detects a fire. The second one is to change the routing methodology. In presented scheme, high priority nodes transmit data to more survival node, while lower priority nodes transmit data to less survival node. In addition, the authors send high priority data ahead of low priority for low dropped ratio and delay. In order to show the effectiveness of proposed schemes, they are compared with MUP by evaluating the dropped rate and end-to-end delay of high priority data through computational simulation. Two environmental situations are considered. The first case is without wind, and the other one is with wind. The authors show that presented scheme can improve both of dropped ratio and end-to-end delay, and have less effect of wind.

The remaining of the paper is structured as follows: related work is described in Section 2. The conventional MUP is presented in Section 3. Section 4 explains the network configuration and forest fire scenario used in the simulation. Simulation results and analysis are discussed in Section 5. The paper ends with conclusions in Section 6.

2. Related Work

There are many fire forest-specific routing protocols. Environmental Monitoring Aware routing (EMA) [14] and

Delay-bounded Robust Routing protocol (DRR) [15] are proposed as path predictable methods in a fire event. In EMA, when nodes detect a fire, they send information to the data collector and then it informs every node in the network of the fire event. Therefore, only the safe nodes relay fire alert data to the sink. However, node state information might quickly become antiquated since the fire spreads very fast. On the other hand, DRR sends fire alert data and only uses more survivable nodes by leveraging neighbor node's state. Thus, DRR achieves better-dropped data ratio and delay. However, DRR does not consider network lifetime. Therefore, MUP [13] has been proposed. MUP extends the network lifetime by making the most of unsafe state nodes, which have detected fire and will be burned sooner or later. Moreover, it is important to know the forest fire shape in order to fight a fire. Yuanyuan *et al.* proposed a reliable wildfire monitoring system based on WSN [16]. In this system, fire-detecting nodes periodically send data including temperature to know fire point certainly. Serna *et al.* proposed method to obtain an approximation of the fire shape by analizing the data of WSN [17]. Fire spreads in response to the wind influence and a strong wind accelerates fire spreading. Since sensor nodes have to monitor in case of fire, how fast fire spreads by wind is important to be evaluated in the simulation. Kim develops a flame spread velocity model by testing fuel combustion and flame characteristics, and research about fire spreading with wind [18].

3. Conventional MUP Method

MUP selects nodes, which are going to be burnt earlier as forwarding nodes, in order to save the other nodes energy. MUP defines each node five levels of health status:

- safe – initial stage and while there is no fire,
- low safe – one-hop away from a detected fire,
- unsafe – fire detected,
- almost-failed – just about to be destroyed,
- dead – destroyed by fire or battery discharged.

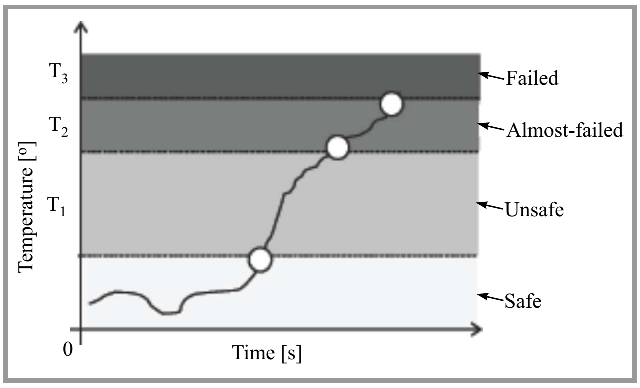

Fig. 1. Change state of detecting fire nodes.

In MUP, whenever a node detects temperature higher than a threshold, it changes its state. Figure 1 shows a node health status example versus measured temperature. Nodes always have safe status in normal situation. If a node detects fire and when the temperature increases above threshold $T_1 = 60°C$, its health status changes to almost-failed when the temperature reaches $T_2 = 100°C$. The node is considered totally burnt (failed) when the temperature reaches $T_3 = 130°C$, which is the maximum possible operating temperature. Nodes change state low safe from safe when a neighbor located one-hop away from the node-detected fire. All nodes send routing management messages including its own health state periodically.

3.1. Data Flow

During fire forest monitoring in WSN, nodes send measured data to the gathering host (sink) by relaying to other network nodes. Normally, all nodes periodically send data to the host at long interval e.g. 100 s. But, when nodes detect fire, the interval is much shorter e.g. 10 s.

If a node detects a fire, it changes its parent. MUP selects a parent node that must be in dangerous area in order to utilize its energy before being burnt in the fire. If one node has the lowest hop to the sink, then it will be selected as the parent. However, if there is more than one node, the mechanism considers the node's health status in the following order: unsafe, low safe and safe. The decision algorithm can be simplified as follows:

- nodes search the node with the lowest hop to the sink,
- if there is more than one node, then selects the node according to these health statuses in the following order: unsafe, low safe, and safe.

The almost-failed nodes are excluded from forwarding candidates to avoid broken paths due to failures. However, the routing mechanism selects them as the parent if there are almost-failed nodes only. Figure 2 shows an example of

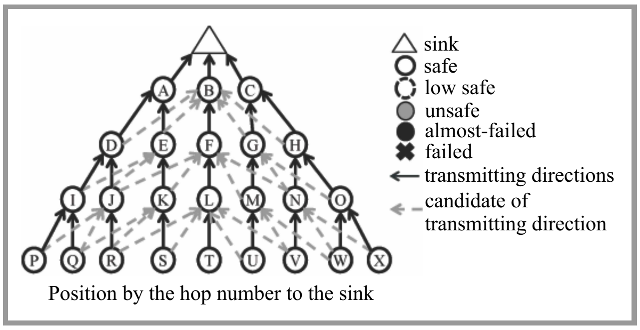

Fig. 2. Destination node selection without fire.

the way the MUP algorithm changes the routing tree of the network when nodes have detected fire and then be burnt. Figure 3 shows that MUP can utilize the energy of node K which will be burnt.

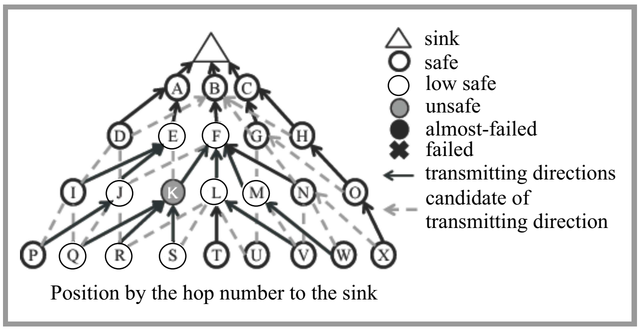

Fig. 3. Destination node selection with unsafe nodes.

3.2. Problem in MUP

Although MUP selects a node that must be in a dangerous area, many data are accumulated in its buffer. Thus, data overconcentration causes possibility of loss, while node is burnt before sending all information. At the same time, high priority alert data can be dropped because MUP does not consider the each alert data priority. Figure 4 shows an example of dropped high priority data. In Fig. 4, fire diffusion around node. Nodes J, L, Q, R and S select node K as their parent and then send high priority fire detection data. These high priority data may be dropped in node K due to the superfluous data concentration with fire.

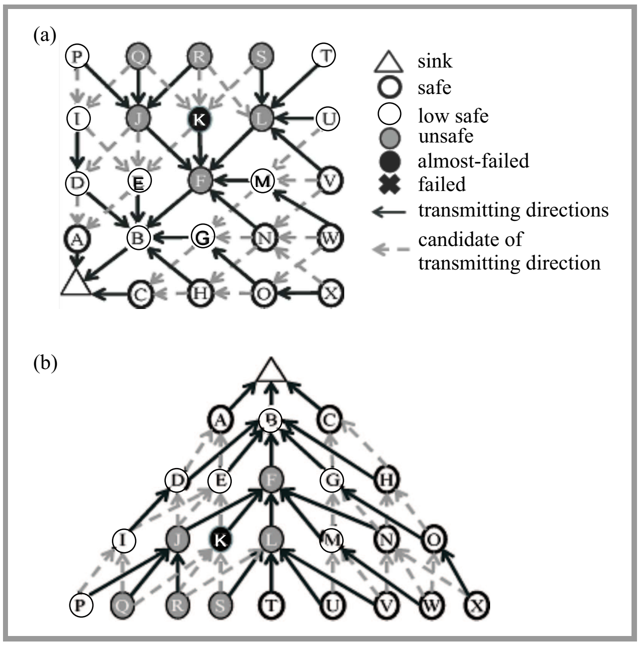

Fig. 4. Example of dropped high priority data: (a) the real position, (b) position by the hop number to the sink.

4. Proposed Method

To avoid data losses to send high priority data to more survivable nodes in order to reduce dropped rate of high priority fire detection data is proposed. The high priority is set only after fire detection and just before destruction by fire. Furthermore, nodes send most important data ahead of low priority data to improve dropped ratio and delay. This can be executed by sorting the buffer content.

4.1. Priority Fire Detection Data

The proposed method attempts to select a parent node depending on the priority of alert data. The almost failed nodes are removed from forwarding node candidates in order to avoid broken paths due to failures. Therefore, the three levels of priority were set to alert data depending on three node status (unsafe, low safe and safe). The highest priority 3 is the most important alert data when each node detects fire since fire detection in an early stage. The priority 2 was set with a probability P_1 for the fire detection data to be dropped, and priority 1 with a probability $1-P_1$ in order to avoid excessive increase of high priority data. And furthermore, the priority 2 was set to the alert data of changing state to almost-dead, because the node may not generate any more data by the destruction. Similarly, the priority 2 was set to the alert data of changing state to almost-dead in order to avoid an excessive increase of the highest priority information.

4.2. Parent Election

To avoid all priority data concentrated on a specific node high priority data is sent to a node far from the fire. All nodes inform about their health state in time, and each node recognizes neighbor node's state. Table 1 shows the parent election depending on the priority of alert data. Each node checks the neighbors' state with fewer numbers of hops to the sink than itself in order shown in Table 1. When an applicable node is found, the node transmits data to him. If there is no candidate, the node looks up the routing table and finds the parent candidate who is as far as or further than itself.

Table 1
The parent election depending on the priority of alert data

Order	Priority 1	Priority 2	Priority 3
1	Unsafe	Low safe	Safe
2	Low safe	Safe	Low safe
3	Safe	Unsafe	Unsafe
4	Almost-dead	Almost-dead	Almost-dead

For example, when a node has three hops to the sink but only has neighbor nodes with four hops, it selects a node with four hops to the sink in order shown in Table 1. When a node transmits the priority 3 data, each node checks state of neighbor nodes in turn from the safe state to almost-dead. Then the node who has data of priority 2 checks in turn from the low safe state to almost-dead state. Therefore, the dropped ratio of the priority 2 data gets lowered and priority 2 data go through the different path. Furthermore, each node has the fire detection priority 1 data checks in turn from the unsafe state to almost-dead state. Thus the energy could be utilized before it is lost on fire. Figure 5 shows transmission of priority 2 data on fire detection. In Fig. 5, priority 2 data concentrate node B whose state is low safe according to the Table 1. However, low safe nodes can send more data than unsafe state nodes because of more time until being burnt by fire. Furthermore, low safe state nodes have high possibility of destruction by fire, so its energy could be utilized before it melts down.

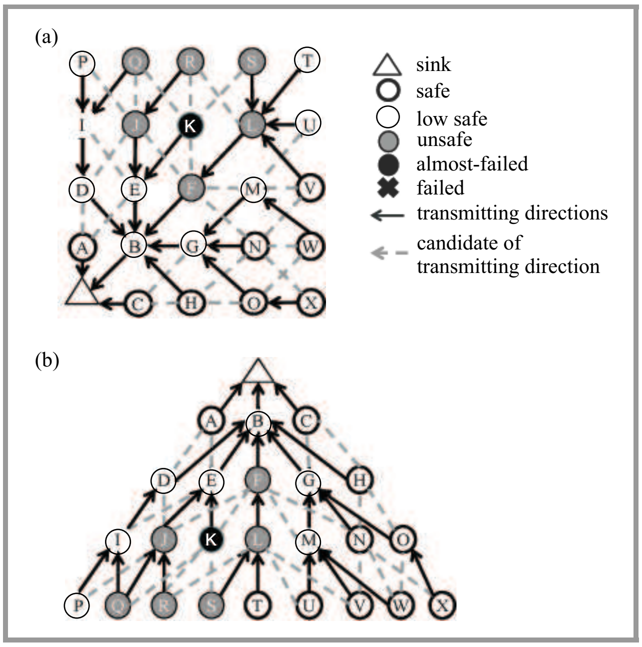

Fig. 5. Data transmission of priority 2 data of fire detection: (a) real position, (b) position by the hop number to the sink.

4.3. Sort in Buffer

Data in the buffer are rearranged to send high priority data ahead of low priority. Therefore, they reach the sink early. Moreover, dropped data ratio (DDR) decreases by transmitting high priority data before fire spreading.

5. Performance Evaluation

5.1. Simulation Model

The performance of the conventional and proposed schemes are evaluated in terms of the fire alarm DDR, end-to-end delay of fire alarm data, total number of transmitted and received data by detection fire node and total residual energy in survivable nodes after fire. The total number of transmitted and received data by fire detection node are evaluated to show if the proposed method efficiently utilizes the energy before the destruction as much as the conventional method. The authors define DDR as the data coefficient, which are not reached the sink to all data generated by alive nodes. Similarly, the MUP performance with priority and with sort are evaluated to show its effectiveness by considering priority or sort data in MUP method. MUP with priority means the combination of node health status by MUP and proposed parent selection. In other words, it is equivalent to the proposed method without sorting buffer. MUP with sort means the combination of MUP parent selection and proposed data sort method.

Table 2
Simulation specifications

Number of sensing nodes	100 $(10 \cdot 10)$
Distance between nodes	100 m
Node arrangement	Grid
Number of sink nodes	1
Fire spread speed	5 m/s
Time interval between fire alarm data	10 s
Wind directions	9 directions
Wind speed	2 m/s
Node status	5 levels
P_1	0.2, 0.4, 0.6, 0.8, 1
Bit rate	256 kb/s
Data size	2560 B
Priority of fire alarm data	3 levels
Tx power consumption	345 mW
Rx power consumption	260 mW
Power consumption on idle state	13 mW
Power consumption on sleep state	0.19 mW
Simulation tool (language)	C

Table 2 shows the used simulation parameters based on [13], [18]. Figure 6 shows a simulation topology model. In presented model, initial fire randomly occurs from the node except the sink and spread towards 40% of the network. The authors consider two fire-spread situation: with and without wind. When no wind situation, fire is diffused concentrically. Fire spreads in the direction of the

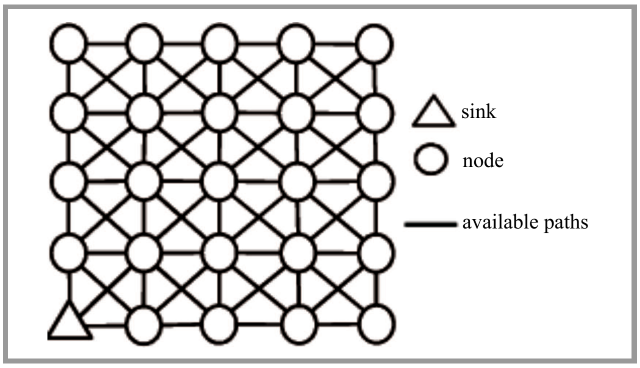

Fig. 6. Topology model.

wind [18], and nodes might be isolated by fire when wind blows to the specific direction. Figure 7 shows that a graphical model of nodes isolated by fire spread. The wind blows from right to left and fire spreads this direction. Therefore, upper left nodes will be isolated and cannot send data to the sink via the shortest path. The data of these nodes tend to be dropped or delayed. Figure 8 shows used wind directions. The nine patterns ($W_N, W_1, \ldots, W_8$) including the no wind (W_N) was used. W_2 blows to the sink and W_6 blows to the opposite direction. Note that the wind direction and P_1 remain unchanged and the wind blows towards only one direction during each trial of the simulation. It is assumed that sensors are deployed so tall as to avoid obstacles, e.g. grass [19].

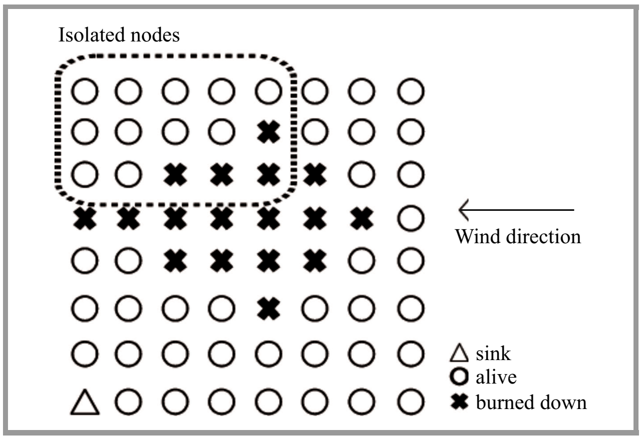

Fig. 7. Graphical representation of nodes isolated by fire spread.

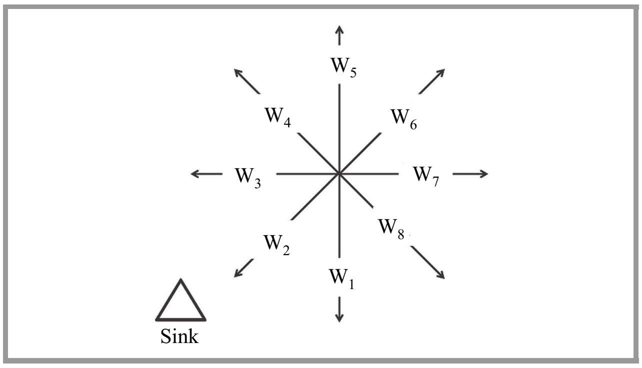

Fig. 8. Wind directions.

5.2. DDR of Fire Alarm Data

Figure 9 shows the fire alarm DDR versus data priorities. The DDR is presented by averaging the results of all wind directions ($W_N, W_1, \ldots, W_8$) and P_1 ($P_1 = 0.2, 0.4, \ldots,$ and 1).

Although MUP does not consider fire alarm data priority, the high priority DDR is lower than that of low priority data. This is because priority 3 data is first sent before spreading fire. Figure 9 shows that the proposed scheme and MUP with priority achieve better DDR in all priorities than MUP. The proposed scheme reduces the DDR of priority 2 by 13% and priority 3 by 10% compared to MUP. This is because the proposed method transmits high priority data to more survivable nodes. In MUP with sort, the higher priority data, the better DDR can be achieved. This is because this method sends high priority data ahead of low priority data by sorting the buffer content at each intermediate node. Figure 10 shows the fire alarm DDR versus probability P_1. Figure 10 shows the DDR by averaging the results of all wind directions. The DDR of priority 1 gets decreased as P_1 decreases. This is because P_1 controls how frequent the priority 1 data is chosen after priority 3 data are sent. On the contrary, the DDR of priority 2 data is increasing as P_1 decreases in Fig. 10. Moreover, as P_1 decreases, the DDR of priority 2 is more moderately decreased than that of priority 1. This is because data concentration is relaxed as P_1 decreases.

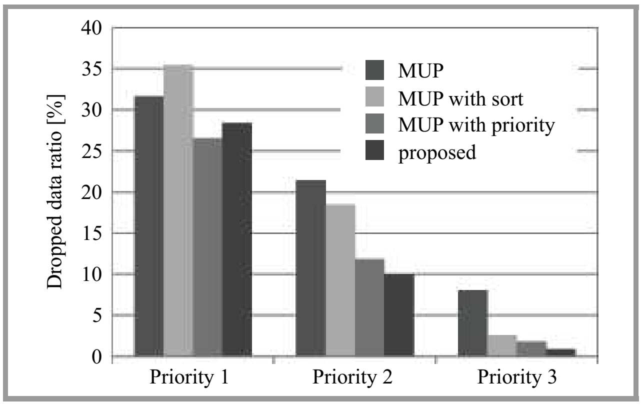

Fig. 9. Fire alarm dropped data ratio versus data priorities.

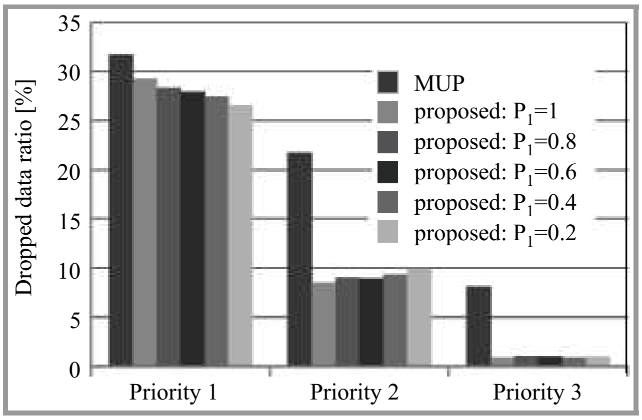

Fig. 10. Fire alarm versus P_1 data priorities.

Figures 11 and 12 show the DDR of priorities 2–3 data per wind directions. It is shown that the proposed method decreases the priority 3 DDR less than 2%, regardless wind directions. This is because the proposed method relays priority 3 data to more survival node before being surrounded by burning down nodes. It is shown from Fig. 11 that

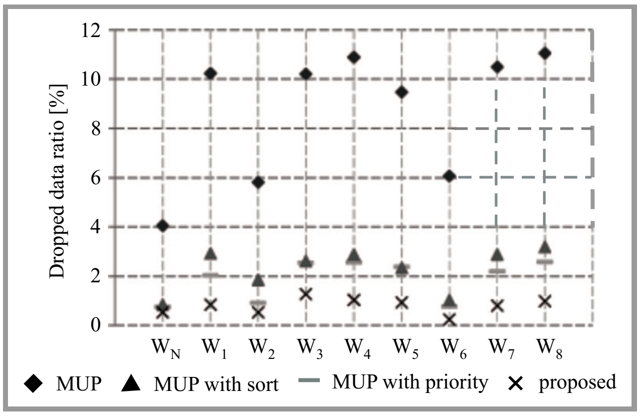

Fig. 11. Priority 3 DDR as a function of wind directions ($P_1 = 0.4$).

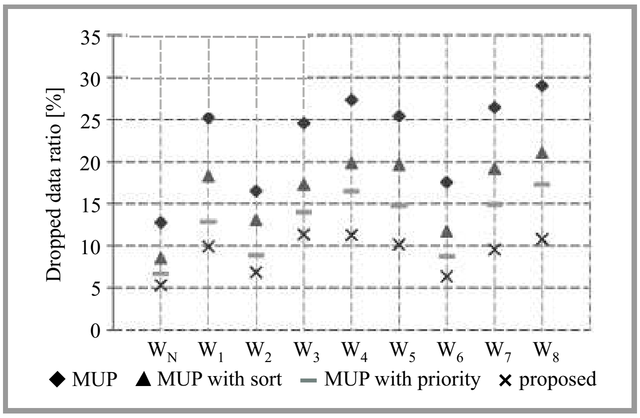

Fig. 12. Priority 2 DDR versus wind directions ($P_1 = 0.4$).

the DDR of W_6 is lowest in the proposed method. The fire spreads slowly toward the sink node with W_6 direction and the proposed method with W_6 sends high priority data before being surrounded by burning down nodes. Figures 11 and 12 show that proposed method decreases the DDR more than MUP in the case if wind is present. The proposed method sends high priority data before being isolated, as shown in Fig. 7. Figure 13 shows the priority 1 DDR as a function of wind directions. It is shown that the DDR without wind (W_N) is the lowest. With wind present,

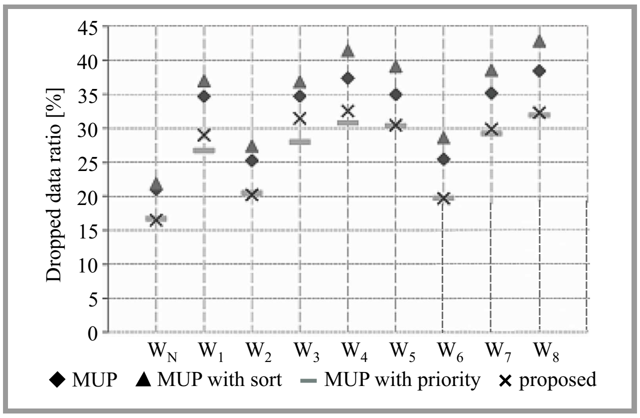

Fig. 13. Priority 1 DDR versus wind directions ($P_1 = 0.4$).

nodes are more quickly burnt down when the wind blows towards them. It is shown that the DDR of W_4 and W_8 are higher than other wind directions. In W_4 and W_8 directions, fire spread perpendicularly for the direction toward the sink node and more relay nodes which cause missed routing path disconnect tend to be burnt down. Although the ratio of dropped data varies, each scheme is similarly influenced by the effect of wind (Fig. 13). In MUP with priority and with proposed method, data concentration of priority 1 are relaxed.

5.3. Delay of Fire Alert Data

Figure 14 depicts the delay of fire alarm data as a function of data priority without wind (W_N). Figure 14 shows the delay by averaging the results of all wind directions and $P_1 = 0.2$, 0.4, …, and 1. By definition the delay is the time from generating a data in a node to receiving it by the sink. Figure 14 shows that the proposed method

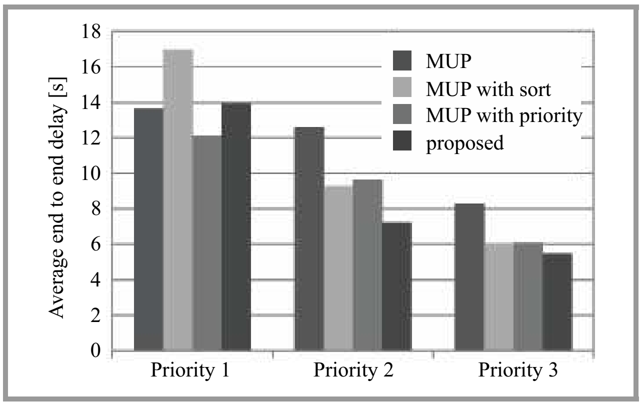

Fig. 14. Delay of fire alarm data versus data priorities.

achieves better fire alert priority 2 data delay by 38% and priority 3 by 29% than MUP. The high priority data collisions are avoided by parent election of the proposed method. Moreover, the proposed method and MUP with sort is better on priority 1 data delay compared with MUP because each node sends high priority data ahead of low priority. Figure 15 shows delay of fire alarm data ver-

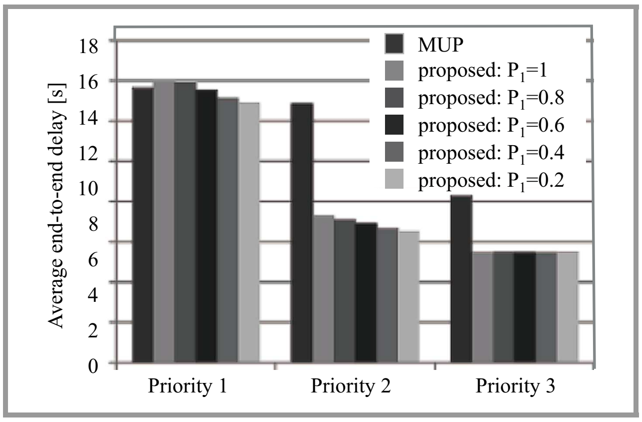

Fig. 15. Delay of fire alarm data versus P_1 data priorities.

sus data priorities changing P_1. It presents the delay by averaging the results of all wind directions. The delays of proposed method with $P_1 = 1$ and $P_1 = 0.8$ are longer than MUP. It is a result of change the order, and each node sends high priority data ahead of low priority. Moreover, these delays are decreasing with P_1 because the data concentration is relaxed as P_1 decreases. Figure 16 shows

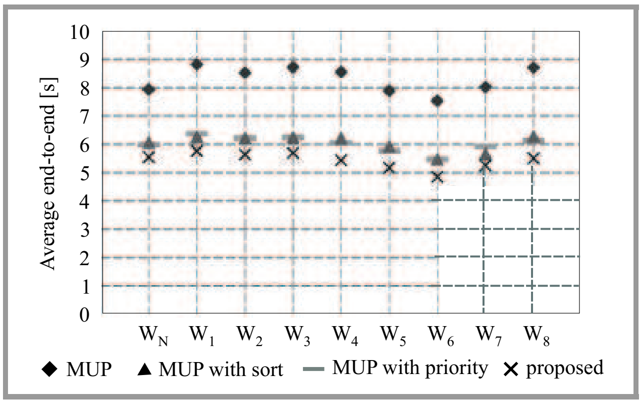

Fig. 16. Priority 3 data delay versus wind directions ($P_1 = 0.4$).

priority 3 data delay as a function wind directions. The delay of MUP with sort, MUP with priority and proposed method are almost the same and lasts about 6 s. The fire is prevented from spreading to a large area by detecting in about six minutes [20]. Therefore mentioned 6 s delay is enough to detect fire.

5.4. Data Amounts

Figure 17 shows the total of transmitted and received data by fire detection nodes without wind by averaging the results of all wind directions and $P_1 = 0.2, 0.4, \ldots,$ and 1.

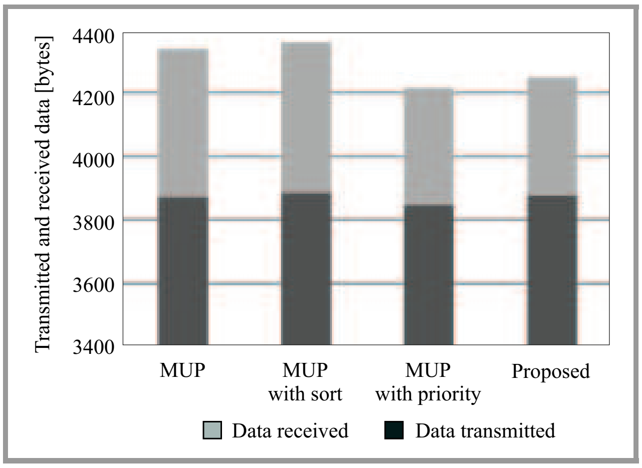

Fig. 17. Transmitted and received data by fire detection nodes.

It is shown that the proposed scheme achieves more transmitted data by fire detection node than the MUP. The total transmitted and received data of the proposed method is as many as in the MUP method, although the proposed method decreases the number of received data by fire detection node compared with MUP. It is also shown that total transmitted and received data in MUP and MUP with sort are almost the same. The reason is that parent selection method of these schemes are the same. The proposed method controls the total transmitted and received data by the fire detection nodes approximately 2% of decrease compared with MUP method. Figure 18 shows the total transmitted and received

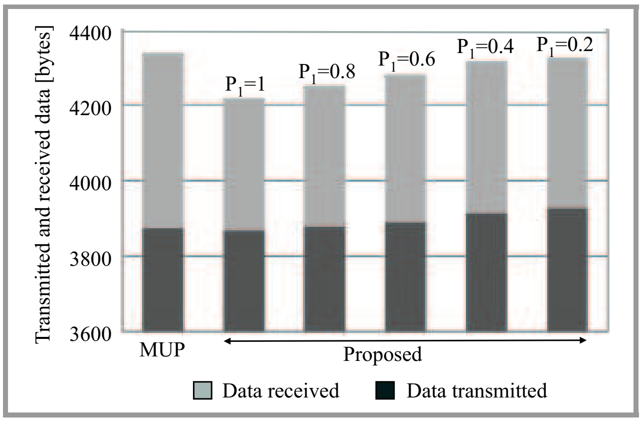

Fig. 18. Transmitted and received data by fire detection nodes versus P_1.

data by fire detection nodes versus P_1 by averaging the results of all wind directions. It is shown from Fig. 18 that presented scheme achieves same transmitted and received data compared with the conventional scheme.

6. Conclusion

In this paper, a new forest fire monitoring system is proposed to reduce dropped rate of high priority fire detection data, by specifying a high priority on data immediately after fire detection and just before destruction by fire. Furthermore, the node only transmits high priority data to a node, which had low possibility of destruction by fire to achieve low end-to-end delay of high priority fire detection data. The simulation results showed that proposed scheme could reduce dropped rate of high priority data and the end-to-end delay, and have less effect of wind compared with the conventional solutions.

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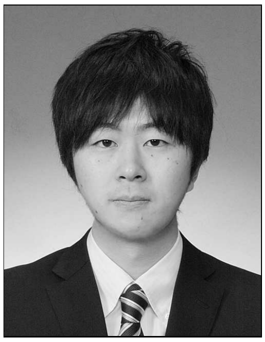

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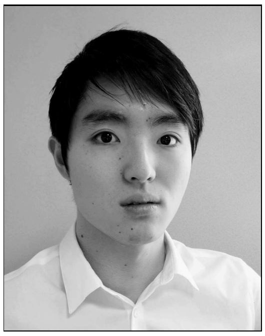

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Paper

On-demand QoS and Stability Based Multicast Routing in Mobile Ad Hoc Networks

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Abstract—Finding a connection path that remains stable for sufficiently longer period is critical in mobile ad hoc networks due to frequent link breaks. In this paper, an on-demand Quality of Service (QoS) and stability based multicast routing (OQSMR) scheme is proposed, which is an extension of ad hoc on-demand multicast routing protocol (ODMRP) to provide QoS support for real time applications. The scheme works as follows. Each node in the network periodically estimates the parameters, i.e., node and link stability factor, bandwidth availability, and delays. Next step is creation of neighbor stability and QoS database at every node by using estimated parameters. The last sequence is multicast path construction by using, route request and route reply packets, and QoS and stability information, i.e., link/node stability factor, bandwidth and delays in route information cache of nodes, and performing route maintenance in case of node mobility and route failures. The simulation results indicate that proposed OQSMR demonstrates reduction in packet overhead, improvement in Packet Delivery Ratio (PDR), and reduction in end-to-end delays as compared to ODMRP, and Enhanced ODMRP (E-ODMRP).

Keywords—mobile ad hoc network, mobility, multicast routing, QoS, stability.

1. Introduction

Mobile Ad hoc Networks (MANETs) are self-organizing networks consisting of mobile nodes which can be rapidly deployable in emergency situations like battlefields, earthquakes, tsunami, floods, or any major disaster areas. MANETs are deployed without base stations and do not have wired infrastructure. They must adapt to traffic and node mobility patterns. In MANET, a mobile node can act as a router as well as a host. Two nodes can communicate with each other eventhough they are outside their transmission range. The successful communication in such a situation depends upon the intermediate node mobility and failure probability [1].

Normally, MANETs are used for group communications, where multicast protocols are efficient compared to unicast protocols since they improve the efficiency of the wireless links in MANETs and when an application demands for sending multiple copies of messages from multiple sources to multiple receivers. Multicasting reduces the communication costs by sending the single copy of the data to multiple recipients rather than sending multiple copies by using multiple unicasts. Thus it minimizes the link bandwidth, processing, and transmission delay [2].

In broad sense, there are two types of multicast protocols: mesh and tree based. Tree based structures are not stable since they need to be reconstructed when topology is changing frequently [3]. Once the tree is established, a packet will be sent to all nodes in the tree. A packet traverses each node and link only once. It is not suited for MANETs since the tree could break any time due to changes in the topology. Therefore, focus of this work is on mesh based routing since it provides better service when a network is highly dynamic.

A mesh based structure can have multiple parents and a single mesh structure can connect all multicast group members with multiple links. When a primary link breaks away due to mobility of a node, alternate links are immediately available. For long duration connections, nodes/links on a path must be stable so that connection failures can be reduced. Stable connection facilitates data transfer without interruption. The probability of route failure can be reduced by lowering either the link failure rate or the number of links that compose the route. It is important to note that delay bounded route selection avoids larger delays.

Constructing and maintaining a multicast mesh should be simple so as to keep minimum control overheads. Most of the multicast routing protocols require periodic transmission of control packets in order to maintain multicast group membership; thus requires more bandwidth. The objective of presented work is to design and analyze a multicast mesh based on-demand routing scheme in MANET, which is enhanced version of On-Demand Multicast Routing Protocol (ODMRP), to provide bandwidth satisfied, reliable and robust route.

The rest of the paper is organized as follows. Section 2 presents an overview of existing MANET multicast protocols, Section 3 discusses the proposed work in detail. Simulation and result analysis are presented in Section 4, and conclusions and future works are given in Section 5.

2. Related Works

With the rapid development of multimedia applications in MANETs, there is an increasing need for QoS guarantee for a real time application. Therefore, protocols designed for MANETs should involve satisfying application requirements while optimizing network resources. In the design of routing protocols, finding the stability of nodes play an important role in establishing a stable and QoS path that offers better packet delivery ratio and low latency.

ODMRP is a protocol which makes use of group of forwarding nodes to establish a mesh of nodes for every multicasting group [4], [5]. The work on ODMRP in [6], considers node's energy in route selection from source to destination and results confirm that there is an improvement in stability of the route due to low energy consumption based routes.

In [7], E-ODMRP is presented which is an enhancement of ODMRP. It does not have the forwarder lifetime where as ODMRP's forwarder has a timeout which is 3 times the refresh interval. The route refresh rate is dynamically adapted to the environment rather than refreshing at fixed intervals as in ODMRP, which is a key parameter that has critical impact on the network performance.

In [8], the stable paths are found based on selection of forwarding nodes that have high stability of link connectivity. The work given in [9], proposes a QoS – aware Multicast Routing Protocol (QMRP) based on mesh architecture which offers bandwidth guarantees for applications in MANETs. QMRP takes an adaptive approach and starts with single path routing. When a single path routing fails, it switches to multipath routing by adding new searching.

In [10], a Source initiated Mesh based and Soft-state QoS Multicast Routing Protocol (SQMP) for MANETs is proposed. The ant colony optimization technique is used for finding best route to its destination through the cooperation with other nodes. In [11], a weighted multicast routing algorithm for MANET is proposed to find stable routes in which the mobility parameters are assumed to be random variables with an unknown distribution.

In [12], a stability-based unicast routing mechanism is discussed in which both link affinity and path stability are considered in order to find out a stable route from source to destination. It is then extended to support multicast routing where only local state information (at source) is utilized for constructing a multicast tree. The work given in [13], proposes a new algorithm for tree-based optimization. The algorithm optimizes the multicast tree directly, unlike the conventional solutions which find paths and integrating them to generate a multicast tree. The fuzzy logic modified Ad hoc On-demand Distance Vector (AODV) routing protocol for multicast routing in MANETs is discussed in [14]. The fuzzy weighted logic multi-criteria are based on the parameters like remaining battery power of the nodes, number of hop-counts and sent packets.

In [15], a multi-constrained QoS multicast routing scheme is presented using genetic algorithm. The scheme applies limited flooding using the available resources and minimum computation time in a dynamic environment. In [16], only the nodes that satisfy the delay requirements are used to flood the route request messages. The nodes are modeled as M/M/1 queuing systems, in which delay analysis is made based on random packet arrival, service process, and random channel access.

The Mesh-evolving Ad hoc QoS Multicast (MAQM) routing protocol presented in [17], achieves multicast efficiency by tracking the availability of resources for each node within its neighborhood. The QoS status is observed continuously and updated periodically to perform QoS provisioning. In [18], authors have evaluated the performance of mesh and tree-based multicast routing schemes relative to flooding, and also proposed two variations: flooding, scoped and hyper flooding, as a means to reduce overhead and increase reliability, respectively.

In [19], a multi-path QoS multicast routing (MQMR) protocol is proposed. The scheme offers dynamic time slot control using a multi-path tree. Work given in [20], proposes a novel Efficient Geographic Multicast Protocol (EGMP). EGMP uses a virtual-zone-based structure to implement scalable and efficient group membership management.

Effective transmission power control is a critical issue in the design and performance of wireless ad hoc networks. Current design of packet radios and protocols for wireless ad hoc networks are primarily based on common-range transmission control. The work given in [21], analyzes some of the widely used routing protocols with varying transmission range, mobility speed and number of nodes.

The work given in [22], uses the mobility and link connectivity prediction to find routes and forwarding groups, and to reconstruct the path in anticipation of topology changes. The Associativity-based Ad hoc Multicast (ABAM) protocol given in [23], establishes multicast session on-demand and utilizes an association stability concept, which refers to spatial, temporal connection and power stability of node with respect to neighbors. The protocol improves throughput and has low communication overhead. In [24], Selfish Check Negotiation Protocol (SCNP) is presented which allows nodes to negotiate for collaboration. The impact of being selfish and unselfish used in network communication performance are discussed.

In [25], authors present the Multimedia Broadcast/Multicast Service (MBMS) extension, that allows multiple variants of the same content to be economically distributed to heterogeneous receivers, explicitly taking into account the possibility of using either dedicated or common radio channels. In [26], a novel analytical method for performance prediction estimation of single- and multi-layer Multistage Interconnection Networks (MINs) under multicast environments is presented. The "Cell Replication While Routing" is used as a packet routing technique, and the "full multicast" mode as transmission policy is employed in all the MINs under study. The work presented in [27], estimates and selects core for reducing multicast delay variation for delay sensitive applications in Delay Variation Bounded Multicast Tree (DVBMT).

Adaptive Demand-Driven Multicast Routing protocol (ADMR) presented in [28], supports source specific multicast joins and to route along shortest paths, and uses no periodic network-wide floods of control packets, periodic neighbor sensing, or periodic routing table exchanges, and requires no core.

A reliable ODMRP is proposed in [29], for preferable throughput. It constructs multicast routing based on the cluster, and establishes a distributed mechanism of acknowledgment and recovery of packet delivery. A single forwarding path created in ODMRP is vulnerable to node failures, since a set of misbehaving or malicious nodes can create network partitions and mount Denial-of-Service (DoS) attacks. Resilient ODMRP (RODMRP) [30], offers more reliable forwarding paths in face of node and network failures and DoS attacks.

3. Proposed Work

This section presents node and link stability, bandwidth and delay estimation models, route discovery and maintenance phases.

3.1. QoS Metrics

The authors propose certain parameters to describe the Quality of Connectivity (QoC) for extracting the stable and QoS links connecting a pair of nodes over time. This is used as a criteria for route selection algorithm. Reliable network requires more stable nodes and high quality links which satisfy bandwidth and delay as QoS constraints. The set of forwarding nodes with higher stability can improve the routing performance. This section presents stability, bandwidth and delay estimation models used in presented scheme.

3.1.1. Node Stability

The stable nodes are necessary in forwarding group to provide better packet delivery services. Node stability in terms of movement around its current position gives an idea of stationary property of node. The authors use node stability metric from their previous work given in [31], to identify stable nodes in a path for forwarding packets from a source to multicast group.

Two metrics to represent node stability as the quality of connectivity is identified: *self stability*, and *neighbor nodes stability*. The steps in finding the stability of a node are as follows:

- all the nodes in MANET find the self stability, i.e., node movement relative to its previous position,
- find neighbors stability of all the nodes in MANET by considering the neighbors self stability. Each node in a MANET will compute the node stability factor based on self stability, and neighbor nodes stability.

Self stability. It can be defined as the node's movement with respect to its previous position. If a node is trying to move away from its position, the distance of the movement and transmission range decides the stability. A node is said to be stable if its movement is within given fraction of its transmission range. Consider the scenario as shown in Fig. 1, where a node with transmission range r moves from position $(x_r,\ y_r)$ to $(x_n,\ y_n)$ in a given time window by a distance d.

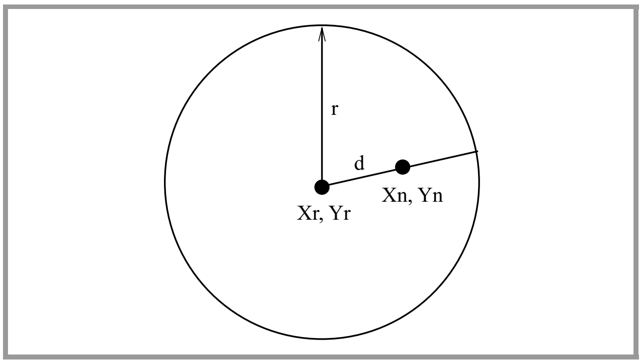

Fig. 1. Node movement.

When a node moves out from its previous position to the next position, its position stability keeps changing with respect to the distance moved. This change in distance (d_i^t) of a node i, in a time window t is estimated by using Eq. (1).

$$d_i^t = \sqrt{(x_n - x_r)^2 + (y_n - y_r)^2}\,. \tag{1}$$

Based on the movement of the distance at every time window, the self stability metric $S_s(t)$ can be estimated as given in Eq. (2). $S_s(t)$ varies in the range 0 to 1. When the movement distance d_i^t of a node increases from its previous position, the self stability value will decrease. For the requirement of the higher degree of movement stability, $r/2$ can be replaced by $r/4$ or $r/8$.

$$S_s(t) = \begin{cases} 1 - \dfrac{d_i^t}{r/2} & \text{if } 0 \le d_i^t < r/2 \\ 0 & \text{otherwise} \end{cases}\,. \tag{2}$$

There are some limitations in calculation of self stability due to influence of GPS accuracy and resolution. Better results can be estimated with higher accuracy and resolution in GPS. This work assumes that GPS accuracy and resolution is limited to 95% and 7.8 meters, respectively [32].

Neighbor node stability. It can be defined as how well a node is being connected by its neighbor in terms of their self stability. The nodes can exchange messages with each other, if they are within the transmission range. Each node accumulates connectivity information and signal stability of one hop neighbors, and maintains a neighbor list.

The degree of a node n is represented as number of links (or nodes) connected to it, and is denoted as ND. The neighbor node stability of a node $N_s(t)$ with respect to neighbors at time t can be expressed as in Eq. (3):

$$N_s(t) = \alpha \times \frac{1}{ND}\sum_{i=1}^{ND} {S_s}^i(t) + (1-\alpha) \times N_s(t-1)\,, \tag{3}$$

where α is the weightage factor (lies between 0 and 1), and is distributed between 0.6 and 0.7, since they yield better results in simulation. $N_s(t-1)$ is the recent neighbor node stability, $S_s{}^i(t)$ is the self stability of neighbor node i. The authors are using the stability model to select nodes with higher self and neighbor stability values such that the selected path through such stable nodes stays for a longer duration.

3.1.2. Link Stability

Link stability between the nodes indicates quality and life time of the connection. The link stability estimated in the scheme is based on two parameters: received signal strength and life time of the link.
The Algorithm 1 represents a pseudocode for updating link stability status between the nodes. The different parameters used in the algorithm are as follows:

- lifetime – duration of continuous connectivity between the nodes,
- lifetime threshold – indicates the maximum limit of link lifetime that decides link stability,
- link stability status – is a boolean variable that defines link stability between the nodes,
- recent – indicates most recent response received for a Hello packet from a neighbor,
- P – number of Hello packets,
- received signal strength – is the strength of signal received from a neighbor,
- signal threshold – is an acceptable signal strength to be received from neighbors.

Algorithm 1: Link stability status between the nodes

```
P = No_of_Hello_Packets;
lifetime = 0;
link_stability_status = 0;
Recent = 0;
lifetime_threshold = P × Hello_Packet_Interval;
while P > 0 do
    if received_signal_strength ≥ signal_threshold then
        lifetime = lifetime + 1;
        Recent = 1;
        P = P − 1;
    else
        Recent = 0;
        P = P − 1;
    end if
end while
lifetime_sec = lifetime × Hello_Packet_Interval;
if (lifetime_sec > lifetime_threshold) and (Recent)
then
    link_stability_status = 1;
else
    link_stability_status = 0;
end if
```

Following parameter values are considered in Algorithm 1: the signal threshold = -8.9 dB [33], No. of Hello Packets = 4, Hello packet exchange interval = 60 s, and lifetime threshold is three times of the Hello packet exchange interval. A typical neighbor information for a node with neighbors A, B, C, etc., is given in Table 1. It comprises of neighbor Id and its related information such as neighbor stability factor, link stability factor, recent, lifetime, and link stability status. For every neighbor node, link and node stability factor will be estimated as discussed in Subsection 3.1.3.

3.1.3. Stability Factor

This section describes computation of stability factor by using node and link stability factor.

Node stability factor. First there is need to map the self stability, and neighbor nodes stability on to a single metric called node stability factor, Nsf. This can be expressed as in Eq. (4). The $Nsf(t)$ in time interval t represents the stability of node at a given time interval with respect to its neighbor movement from their respective positions. Higher the value of $Nsf(t)$ indicates better stability

$$Nsf(t) = f(S_s(t), N_s(t)) = \beta S_s(t) + (1-\beta) N_s(t) \,. \quad (4)$$

The weight factor β denotes the relative importance of the quantities $S_s(t)$ and $N_s(t)$. It is assumed the value of β to be distributed between 0.6 and 0.7, since they yield better results in simulation.
Stability factor of a node is computed only if self stability and neighbor stability is greater than zero. Thus this scheme extracts the highly stable nodes and adjusts the network topology, so as to reduce the probability of route failure.

Link stability factor. A node is capable of estimating its neighbor's time of connection called as life time of a node. The node is assumed to be aware of its direct (or immediate) neighbor's relative speed, called as v. The relative speed is calculated based on [34]. Let's denote the range of a node as r, and the distance moved by the node as d. The remaining distance is (r–d) for which connectivity may still exist. A relationship between these parameters when the link_stability_status = 1, is given in Eq. (5), called as link stable duration (Lsd):

$$Lsd = \frac{(r-d)}{v} \,. \quad (5)$$

Link stable duration can be normalized by using a lifetime_threshold (LTT), which has a higher value than any Lsd's may be observed. Normalized Lsd, denoted as link stability factor, Lsf at a given time interval t is given in Eq. (6):

$$Lsf(t) = \begin{cases} \dfrac{Lsd}{LTT} & \text{if } Lsd \leq LTT \\ 1 & \text{otherwise} \end{cases} \,. \quad (6)$$

Table 1
Neighbor information table

Neighbor Id	Neighbor stability factor	Link stability factor	Recent	Lifetime	link_stability_status
A	0.9	0.2	0	3	0
B	0.8	0.4	1	4	1
C	0.6	0.3	0	3	0
...	...	...	...	...	...
...	...	...	...	...	...

Stability-Factor-Between-Nodes. Proposed routing scheme makes use of node stability factor coupled with link stability factor called as Stability Factor Between Nodes (SFBN). SFBN is used for QoS based applications to find the route from a source to destination. SFBN (a normalized value) is given in Eq. (7), which helps in selecting stable nodes and links for routing in multihop networks which can stay together for a longer duration

$$SFBN(t) = \frac{1}{2}\big(Nsf(t) + Lsf(t)\big). \tag{7}$$

The path from source to multicast group will be forwarded through intermediate links, and the link with minimum SFBN is selected as PathSFBN at a given time interval t. This is given in Eq. (8), and is denoted by PathSFBN for N intermediate links

$$PathSFBN(t) = \min(SFBN_i(t)); \forall i = 1 \ldots N. \tag{8}$$

3.1.4. Delay Estimation

For delay estimation, an arbitrary node that contributes to traffic forwarding using the M/M/1 queuing system is modeled. This queue represents a single queuing station with a single server [35]. The authors assume that the contributing nodes are served by a single server with first come first serve queuing policy. Packets arrive according to a Poisson process with rate λ, and the probability distribution of the service rate is exponential, denoted by μ. The maximum size of the queue in every node is represented by K.

To satisfy delay requirements in multimedia real time applications, packets must be received by multicast receivers which satisfies the application delay constraints. When a packet is to be sent either by a source node or forwarding group of nodes; it experiences three types of delays: queuing, contention and transmission delay. The total delay considered over a link between two nodes is given by

$$d_{Total} = d_Q + d_C + d_T. \tag{9}$$

The queuing delay denoted by d_Q is the delay between the time the packet is assigned to a queue and the time it starts transmission. During this time, the packet waits while other packets in the transmission queue are transmitted. This is the amount of time a packet is spent in the interfacing queue. The average contention delay, denoted by d_C is the time interval between the time the packet is correctly received at the head node of the link and the time the packet is assigned to an outgoing link queue for transmission by the physical medium. The transmission delay denoted by d_T is the one between the times that the first and last bits of the packet are transmitted over the physical medium successfully. In proposed model, every node will estimate single hop delay with its neighbor nodes. The maximum value of d_{Q+C} is approximated as the ratio of maximum queue size over the service rate in a node, and is given by

$$d_{Q+C} \approx \frac{K}{\mu}. \tag{10}$$

Transmission delay. Transmission mechanism used for multicasting is different from unicast in random access wireless communications. To transmit data packets over a physical media, random access MAC model is employed. Source node uses carrier sense multiple access with collision avoidance protocol (CDMA/CA) to avoid packet collision.

When a node has data to send, it senses the physical medium. If the medium is idle, the packets are injected into the network. Otherwise, it waits until the medium gets idle and then it counts down a certain period of time called back-off time before sending a data packet. When backoff reaches zero, the packet is transmitted. When a collision is detected, the contention window size is doubled and the process is repeated. After a fixed number of retry attempts, the packet is dropped. The time for which channel is available for an arbitrary node with ϕ interfering nodes can be expressed as

$$d_{BussyChannel} = \frac{\phi \times m}{bw}, \tag{11}$$

where m represents the packet size and bw denotes the single hop bandwidth between two nodes. Therefore the time that the channel is available for data transmission in time unit (1 s) is

$$d_{FreeChannel} = 1 - d_{BussyChannel} = 1 - \frac{\phi \times m}{bw}. \tag{12}$$

The service time can be defined as

$$T_{serviceTime} = \varepsilon + \frac{m}{bw}, \tag{13}$$

where ε is the duration of the back-off time during which channel keeps sensing for idleness. The packet will be transmitted if the backoff window counts down to zero. In fact this time depends on the network load, since the process of countdown will be halted because the medium is found to be busy. The pausing period of a packet stops transmitting, which depends on the backoff interval and this in-turn depends on the network load. Finally, the mean transmission time required to transmit a packet is defined as the ratio of the service time over the fraction of time the channel is free. Hence, mean transmission delay is

$$d_T = \frac{\varepsilon + \dfrac{m}{bw}}{1 - \dfrac{\phi \times m}{bw}}. \quad (14)$$

Now, the total single hop delay between two nodes is the sum of all the delays mentioned in Eq. (9), and it is

$$d_{Total} = d_{Q+C} + d_T = \frac{K}{\mu} + \frac{\varepsilon + \dfrac{m}{bw}}{1 - \dfrac{\phi \times m}{bw}}. \quad (15)$$

By using Eq. (15), each node will estimate the single hop delay. The path delay or end-to-end delay from source to destination is the delay through intermediate links and is additive in nature. It is given by Eq. (16), denoted by $Delay(P_i)$ where P_i is the i-th path, N is the number of intermediate links, and for each path:

$$Delay(P_i) = \sum_{j=1}^{N} d_{Total_j}. \quad (16)$$

3.1.5. Bandwidth Estimation

The bandwidth information is one of the important metric of choice for providing Quality of service (QoS). The authors considered their previous work presented in [36], to estimate the available bandwidth based on the channel status of the radio link to calculate the idle and busy periods of the shared wireless media. By observing the channel utility, the measure of the node activities can be taken as well as its surrounding neighbors and thus obtain good approximation of bandwidth usage.

In IEEE 802.11 MANETs, due to the contention based channel access, a node can only transmit data packets after it gains the channel access. Hence, a node first listens to the channel and estimates bandwidth by using the idle and busy times for a predefined interval. This is expressed in following equation

$$BW = \frac{T_{idle}}{T_{interval}} \times C, \quad (17)$$

where T_{idle} denotes the idle time in an interval $T_{interval}$, and C denotes the channel capacity. $T_{interval}$ comprises of the following time periods: idle time of the channel T_{idle}, time taken for actual transmission of the data T_{tx}, time taken for retransmission of packets T_{rtx}, and time taken for backoff $T_{backoff}$. Equation (17) can be rewritten as

$$BW = \frac{T_{idle}}{T_{idle} + T_{tx} + T_{rtx} + T_{backoff}} \times C. \quad (18)$$

The time periods are measured individually and are incorporated in estimating the bandwidth. The path from source to destination will be forwarded through many intermediate links, and the link which is having minimum bandwidth (bottleneck BW) will be selected as Path bandwidth as given in Eq. (19) denoted by PathBW for N intermediate links

$$PathBW = min(BW_i) \quad \forall i = 1 \ldots N. \quad (19)$$

3.2. Route Establishment

OQSMR is an enhancement of ODMRP, since it is designed to reduce repeated usage of control packets, so that bandwidth consumption can be reduced. There are incorporated changes in structure of ODMRP route request (Join Query) and route reply (Join Reply) packets along with forwarding mechanism of route request packets. The databases for routing include QoS and Stability factors. The request and reply packets include QoS and stability factors.

Route establishment process of OQSMR makes use of parameters like SFBN, delay estimation and available bandwidth information at each node. It considers a stability and QoS database at each node for route request propagation and path(s) finding between source to multicast receivers. The scheme also uses a routing information cache at each node that facilitates route finding by providing path information. This will reduce route request propagation overheads. This section presents stability and QoS database (NSQB), route request (RR) packets, route reply (RP) packets, route error (RE) packets, and routing information cache (RIC).

3.2.1. Neighbor Stability and QoS Database

When a node establishes connections with its one hop neighbors, it maintains a database. This database contains information regarding neighbors that include: id of neighbor, its SFBN, bandwidth and delay values.

To explain the fields of the NSQB, let's consider the network topology given in Fig. 2, where S, A, B, C, R1, and R2 are the nodes connected in the network. S is the source

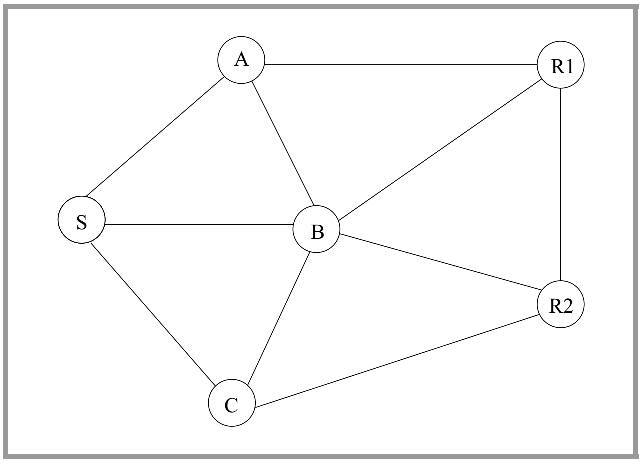

Fig. 2. Network topology.

Table 2
Neighbor Stability and QoS Database (NSQB)
at source node S

Neighbor id	SFBN	BW [Mb/s]	d_{Total} [ms]
B	0.58	2.2	10
C	0.6	2.4	9.8
...	...	...	...
...	...	...	...

node, R1 and R2 are the receiver nodes and remaining are the intermediate nodes. The links between two nodes which are having SFBN below the SFTH will not be selected. The authors have found through simulation that SFTH with values between 0.5 to 0.9 and end-to-end delay threshold of 100 to 200 ms yield better PDR and reduced latency. Table 2 shows a typical neighbor information table for source node S. The information in the table are neighbor id, SFBN, available estimated bandwidth (BW), and single hop delay, d_{Total}.

3.2.2. Route Request, Route Reply and Route Error Packets

To create a multicast stable QoS route in a MANET from source to group of receivers, various control packets such as route request (RR), route reply (RP) and route error (RE) packets are used. In this section, some of the control packet components required for multicast stable QoS path creation are described, and handling link failure situations are shown. Some important fields of RR packet are:

- Source address – it is the address of the source from where the path has to be established to the multicast receivers. It originates the packet.
- Multicast receivers address – group of receivers address where packet has to be forwarded. It helps in accommodating the routes created by RR packets and RP packets.
- Time to live – it is the number of hops RR packet can travel. The value is decremented by one every hop.
- Next hop address – it is the address of the neighbor connected with in the transmission range for propagating RR and RP packet.
- Sequence number – the sequence number assigned to every packet delivered by the source that uniquely identify the packet. It is used to avoid multiple transmission of the same RR packet.
- Route record – it has the addresses of the visited previous nodes recorded in visiting sequence. This information will be used during the return journey to RR packet originator by corresponding RP packet.
- SFBN record – it has the values of SFBN associated with each link which are visited in sequence from the source to group of receivers. This will help in finding PathSFBN, which will be used by RP packet to update RIC.
- Available bandwidth record – it is the estimated available bandwidth value associated with each link visited in sequence from source to group of receivers. This will help in finding path available bandwidth, which will be used by RP packet to update RIC.
- Delay record – it is the estimated delay associated with each link visited in sequence from source to group of receivers. This will help in finding the total path delay, which will be used by RP packet to update RIC.
- Application bandwidth requirement – it is bandwidth required by an application at the source node.

RP packet format for multicast creation is almost similar to RR packet with few changes. The changes in RR packet to convert it into RP packet are as follows. When RR packet reaches any of the group receivers, source address and receiver address are interchanged, SFBN record will be replaced by PathSFBN, bandwidth record will be replaced by path available bandwidth, delay value will be replaced by the end-to-end delay and contents of route record will be reversed. RP packet from group of receivers will be sent to source on a route given in its route record.
RE packet is generated when a node is unable to send the packets. Some of the fields of this packet are source address, receivers address, sequence number. Whenever a node identifies link failures, it generates RE packet to either source or nearest receiver. If link failure occurs in forward journey of a RR packet (from source to multicast receivers), RE packet is sent to the source. On the other hand if link failure occurs for reverse journey of the RP packet (from particular receiver to the source), RE packet is sent to that receiver. Nodes receiving RE packet updates their route information cache by removing paths having failed links and also examine its route cache for an alternate path. If an alternate path is found, it modifies the route, otherwise packet is dropped.

3.2.3. Routing Information Cache

Routing Information Cache (RIC) is used to store the latest routes to group of receivers learned through RR and RP packets. This avoids unnecessary route discovery operation each time when a data packet is to be transmitted. This reduces delay, bandwidth consumption, and route discovery overhead. A single route discovery may yield many routes to the group of receivers, due to intermediate nodes replying from local caches. When source node learns that a route to a particular identified receiver is broken, it can use another route from its local cache, if such a route to that receiver exists in its cache. Otherwise, source node

Table 3
Routing Information Cache at source node *S*

Receiver's address	Path information	PathSFBN	RPathBW [Mb/s]	Delay [ms]	Rec-Timestamp [H:Min:Sec]
R1	S-A-R1	0.6	1.8	100	0:0:0.4
	S-C-R2-R1	0.8	1.6	120	0:0:0.6
R2	S-C-R2	0.7	1.0	89	0:0:0.8
...	...	...	...	...	...
...	...	...	...	...	...

initiates route discovery by sending a route request. Use of RIC can speed up route discovery and it can reduce propagation of route requests. The contents of RIC will be removed at every periodic interval, if it is not updated for certain time (may be 180 to 360 s).
Each node in the network maintains its own RIC that aids in forwarding packets to neighbors. For every visited RP packet at a node, RIC is updated by using some of the fields in RP packet required for establishing stable QoS paths. Table 3 presents a typical RIC at node S for topology given in Fig. 2. Various Fields in the table are explained as follows:

- Receivers address – it is the address of the node where packet has to be forwarded (extracted from RP packet destination address and route record). It helps in accommodating the routes for RR packets.
- Path information – it represents a complete path (a sequence of links).
- PathSFBN – it is the combined stability factor of path as given in Eq. 8.
- Delay – it is the end-to-end delay to meet the total delay constraint of the application as given in Eq. 16, and it must be less than the threshold value.
- RPathBW – it is the remaining path bandwidth which is the difference of PathBW and application bandwidth.
- Recorded timestamp – it contains the time at which RIC is updated by using RP packet.

3.3. Route Discovery Process

Multicast stable QoS path creation involves two phases: a request and a reply phase. Request phase invokes route discovery process to find routes to group of receivers using stable and QoS intermediate nodes. Reply phase involves updating of RIC and conforming the routes found in request phase. Stable nodes are the one who satisfy stability criteria based on our module given in Subsection 3.1 as well as accommodate bandwidth and delay requirement of application. These stable and QoS nodes act as intermediate nodes that help to create multicast mesh from source to group of receivers.

3.3.1. Request Phase

This section presents the process of request phase, reply phase, and route maintenance that helps in discovering a path.
A source node finds the route to its group of receivers by using RR packets. The sequence of operations that occur are as follows:

1. Source node prepares a RR packet with application bandwidth and delay requirements.
2. Selective transmission of RR packet to neighbors who satisfy stability criteria, i.e., SFBN greater than SFTH, and bandwidth requirement, i.e., estimated bandwidth greater than twice the application requirements.
3. A node receiving RR packet will discard it, if it is already received (by using sequence number and source address).
4. If RR packet is not a duplicate, checks RIC for availability of route; if available, RP packet will be generated and start reply propagation to source.
5. If RR packet is a duplicate, then discard it and stop transmission of RR packet.
6. If not duplicate and no route available in RIC, transmit the RR packet by updating its fields (route record, SFBN record, bandwidth record, delay record, time to live, and nexthop address) to its neighbors as in step 2.
7. Perform steps 3 to 6 until destination is reached.
8. If receiver is not reached within certain hops, send RE packet to the source node.

Figure 3 illustrates the basic operation of route request phase for the network topology of Fig. 2:

- Source node S prepares a RR packet with application bandwidth and delay requirements.
- Broadcasts RR packet to discover the routes to multicast receivers R1 and R2.

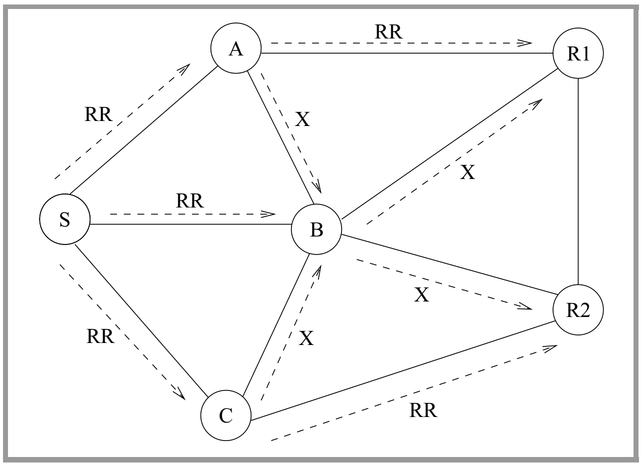

Fig. 3. Route request paths from S to R1 and R2.

- Nodes A, B and C receive RR packet from source S, with assumption that they satisfy the SFBN, BW and delay requirements.
- Check RIC for availability of route at A, B and C to R1 and R2.
- Node A broadcasts RR packet to R1 and B. Node C broadcasts to B and R2. Node B broadcasts to A, R2, R1 and C.
- Node B finds that the packets received through A and C are same as that received by S. Thus duplicate packets are eliminated, as indicated by cross mark in Fig. 3. Similar elimination of duplicated packets are done at nodes A and C which are being received by B.
- R1 and R2 eliminates duplicate packets from nodes B and C respectively.
- If A, B and C have no direct routes to R1 and R2, they update and modify the RR packet (for route record, SFBN record, BW record, end-to-end delay, Time to live and next-hop add) and transmit to next forwarding group of nodes.
- As R2 and R1 are the receiver nodes, they updates RIC and modify the RR packet.
- Finally now, R1 and R2 have paths to the source S: R1-A-S, R1-B-S, R2-C-S, and R2-B-S.

3.3.2. Reply Phase

Multicast receivers initiates the reply phase. When RR packet reaches the receiver node, following operations are performed in the reply phase.

1. RP packet is generated from RR packet by performing following changes in RR packet; receiver and source node addresses are interchanged, route record is reversed, update SFBN record with PathSFBN, update bandwidth record with PathBW and delay record with end-to-end delay.
2. Update RIC at receiver node with receiver id, path information, PathSFBN, PathBW, delay and time.
3. RP packet is forwarded to nexthop node as per the route record if PathBW, and end-to-end delay are satisfied.
4. Node receiving RP packet checks whether available PathBW is greater than application requirement, and end-to-end delay less than the delay threshold, if so, updates RIC by using contents of RP packet. Updates will happen only if current time is greater than the time recorded in RIC. If bandwidth is not available, and end-to-end delay not less than the threshold, send RE packet to receiver and visited intermediate nodes to stop RP packet propagation.
5. Perform steps 3 and 4 until source is reached.
6. If source is not found due to link breaks, send RE packet to the receiver.
7. The source node chooses one of the received paths with higher bandwidth availability and delay with lesser time and keeps other paths as backup paths.

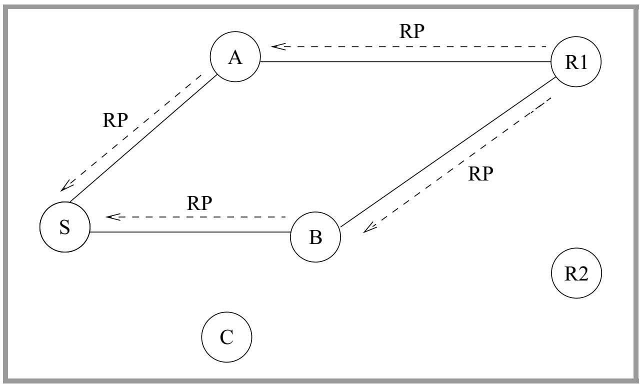

Fig. 4. Reply paths from R1 to S.

Figure 4 illustrates the basic operation of reply phase from receiver R1 to source S, for the network topology of Fig. 2.

- Receiver node R1 prepares RP packets for the RR packets in two directions R1-A-S and R1-B-S.
- Route for one RP packet is R1-A-S and for other RP packet is R1-B-S. PathSFBN, PathBW and delay in the RP packets are updated.
- Both the RP packets are assumed to flow through the paths and reach the source S. The visited intermediate nodes will update paths to node A and B in their RIC's.
- RIC at node S will be updated after receiving RP packets in both directions.

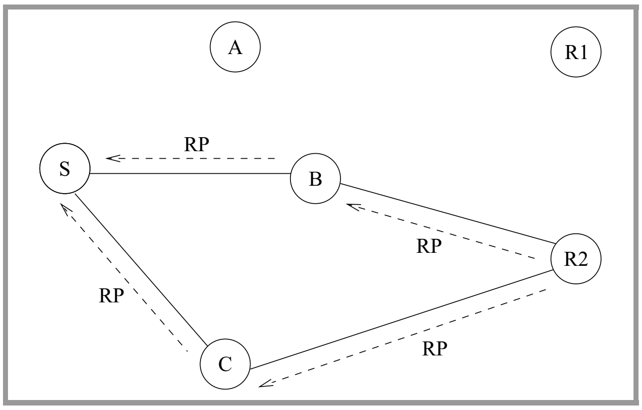

Fig. 5. Reply paths from R2 to S.

Figure 5 illustrates the basic operation of reply phase from receiver R2 to source S, for the network topology of Fig. 2.

- Receiver node R2 prepares RP packets for the RR packets in two directions R2-B-S and R2-C-S.
- Route for one RP packet is R2-B-S and for other RP packet is R2-C-S. PathSFBN, PathBW and delay in the RP packets are updated.
- Both the RP packets are assumed to flow through the paths and reach the source S. The visited intermediate nodes will update paths to node B and C in their RIC's.
- RIC at node S will be updated after receiving RP packets in both directions.

The mesh structure created between source S and group of receivers R1 and R2 in our example with A, B and C as forwarding nodes is given in Fig. 6. In OQSMR, selection of stable forwarding nodes plays an important role in creating mesh structure which satisfies stability, bandwidth and delay requirements. A forwarding node always checks for higher value of the stability factor, minimum bandwidth and less delay. Thus created mesh is the reliable and robust structure which can be used for multimedia real time application.

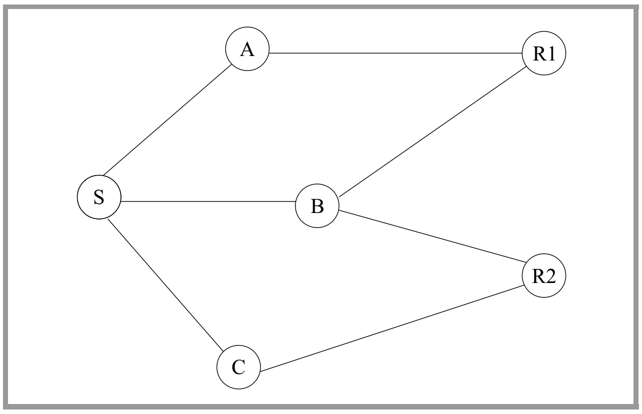

Fig. 6. Mesh created between source S and receivers R1 and R2.

3.4. Route Maintenance

Route maintenance is required in case of link failures. There are three cases: link failure between stable intermediate nodes, between source and stable intermediate node, and between receivers and stable intermediate node. The problem can be tackled in following ways. In case of link failure between two stable intermediate nodes, the node detecting failure condition will use RR and RP packets to find stable QoS path between itself and the receiver. The new path from intermediate node to destination will be informed to source. If a new path is not found, the node sends RE packet to source to rediscover the paths. In case of link failure between source and stable intermediate node, source node will probe backup path, if it is working, it will use backup path. Routes will be rediscovered if backup path does not exist. In case of link failure between receiver and stable intermediate node, the intermediate node will use RR and RP packets to discover paths to receiver from itself and informs the source about the path. If route is not discovered, the node sends RE packet to source to initiate route rediscovery. The source constructs a new path in all the cases for further routing of packets.

4. Simulation and Performance Evaluation

In this section, the performance of proposed protocol with ODMRP [4], and E-ODMRP [7] is compared, through extensive set of simulations. These protocols have been taken for comparison because both are mesh based. These protocols are compared in terms of packet delivery ratio, control overhead, and average end-to-end delay. Simulation considers the values of the performance parameters taken for several iterations, and the values are used for computing the mean. The values lying with in 95% of the confidence interval of the mean are used for computing the mean value, which are plotted in the graphs in result analysis section.

The various network scenarios have been simulated using discrete event simulation model developed by C programming language. Simulation environment consists of four models: Network, Channel, Mobility, and Traffic. In network model an ad hoc network is generated in an area of $l \times b$ square meters. It consists of N number of mobile nodes that are placed randomly within a given area. The coverage area around each node has a limited bandwidth that is shared among its neighbors. It is assumed that, the operating range of transmitted power and communication range r are constant.

Channel Model assumes free space propagation model and error free channel. To access the channel, ad hoc nodes use Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) media access protocol to avoid possible collisions and subsequent packet drops is used. In mobility model: a random way-point (RWP) mobility model based upon three parameters: speed (Mob) of movement, direction for mobility and time of mobility. In RWP, each node picks a random destination uniformly within an underlying physical space, and nodes travel with a given speed. The node pauses for a time period Z, and the process repeats itself. The traffic model is a constant bit rate model that transmits certain number of fixed size packets at a given rate.

4.1. Performance Parameters

Following metrics have been used to analyze the performance:

– Packet Delivery Ratio (PDR) – it is the ratio of number of average data packets received at the multicast receivers to the number of data packets sent by the source;

– Packet Overhead – it measures the ratio of control packets sent to the network to the total number of average data packets delivered to the receivers;

– Average end-to-end Delay – it is the average delay experienced by the successfully delivered packets in reaching their receivers.

Simulation parameters used are summarized in Table 4.

Table 4
Simulation parameters description

Parameter name	Value
Topology	1000 m × 1000 m flat-grid area
Number of nodes	50
Multicast group size	10–50
Number of sources	1–6
Node placement	Random
Mobility model	Random way-point
MAC layer	IEEE 802.11 DCF
Channel capacity	2 Mb/s
Transmission range [m]	250
Carrier-sense range [m]	500
Antenna type	Omnidirectional
Node speed [m/s]	1–50 m/s
Traffic type	CBR
Packet size [bytes]	512
Traffic rate [packets/s]	4 to 32
Minimum bandwidth [Kb/s]	40
Maximum delay [s]	0.1
SFTH [Min.]	0.5
SFTH [Max.]	0.9
Simulation time [s]	500

4.2. Simulation Procedure

Simulation procedure for the proposed scheme is as follows:

1. Generate ad hoc network with given number of nodes.

2. Estimate stability factor based on self node stability and neighbor node stability.

3. Compute link stability factor using Table 1 and Lsd.

4. Compute bandwidth at each node to satisfy application requirement.

5. Update NSQB at each node considering their neighbors.

6. Initiate Route Discovery Process using RR, RP and RE, and accordingly update RIC.

7. Establish the path(s) from source to receivers, and send the data packets,

8. Compute performance parameters of the system.

4.3. Result Analysis

Effect of multicast group size. OQSMR performs better than ODMRP, and E-ODMRP in terms of PDR for the multicast group size (10 to 50) as shown in Fig. 7. The reasons for achieving high PDR (around 95%) in OQSMR are, use of high stable nodes, avoiding nodes with higher delays, and long duration links, and maintaining route cache at every node which avoids unnecessary route discovery. The performance analysis of packet overhead against number of multicast group size is shown in Fig. 8. The overhead is

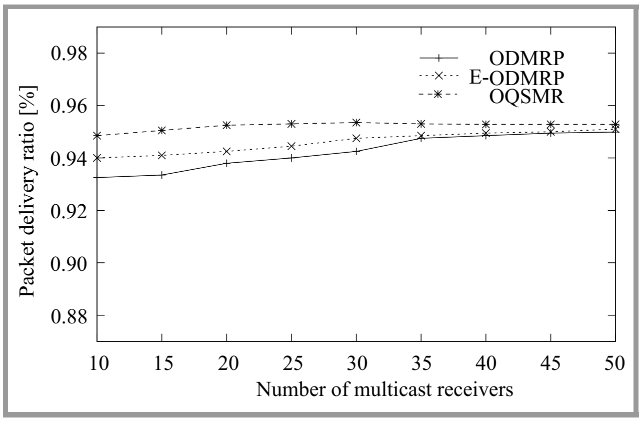

Fig. 7. Packet delivery ratio vs. multicast group size (1 multicast group, 1 source, and 20 m/s maximum speed).

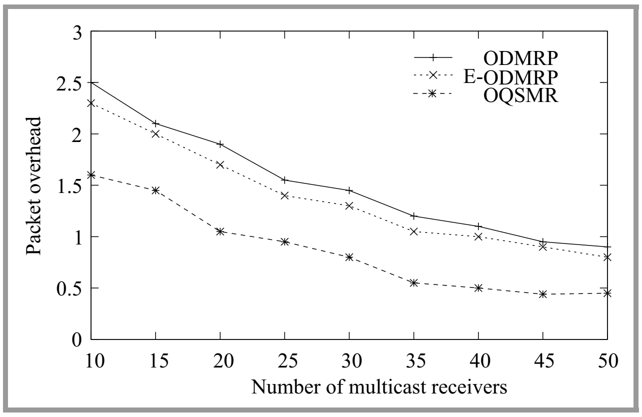

Fig. 8. Packet overhead vs. multicast group size (1 multicast group, 1 source, and 20 m/s maximum speed).

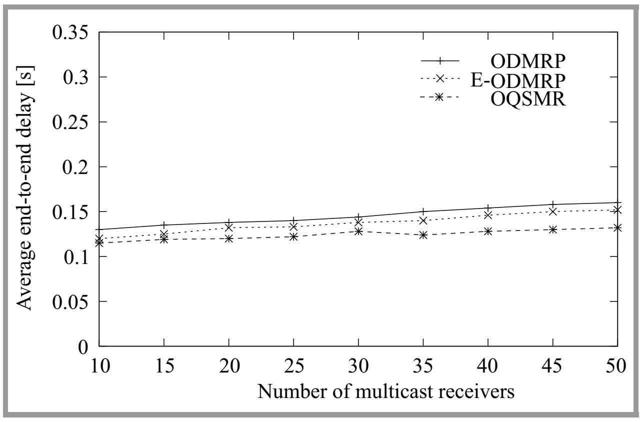

Fig. 9. Average end-to-end delay vs. multicast group size (1 multicast group, 1 source, and 20 m/s maximum speed)

reduced in OQSMR compared to ODMRP and E-ODMRP. The following reasons are given to claim the reduced overhead: strong mesh creation through stable nodes and longer lifetime of links, maintenance of route cache to store the latest routes to group of receivers, avoids unnecessary route discovery, and more efficient forwarding mechanism is created when multicat group size increases. From Fig. 9, OQSMR exhibits lower average end-to-end delay can be observed that compared to ODMRP and E-ODMRP because the multicast traffic is initiated through the nodes those come in non-congested areas, and links established through such stable nodes will have higher link lifetime.

Effect of multicast traffic load. Figures 10–12 exhibit the effect of increase in traffic load on network performance. The sending packet rate varies from 4 to 32 per second with a fixed packet size of 512 bytes, for one multicast source and 20 receivers in the multicast group. The maximum node movement is considered as 20 m/s.

Figure 10 depicts degradation in performance when packet sending rate is increased. High packet sending rate causes higher congestion and packet loss in the network. Results reveal that OQSMR outperforms compared to ODMRP and E-ODMRP. This is because, in OQSMR, nodes avoid intensive flooding of query messages. The direct implication

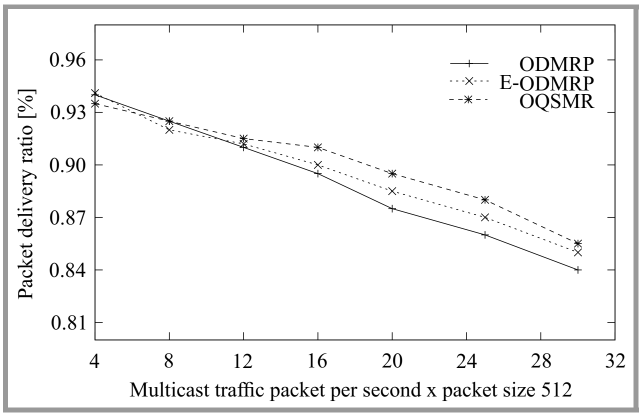

Fig. 10. Packet delivery ratio vs. multicast traffic (1 source, 20 multicast receivers and 20 m/s maximum speed).

is that more bandwidth is allocated to the nodes, and hence packet loss can be reduced. Furthermore, it improves the end-to-end delay as packet sending rate increases.

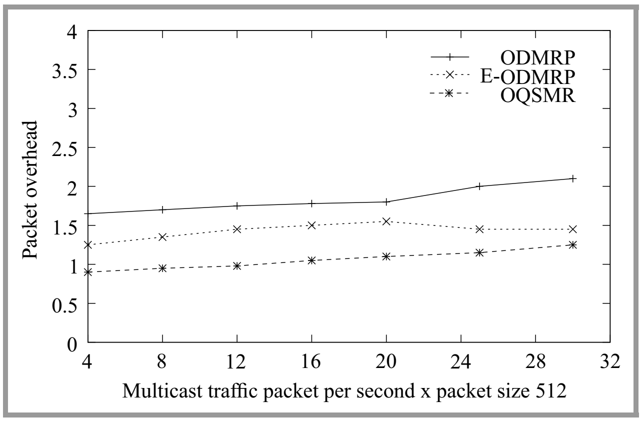

Fig. 11. Packet overhead vs. multicast traffic (1 source, 20 multicast receivers and 20 m/s maximum speed).

As depicted in Fig. 11, packet overhead in OQSMR remains lower than that of ODMRP and E-ODMRP, because it greatly reduces the cost of discovery mechanism due to the mesh architecture created among stable nodes. Figure 12 of the result analysis shows average end-to-end

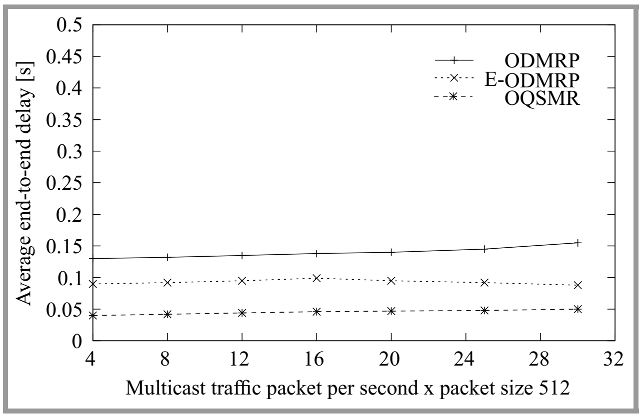

Fig. 12. Average end-to-end delay vs. multicast traffic (1 source, 20 multicast receivers and 20 m/s maximum speed).

delay against multicast traffic. As the sending traffic rate increases from 4 to 32 packets per second (for a fixed packet size of 512 bytes), ODMRP progresses slightly in upward direction which indicates increase in average end-to-end delay. This is because of extensive increase in the query messages at higher traffic load and service time delay among contributing nodes. It is relatively less in E-ODMRP and OQSMR. Presented protocol shows reduced end-to-end delay compared to other two protocols, since the query messages and their service time is reduced at high traffic load.

Effect of number of multicast sources. Figure 13 illustrates the effects of multicast sources on packet delivery ratio when a single multicast group is considered. The number of multicast source nodes from 1 to 6 is varied and keeping the number of receiver nodes as 20. Although mesh structure of routing protocols provides good delivery ratio, it suffers from poor packet delivery ratio in scenarios where multiple sources generate multicast traffic.

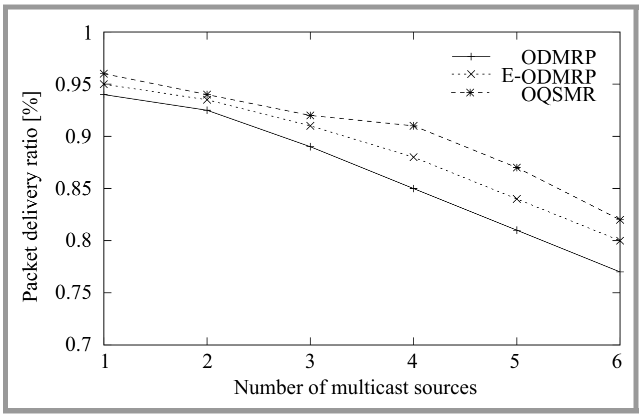

Fig. 13. Packet delivery ratio vs. multicast sources (1 multicast group, 1–6 sources, 20 multicast receivers and 20 m/s maximum speed).

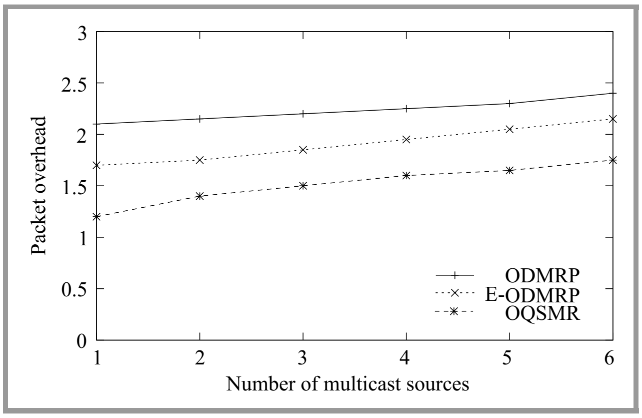

Fig. 14. Packet overhead vs. multicast sources (1 multicast group, 1–6 sources, 20 multicast receivers and 20 m/s maximum speed).

This will create congestion and packet loss within the network. It is observed that OQSMR has relatively higher PDR compared to ODMRP and E-ODMRP when number of sources are increased. Results in Fig. 14 reveal that presented method induces relatively lower packet overhead as the number of traffic sources increase compared to ODMRP and E-ODMRP. High packet overheads under high traffic loads are observed in ODMRP and E-ODMRP. This is because in scenarios where the number of multicast sources increase, a large number of request messages are injected into the network by non-active forwarding nodes resulting in higher network congestion and packet overhead.

Effect of node speed. Figures 15 to 17 show the effect of mobility on the performance of routing protocols. The maximum node speed varies from 1 to 50 m/s for 20 multicast receivers. The speed of 30 to 50 m/s can be applicable to class of MANETs such as VANETs (Vehicle Ad hoc Networks). Basically, the mesh nature and path redundancy in multicast based routing protocols compromise frequent link breakage. This is true in scenarios where the nodes move with high speed. The fault tolerance capabilities keep packet delivery ratio high by creating multiple forwarding routes and avoiding high packet loss rate due to link breakage. Figure 15, shows that OQSMR performs relatively better than ODMRP and E-ODMRP in

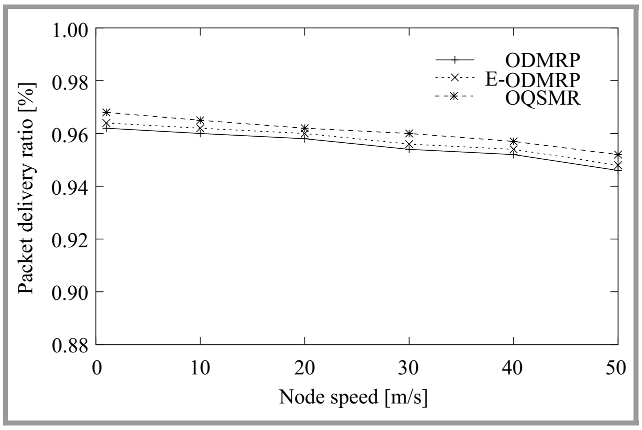

Fig. 15. Packet delivery ratio vs. node speed (1 group, 1 source, 20 multicast receivers).

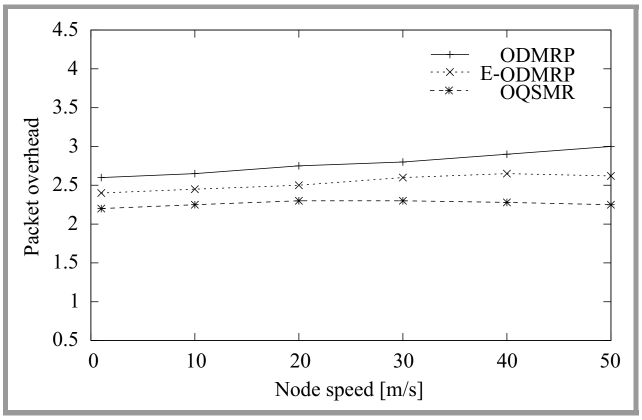

Fig. 16. Packet overhead vs. node speed (1 group, 1 source, 20 multicast receivers).

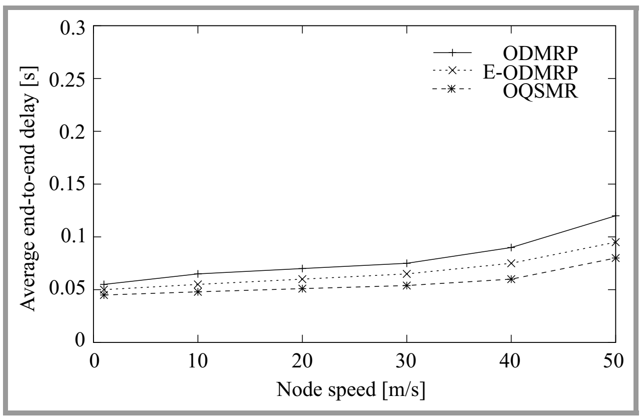

Fig. 17. Average end-to-end delay vs. node speed (1 group, 1 source, 20 multicast receivers).

terms of PDR with variation in node speed. This is because, path constructing techniques used in OQSMR employs stable nodes and stronger links. Figure 16 represents the routing overhead as a function of node mobility. As indicated, the gap between ODMRP to E-ODMRP and E-ODMRP to OQSMR is relatively large at nodes' high speed. The non-stable forwarding nodes impose frequent message rebroadcasting, which effects the performance of ODMRP and E-ODMRP compared to OQSMR. Average end-to-end delay performance with increase in node speed is better in OQSMR as compared to ODMRP and E-ODMRP, as depicted in Fig. 17.

5. Conclusions

Node's and link stability, delay, bandwidth are the important reliability and QoS metrics among several parameters for providing an efficient, low overhead QoS support for mesh based multicast routing in Mobile Ad hoc Networks. In this paper, an on demand QoS and stability based multicast routing (OQSMR) is proposed which is an enhancement of ad hoc on demand multicast routing protocol (ODMRP) to provide stable connection and QoS support for real time applications. The general conclusion from presented simulation experiments reveals that proposed OQSMR routing protocol performs better than ODMRP and E-ODMRP in terms of packet delivery ratio, packet overhead, average end-to-end delay as a function of varying number of receivers, sources and nodes speed.

In future works, the authors aim to study more by comparing our On-demand QoS and Stability based Multicast Routing (OQSMR) protocol with some more QoS based routing protocols in MANETs. The work can be extended by considering delay distribution among nodes in the path such that request packets may not be forwarded, if node/link delay does not satisfy the required node/link delay, and work out jitter based model at the nodes such that scheme must choose a node with less delay jitters. The plans cover also to work on any cast routing protocols to check the efficiency under high throughput applications, e.g. multimedia applications by employing negotiation parameters in route request packet in finding nearest server through non congested paths.

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Paper

Design of a Superconducting Antenna Integrated with a Diplexer for Radio-Astronomy Applications

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Abstract—This paper presents the design of a compact front-end diplexer for radio-astronomy applications based on a self complementary Bow-tie antenna, a 3 dB T-junction splitter and two pass-band fractal filters. The whole diplexer structure has been optimized by using an evolutionary algorithm. In particular the problem of the diplexer design is recast into an optimization one by defining a suitable cost function which is then minimized by mean of an evolutionary algorithm namely the Particle Swarm Optimization (PSO). An X band diplexer prototype was fabricated and assessed demonstrating a good agreement between numerical and experimental results.

Keywords—diplexer, microwave antenna, optimization techniques, particle swarm algorithm, radio-astronomy.

1. Introduction

In the last years there has been a growing demand of wireless services that has led to an overexploitation of the available radio frequency resources in terms of channels and frequency bands availability. This has forced researchers to find strategies to protect the frequency bands typically used for radio-astronomy applications by introducing suitable filters able to remove or strongly reduce interference signals, while keeping at the same time the weak signals integrity received by the radio-telescopes. Moreover it is necessary to limit the bandwidth to the frequency range of interest to minimize the external noise received by the detector. In this frame it is mandatory to use filters characterized by a high selectivity and low losses. This can be easily accomplished with circuits based on superconducting materials. Most of time low critical temperature superconductors (LTS) are used for the detector and its surrounding circuitry, that are both cooled at, or below, the temperature of liquid helium. High critical temperature superconductors (HTS) can also be used in the THz range, when the frequency is above the gap frequency of LTS [1] since, in this frequency range, typically above 700 GHz for niobium, LTS exhibit too many losses for sensitive detection. On the other hand the design of superconducting passive devices can be first obtained by considering copper microstrip structures and commercial dielectric material with the goal of testing the design methodology at room temperature before moving toward a cryogenic design. With such an approach, one has to keep in mind that, for a given geometry and physical parameters like the dielectric constant of the substrate and insulating layers, the kinetic inductance of superconducting films [2] is not properly taken into account, which results in different propagation velocity and characteristic impedance of transmission lines. This effect is particularly significant for thin film microstrip designs, while, most of time, coplanar designs give closer results between normal metal and superconducting films designs [3]. The use of a superconducting diplexer equipped with efficient antennas and filters able to select simultaneously two or more frequency bands of the signal is of interest for Cosmic Microwave Background (CMB) observations relying on the spectral modification of the Planck law by the Sunayev-Zeldovich (SZ) effect [4]. Self-complementary bowtie antennas [5] seem to be appropriate candidates for achieving good performances since their use, combined with fractal geometries for filters, has been proven very efficient to achieve miniaturization, enhanced bandwidth and good performances [6], [7]. Indeed, the miniaturization of the antenna front-end is important for future multi-pixel microwave receivers based on imaging arrays. The proposed compact broadband diplexer is a complex system, and conventional microwave design techniques are quite effective only for the design of basic microwave active as well as passive devices [8]–[11]. These techniques are not able to models the interactions between the different components of complex systems with efficacy, usually a final tuning that could dramatically increase the costs of the device, and increase the number of design/fabrication cycles, is mandatory to obtain working devices. In last years microwave CAD tools [12]–[14] have been proposed for the design of complex microwave systems, have been successfully adopted in many areas of applied electromagnetism such as antenna design [15]–[16], control [17] and other interesting applications [18]–[22]. In fact, these tools can analyse, design and modify, microwave devices in an unsupervised manner. Certainly they can't completely replace an experienced microwave engineer but they can offer a help the designer to strongly reduce the time necessary to design complex microwave systems. In these tools, the design problem is usually recast as an optimization problem that can be handled by means of a suitable optimization algorithm and a suitable cost function. The latter represents the distance between the required performances and the obtained trial solution. These design tools usually consist of an optimizer and a commercial numerical simulator, and in recent years they have been integrated into commercial microwave simulators.

This work presents the optimized synthesis [23] of a superconducting diplexer based on a broadband self-complementary Bow-tie antenna and two fractal passband filters that operate at two frequency bands centered at 10 and 14.66 GHz respectively. The optimization of the receiving structure is carried out considering a numerical procedure based on a PSO algorithm [24]. The key of force of this method is that it takes into account the different interactions and coupling phenomena, always present when a complex microwave system or circuit is developed. At the end of the optimization procedure the proposed method provides, not only the design of the single devices of the diplexer but a complete systems where the requirements of each microwave component, namely the antenna, the filters and the splitters respect the initial system requirements. An experimental prototype has been designed, fabricated and assessed experimentally.

The paper is organized as follows. Section 2 reports a detailed description of the proposed superconducting broadband antenna integrated with the diplexer structure. Section 3 summarize the optimization procedure based on the particle swarm optimizer (PSO) adopted for the design of the diplexer structure. In Section 4 an experimental diplexer prototype, obtained with the design procedure described in Section 3, will be designed, fabricated and experimentally assessed. Finally in Section 5, conclusions are drawn and areas for future works are examined.

2. Superconducting Antenna and Diplexer Structure

The receiving diplexer structure proposed in this work is reported in Fig. 1, it consists of a self-complementary Bow-tie antenna [5], a 3 dB T-junction splitter and two bandpass filters based on fractal resonators. The T-junction and the two filters, implement a two-ports diplexer. The proposed microwave system must be able to receive an incoming electromagnetic signals in a frequency band between 9 and 15 GHz, to reach this goal a broad band antenna is mandatory, in particular in this work a self complementary Bow-tie antenna is considered. The main advantage of a self-complementary antenna is the constant impedance independent of the source frequency. In particular the input impedance of a self complementary antenna is given by the well known Mushiake formula $Z_a = Z_m/2$ where Z_a is the antenna impedance, and Z_m is the intrinsic impedance of the medium. As it can be noticed from the Fig. 1, the incident electromagnetic wave impinges on the self-complementary antenna and it is equally splitted by means of a 3 dB T-junction microstrip power splitter [8]. A T- junction 3 dB power splitter has been chosen because it is a lossless and broadband device. Half of the signal power travels on the left side of the circuit and it is filtered

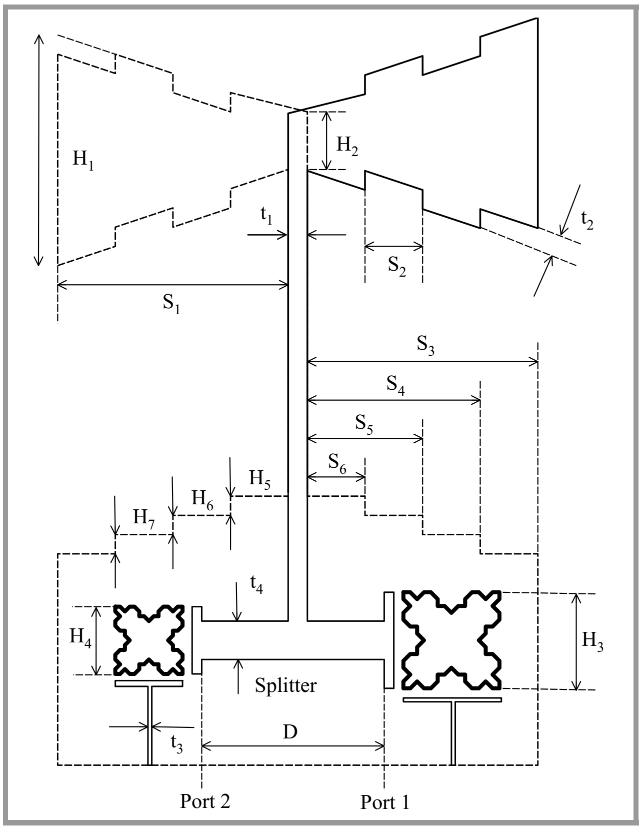

Fig. 1. Geometry of the diplexer structure under study.

by the first band-pass fractal filter which presents a bandpass frequency band centered at 14 GHz, then the signal leaves port 1 providing the first channel. The other half power of the signal is delivered toward the second fractal filter, characterized by a band-pass frequency band centered at 10 GHz, and after the filtering it leaves port 2. Both filters present a bandwidth of about 100 MHz. The use of fractal filters permits to strongly reduce the dimensions of the diplexer structure, fractals geometries demonstrated their effectiveness in the design of microwave antennas as in [25]–[27]. In particular a reduction of about $\frac{1}{3}$ has been obtained with respect to standard rectangular or circular resonators based pass-band filters. The two output channels of the diplexer must be connected with two power meters characterized with an input impedance of 200 Ω, and to guarantee a perfect match with a $S_{11} < -10$ dB between the ports of the diplexer, the coplanar microstrip waveguide which connect the antenna, and the 3 dB T-junction power splitter a multi-section broadband transformer has been considered [8]. In the next section the proposed design methodology based on the PSO algorithm will be detailed.

3. Design Methodology Based on the PSO Algorithm

The PSO is a powerful multiple-agents optimization algorithm developed by Kennedy and Eberhart [20] in 1995 that imitates the social behavior of groups of insects and animals such as swarms of bees, flocks of birds, and shoals of fish. PSO has been used to solve complex antenna design and electromagnetic problems [28], [29]. The standard PSO implementation considers a swarm of trial solutions (called particles). Each particle travels in the solution space by improving its position according to suitable updating equations, on the basis of information on each particle's previous best performance (cognitive knowledge) and the best previous performance of its neighbors (social knowledge). For real-number spaces, the trajectories of the particles are defined as changes in the positions on some number of dimensions. With respect to other evolutionary algorithms such as genetic algorithms (GAs) the PSO shows indisputable advantages in particular. The PSO is simpler, both in formulation and computer implementation, than the GA, which considers three-genetic operators (the selection, the crossover, and the mutation). PSO considers only one simple operator, called velocity updating equation. PSO allows an easier calibration of its parameters and, in general, a standard configuration turns out to be adequate for a large class of problems and problem sizes. Consequently there is no need to perform a PSO calibration for every design experiment. PSO has a flexible and well-balanced mechanism to enhance the global (i.e., the exploration capability) and the local (i.e., the exploitation capability) exploration of the search space. Such a feature allows one to overcome the premature convergence (or stagnation) typical of GAs and it enhances the search capability of the optimizer. PSO requires a very small population size, which turns out in a reduced computational cost of the overall minimization by allowing a reasonable compromise between the computational burden and the minimization reliability. The antenna/diplexer system design has been formulated as an optimization problem fixing suitable constraints in terms of impedance matching at the input port ($|S_{11}|$ values). Moreover the two output ports of the diplexer will be connected to a power meter with an input impedance of $Z_{in} = 200\ \Omega$ with $|S_{11}| < -10$ dB. The two frequency bands are respectively centered at 10 GHz and 14.66 GHz with a bandwidth of about 100 MHz each. The design is based on microstrip technology. The geometrical parameters shown in Fig. 1 permit to simultaneously maximize the performance and minimize the antenna size and of the fractal resonators. In particular the diplexer structure is uniquely determined by the following vector $\underline{\Gamma} = \{H_i, i = 1, \ldots 7;\ S_j, j = 1, \ldots 6;\ t_k, k = 1, \ldots 4, D\}$ that describes the geometrical diplexer parameters (Fig. 1). To meet the objectives, a cost function, that represents the difference between the requirements and the performances of a trial diplexer geometry, is defined by:

$$\Phi\{\underline{\Gamma}\} = \sum_{h=1}^{H} \max\left\{0; \frac{\Psi(h\Delta f) - |S_{nn}|_{\max}}{|S_{nn}|_{\max}}\right\}, \qquad (1)$$

where n indicates the port number, Δf is the frequency step in the range between 8 and 15 GHz and the function $\Psi(h\Delta f)$ is the $|S_{11}|$ at the frequency $h\Delta f$ when the trial geometry defined by the $\underline{\Gamma}$ vector is considered, and $|S_{11}|_{req}$ represents the return loss requirement in dB. To minimize Eq. (1) a suitable version of the PSO has been used, in combination with a geometrical generator and a commercial electromagnetic simulator (namely HFSS designer), to estimate the characteristics of the trial diplexer geometries. In particular the minimization of Eq. (1) is obtained by constructing a sequence of trial solutions $\underline{\Gamma}_s^k$ (s being the trial solution index, and k the iteration index $k = 1, \ldots, K_{\max}$) following the strategy of the PSO. The iterative optimization algorithm continues until the stopping criteria are reached, in particular when $k = K_{\max}$ or $\Phi\left(\underline{\Gamma}_s^k\right) < \beta$, where $K_{\max}$ and β are respectively the maximum number of iterations and a convergence threshold. At the end of the iterative procedure the optimal solution defined as $\underline{\Gamma}^{opt} = \arg\{\min[\Phi(\Gamma_k)]\}$ is stored and the obtained geometrical parameters are used to fabricate the diplexer prototype.

4. Numerical and Experimental Assessment

To obtain a diplexer prototype (Fig. 2) the cost function given by Eq. (1) is minimized according to the guidelines given in [24], and a suitable implementation of the PSO [17] has been used in conjunction with a circuital generator, and a microwave circuital simulator able to take into account all the interactions between all subsystems. Starting from each of the trial arrays Γ the PSO, the circuital generator changes the geometrical parameters of each sub-system and then it generates the corresponding system

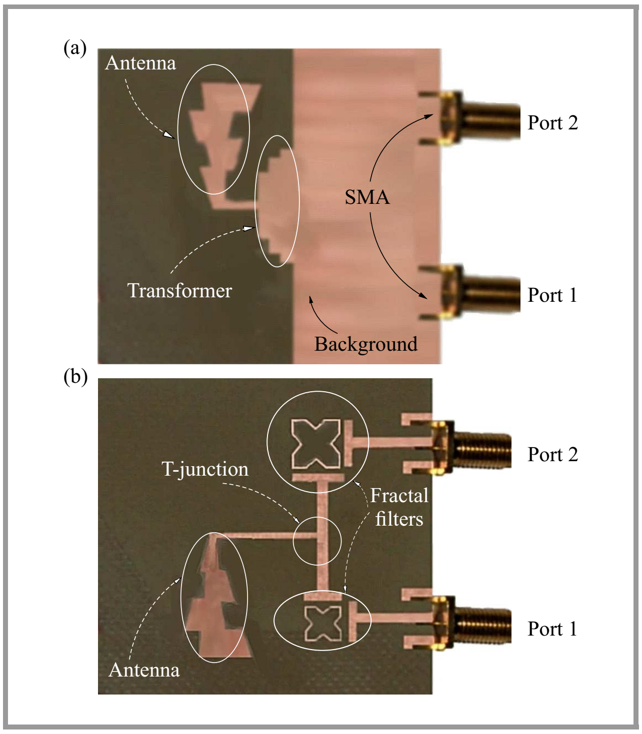

Fig. 2. Photo of the obtained diplexer prototype: (a) top view, and (b) bottom view.

structure. The whole system performance is computed by means of a circuital simulator, which take into account the presence of dielectric substrate, the mutual coupling effects between all the subsystems, and it is used to estimate the cost function (1). The iterative process continues until $k = K_{\max}$ or when a convergence threshold on the cost function is reached. Then the array Γ, that contains the geometrical parameter that define the antenna and diplexer geometrical structure is stored and used for the prototype development. At the beginning of the iterative optimization procedure based on the PSO optimizer, a set of $S = 10$ trial geometrical parameters are randomly initialized and used as starting point for the optimizer. Concerning the specific PSO parameters a population of $S = 10$ individuals, a threshold of $\beta = 10^{-3}$ and a maximum number of iterations $K_{\max} = 100$, and a constant inertial weight $\beta = 0.4$ were used. The remaining PSO parameters have been chosen according to the reference literature [23], [24]. A geometry generator generates a set of trial diplexer geometries that are estimated by means of an electromagnetic simulator, which take into account the presence of the dielectric substrate. Then the cost function is evaluated and, thanks to the PSO strategy, the swarm evolves improving their characteristics. The iterative procedure continues until the maximum number of iterations or the threshold on the cost function is reached. As an illustrative example of the optimization process, Fig. 3 shows the behavior of the cost function versus the iteration number during the optimization of the diplexer geometry. As it can be noticed

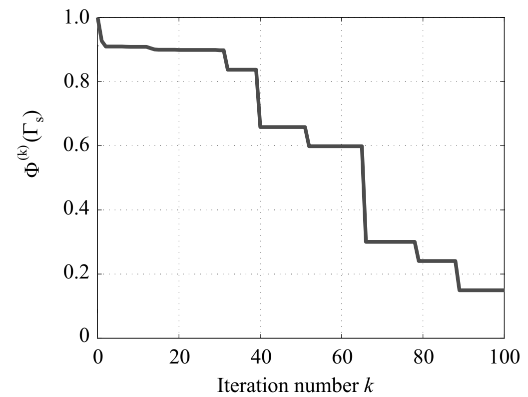

Fig. 3. Behavior of the cost function versus iteration number k.

from Fig. 3, the algorithm end when the maximum number of iteration fixed to $K_{\max} = 100$ is reached. Concerning the computational burden required to accomplish the design of the diplexer, each iteration requires about 60 seconds on a serial machine equipped with 8 gigabytes of memory and four cores processor. The whole design procedure requires about two hours. It is worth noticed that thanks to the intrinsic capabilities of PSO parallelization it is possible to strongly reduces the computational time considering the same techniques proposed in [30], [31]. A diplexer prototype has been fabricated with a milling machine using a dielectric substrate of thickness $t = 0.8$, $\varepsilon_r = 3.28$ and $tan(\delta) = 0.003$. The following geometrical parameters have been considered, $t_1 = 0.8$ mm, $t_2 = 2$ mm, $S_1 = 10$ mm, $H_1 = 7.1$ mm, $D = 15$ mm, $S_3 = S_4 = S_5 = S_6 = 1$ mm, $H_5 = H_6 = H_7 = 1$ mm, $H_4 = 7$ mm, and $H_5 = 9$ mm. Concerning the fractal resonators are made considering the second iteration of the Koch fractal algorithm. Due to mechanical constraints it was not possible to further reduce the dimensions of the resonators by increasing the number of fractal iterations. The photo of the top and bottom side of the prototype is displayed in

Fig. 4. Experimental set-up. Photograph shows the anechoic chamber used for the experimental assessment of the diplexer prototype.

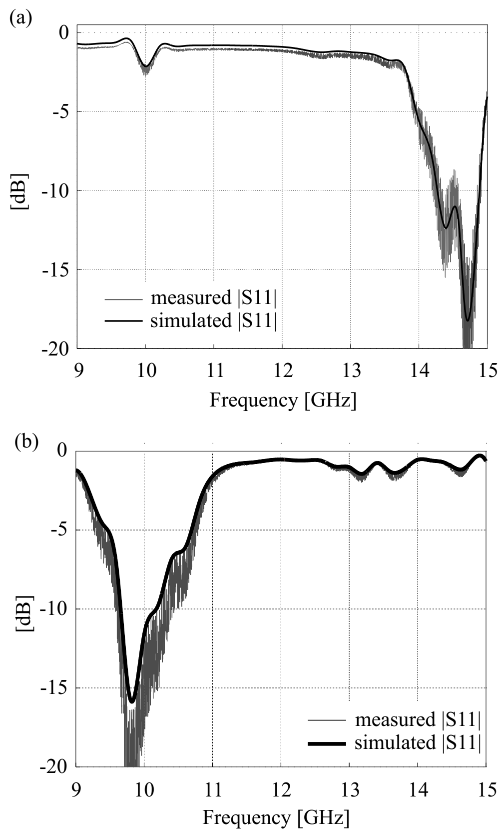

Fig. 5. Behavior of S parameters at the two ports versus frequency. Comparison between numerical and experimental data: (a) return-loss S_{11} at port one, and (b) S_{22} at port two.

Fig. 2. The fabricated prototype has been equipped with two sub-miniature type A (SMA) coaxial connectors. An experimental setup has been arranged inside an anechoic chamber to assess the characteristics of the prototype. The $|S_{nn}|$ parameters were measured at both ports with a network analyzer. The photo of the considered experimental setup arranged inside the anechoic chamber is shown in Fig. 4. The obtained results are reported in Fig. 5. As it can be noticed the obtained results meet the initial requirements: in particular the return loss for the two considered frequency bands is found to be below the initial requirements by about 5 dB at center frequencies. For the sake of comparisons, the measurements have been compared with numerical data obtained with the HFSS software, which has been able to accurately simulate the considered structure, the agreement between numerical and experimental data is quite good, as can be seen in Fig. 5. In particular in the two bands of interest the S_{nn} keeps below -10 dB for the whole range of interest, and the lowest values obtained for the S_{nn} is below -15 dB.

5. Conclusion

In this work the design of a receiving front-end scale model for radio-astronomy applications has been described. The receiver is composed of a broadband self-complementary Bow-tie antenna and a diplexer composed by a 3 dB splitter and two band pass filters. The whole structure has been optimized through a particle swarm algorithm able to act on the geometrical parameters of the Bow-tie self-complementary antenna and of the two passband fractal filters, to comply with the impedance matching constraints. A prototype has been designed, based on microstrip technology, fabricated and experimentally assessed. The comparison between measured and numerical data demonstrate the effectiveness of the proposed design methodology and the potentialities of the proposed front-end structure. The next step will be to scale this design at millimeter wavelengths and include the superconducting properties of the films in the microwave simulations to estimate the final compactness and design performance for future microwave imaging systems.

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