Lecture Notes on Data Engineering and Communications Technologies 27

Isaac Woungang
Sanjay Kumar Dhurandher *Editors*

2nd International Conference on Wireless Intelligent and Distributed Environment for Communication

WIDECOM 2019

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Isaac Woungang • Sanjay Kumar Dhurandher
Editors

2nd International Conference on Wireless Intelligent and Distributed Environment for Communication

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Welcome Message from WIDECOM 2019 General Chair

Welcome to the 2nd International Conference on Wireless, Intelligent, and Distributed Environment for Communication (WIDECOM 2019).

The last decade has witnessed tremendous advances in computing and networking technologies, with the appearance of new paradigms such as Internet of Things (IoT) and cloud computing, which have led to advances in wireless and intelligent systems for communications. Undoubtedly, these technological advances help improve many facets of human lives, for instance, through better healthcare delivery, faster and more reliable communications, significant gains in productivity, and so on. At the same time, the associated increasing demand for a flexible and cheap infrastructure for collecting and monitoring real-world data nearly everywhere, coupled with the aforementioned integration of wireless mobile systems and network computing, raises new challenges with respect to the dependability of integrated applications and the intelligence-driven security threats against the platforms supporting these applications. The WIDECOM conference is a conference series that provides a venue for researchers and practitioners to present, learn, and discuss recent advances in new dependability paradigms, design, and performance of dependable network computing and mobile systems, as well as issues related to the security of these systems.

Every year, WIDECOM receives dozens of submissions from around the world. Building on the success from last year, WIDECOM 2019 presents an exciting technical program that is the work of many volunteers. The program consists of a combination of technical papers, keynotes, and tutorials. The technical papers are peer-reviewed by program committee members who are all experts and researchers, through a blind process.

We received a total of 38 papers this year and accepted 17 papers for inclusion in the proceedings and presentation at the conference, which corresponds to an acceptance rate of about 45%. Papers were reviewed by two PC members, in a single round of review.

WIDECOM 2019 is also privileged to have select guest speakers to provide stimulating presentations on topics of wide interest. This year's distinguished speakers are

- Dr. Elena Pagani, Associate Professor, Computer Science Department, University of Milan, Italy, and Associate Researcher at the Institute for Informatics and Telematics of the National Research Council in Pisa, Italy
- Dr. Francesco Bruschi, Assistant Professor, Dipartimento di Elettronica Informazione e Bio-ingegneria, Politecnico di Milano, Milan, Italy

We would like to thank all of the volunteers for their contributions to WIDECOM 2019. Our thanks go to the authors, and our sincere gratitude goes to the program committee, who gave much extra time to carefully review the submissions.

We are pleased to announce selected papers will be invited to submit extended versions for publication in the *International Journal of Space-Based and Situated Computing* (IJSSC), Inderscience; *International Journal of Grid and Utility Computing* (IJGUC), Inderscience; and *Internet of Things: Engineering Cyber Physical Human Systems*, Elsevier.

We would also like to thank the organizing committee, in particular, the local organizing team Michela Ceria, Luca Casati, and Alessandro De Piccoli, all from the University of Milan, for their support and hard work in making this event a success. Our thanks go to our sponsors:

- Computer Science Department, University of Milan, Italy, for hosting WIDECOM 2019
- Springer, for publishing the conference proceedings

Finally, we thank all the attendees and the larger WIDECOM 2019 community for their continuing support, by submitting papers and by volunteering their time and talent in other ways.

We hope you will find the papers presented interesting and enjoy the conference.

Milan, Italy

Andrea Visconti
WIDECOM 2019 Conference Chair

Welcome Message from the WIDECOM 2019 Program Cochairs

Welcome to the 2nd International Conference on Wireless, Intelligent, and Distributed Environment for Communication (WIDECOM 2019), which was held from February 11 to February 13, 2019, at University of Milan, Italy.

WIDECOM 2019 provides a forum for researchers and practitioners from industry and government to present, learn, and discuss recent advances in new dependability paradigms, design, and performance of dependable network computing and mobile systems, as well as issues related to the security of these systems.

The papers selected for publication in the proceedings of WIDECOM 2019 span many research issues related to the aforementioned research areas, covering aspects such as algorithms, architectures, protocols dealing with network computing, ubiquitous and cloud systems and Internet of Things systems, integration of wireless mobile systems and network computing, and security. We hope the participants to this conference will benefit from this coverage of a wide range of current hop-spot-related topics.

In this edition, 38 papers were submitted and peer-reviewed by the program committee members and external reviewers who are experts in the topical areas covered by the papers. The program committee accepted 17 papers (about 45% acceptance ratio). The conference program also includes two distinguished keynote speeches and three tutorials.

Our thanks go to the many volunteers who have contributed to the organization of WIDECOM 2019. We would like to thank all authors for submitting their papers. We would also like to thank the program committee members for thoroughly reviewing the submission and making valuable recommendations. We would like to thank WIDECOM 2019 local arrangement team for the excellent organization of the conference and for their effective coordination creating the recipe for a very successful conference.

We hope you will enjoy the conference and have a great time in Milan, Italy.

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New Delhi, India — Sanjay Kumar Dhurandher
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WIDECOM 2019 Keynote Talks

From WSNs to VANETs: Paradigms, Technologies, and Open Research Issues for Challenged Networks

Elena Pagani

E. Pagani
Associate Professor, Computer Science Department, University of Milan, Italy
Associate Researcher at the Institute for Informatics and Telematics of the National Research Council in Pisa, Pisa, Italy

Abstract

Wireless technologies are going to revolutionize countless aspects of daily life, touching issues ranging from smart cities to opportunistic networks, from smart vehicles to Industry 4.0. At the base of all these applications, there is the requirement of implementing communication services among devices possibly with scarce resources and also mobile. The purpose of this talk is to analyze the paradigms for mobile ad hoc networking and the characteristics of these infrastructures, to discuss the state of the art of the technologies and solutions available to implement them, and to highlight the research aspects that are still open.

Making Sense(s) of Smart Contracts in a Connected World

Francesco Bruschi

F. Bruschi
Assistant Professor, Dipartimento di Elettronica Informazione e Bio-ingegneria, Politecnico di Milano, Milan, Italy

Abstract

Smart contracts are digital, executable, self-enforcing descriptions of commitments between parts. They were first envisioned in 1994 by Nick Szabo and recently, blockchain-based platforms such as Ethereum, with their very strong guaranties of untampered, deterministic execution, seem to offer an ideal medium for their deployment. Among the main applications of smart contracts is the automatization of interactions such as bets, collaterals, prediction markets, and insurances. One of the main issues that arise in extending the domain of smart contracts regards provisioning reliable information on the blockchain (the so-called oracle problem). In this talk, we will consider the challenges that emerge when using IoT sensors and devices as information providers for smart contracts.

WIDECOM 2019 Tutorials

Tutorial 1: High-Speed Cryptography

Alessandro De Piccoli

A. De Piccoli
PhD student, Computer Science Department, University of Milan, Milan, Italy

Abstract

In today's communications, cryptography is increasingly crucial to ensure the confidentiality of the exchange of data, whether they are related to commercial transactions or simply information. Exploiting the recent optimization of the product between polynomials, especially those having binary coefficients, it is possible to improve the implementation of cryptography algorithms. In fact, when implementing a cryptographic algorithm, efficient operations have high relevance both in hardware and software. Since a number of operations can be performed via polynomial multiplication, the arithmetic of polynomials over finite fields plays a key role in real-life implementations. Through new techniques, we can save time and memory, allowing fast encryption/decryption operations but also faster cryptanalysis. At the same time, this technique may be used by researchers to prevent attacks such as side-channel and timing attacks. The aim of this tutorial is to provide an introductory guide to the mathematical aspects of high-speed cryptography techniques for computer science researchers.

Tutorial 2: Understanding the Key Pre-distribution Aspect of Linear Wireless Sensor Network

Kaushal Shah

K. Shah
Assistant Professor at CGPIT, Uka Tarsadia University, Surat, Gujarat, India

Abstract

There is a set of applications in wireless sensor networks that forms a particular topology through specific placements of sensor nodes. This set is known as linear infrastructure or one-dimensional network. Applications of such networks are subway tunnel or pipeline monitoring and perimeter surveillance. These applications often demand critical security concerns. The distribution of symmetric keys for such networks is different from those that are planned and are widely studied. By considering requirements for such linear infrastructure in detail, we observe that connectivity is an important issue as capturing a single node disrupts the entire network's services. Therefore, we propose a new measure of connectivity that produces the optimal results with acutely lightweight key pre-distribution schemes (KPS). We also show the theoretical analysis of our proposed scheme and prove how it produces optimal results with lesser storage requirements as compared to other schemes. The performance analysis shows that the proposed KPS requires lesser number of keys per node (O(1) constant storage), as compared to the other existing schemes in the literature, to provide the same level of connectivity.

Tutorial 3: Efficient Cryptographic Algorithms for Securing Passwords

Michela Ceria

M. Ceria
Postdoc Fellow, Computer Science Department, University of Milan, Milan, Italy

Abstract

Nowadays, there are several real-life applications that require authentication and so the use of passwords: e-mails, online banking, mobile phones, and so on. A server retaining our passwords must ensure them the highest possible level of security from attacks (e.g., dictionary attacks and brute force attacks), also taking into consideration that the average user is not able to generate a strong password with a suitable number of entropy bits (the average password entropy was estimated at 40.54 bits) and that more than one user likely employ the same password for the same kind of service. This problem can be addressed by means of cpu- and memory-intensive algorithm that slow attackers down. In this talk, we will give an overview of main password hashing algorithms such as Argon2, Catena, Lyra2, Makwa, and yescrypt, explaining their main features and instances.

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Chapter 1
Performance Evaluation of G.711 and GSM Codecs on VoIP Applications Using OSPF and RIP Routing Protocols

Nadia Aftab, Maurin Hassan, Muhammad Nadeem Ashraf, and Akash Patel

1.1 Introduction

VoIP is an emerging technology which involves sending voice packets over an IP network rather than via a public switched telephone network (PSTN)—which uses the traditional circuit switching telephony. VoIP requires low packet loss and low delay to ensure voice quality. Therefore, when implementing VoIP solutions, many factors should be taken into account, including the choice of codec, the routing protocol used, the quality of service (QoS), the delay variation, and the packet loss, just to name a few [1]. Many companies have shifted to VoIP solutions or a hybrid model of PSTN and VoIP in order to save their operational cost by utilizing the data bandwidth. There is a dilemma of designing and optimizing networks to address the ongoing requirement of adding VoIP traffic, which motivated the desire to take a closer look at VoIP solutions.

In real world, an IP-based network consists of various types of traffic, including VoIP, HTTP, FTP, and UDP. In this chapter, our focus is only on VoIP traffic. A key factor for VoIP is QoS, which is meant to ensure that voice packets are not dropped or delayed during the transmission, a call, or an event. VoIP is a transfer of voice packets, which is a real-time application that is very sensitive to delay and jitter. For this reason, QoS has become a very important criterion for VoIP quality. In order to upgrade the quality of voice packet transmission, VoIP QoS can be analyzed and configured using a combination of different parameters such as packet loss, end-to-end delay, echo, and jitter.

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In this chapter, different QoS parameters have been defined and analyzed in order to carry out our simulations. The performance of two different routing protocols (namely, RIP and OSPF) in IPv4 is studied using two different voice encoding codecs, namely G.711 and GSM, and the Riverbed Modeler version 17.5 [2], considering the jitter, end-to-end delay, packet delay variation, and MOS value as performance metrics. Simulation results exhibit the superiority of G.711 over GSM as preferred VoIP solution in terms of the aforementioned metrics.

The rest of the chapter is organized as follows. In Sect. 1.2, some background on Voice Encoding Codecs and considered IP Routing protocols are presented. In Sect. 1.3, related works are discussed. In Sect. 1.4, the network design, configuration parameters, and considered routing scenarios are described. In Sect. 1.5, the performance evaluation of the proposed design is conducted. Finally, Sect. 1.6 concludes the chapter.

1.2 Background

1.2.1 Voice Encoding Codec

Voice transmission is analog, whereas a communication network is digital. The process or algorithm to convert analog voice signal into digital information is performed by an encoder–decoder device (codec). The codec samples the analog signal according to the algorithm into the digital data to be transmitted over a network or the Internet. A codec generally compresses the data to improve the performance by consuming less bandwidth and without degrading the quality (lossless compression). In this process, the data may also be encrypted to achieve security while transmitting it over the network. Hence, a codec is expected to achieve the following three criteria [3]:

- Encoding and decoding
- Compression and decompression
- Encryption and decryption

Many different voice encoding codecs have been proposed in the literature, including G.722, G.729, and G.726 [3]. In this chapter, we only focus on GSM and G.711. Their features are given in Table 1.1.

1.2.2 Routing Protocols

IP routing is an important component of modern data networks. VoIP is a Layer 3 network protocol that uses Layer 2 point-to-point or link-layer protocols [4]. Selecting an appropriate routing protocol is crucial for VoIP network in order to

Table 1.1 Types of voice encoding codec

GSM	G.711
First digital voice encoding standard used in digital mobile phone systems	Narrow band audio codec primarily used in telephony
Also known as full rate	Also known as pulse code modulation (PCM)—a commonly used waveform codec
Quality of coded speech is poor	Provides toll quality audio Passes audio signals in the range of 300–3400 Hz[a] and samples at a rate of 8000 samples/s
Bit rate of codec: 13 kbits/s[a]	Bit rate of codec: 64 kbits/s[a]
It is a good compromise between computational complexity and quality	Lossless data compression with decrease in audio quality and bandwidth
Codec still widely in use	A required standard for many applications

[a]*kbits/s* Kilobits per second, *Hz* Hertz

ensure efficient route convergence as it influences the jitter and latency [5]. There are a number of IP routing protocols available such as IGRP, EIGRP, RIP, OSPF, and IS-IS, each with its own advantages and disadvantages. Depending on the algorithm used to find the best route between two nodes, most of them can be categorized into two classes: distance vector and link state. In this work, we have considered RIP and OSPF.

The Routing Information Protocol (RIP) is a distance-vector protocol based on Bellman and Ford algorithm [6], which sets a limitation on the number of hops allowed in a path from source to destination to prevent routing loops. It is easier to configure than OSPF or IS-IS, but generally yields a poor convergence. RIP allows routers to update their routing tables at periodic timeframes, most commonly at 30-s intervals. Typically, it broadcasts the entire routing table to all connected neighboring routers with all advertisement and generates high traffic on the network.

On the other hand, the Open Shortest Path First (OSPF) [6] is a link-state routing protocol which is based on the Dijkstra's algorithm. Typically, it computes the shortest path tree from source to destination, and then sends out the network updates to all neighbors. Upon receipt of these updates, each neighbor relays them to each of its own neighbors. Then, each router gathers this information in order to build a map of the entire network so as to enable communication and data transfer among all nodes.

1.3 Related Work

Call quality for VoIP technology is dependent on the design and implementation of the VoIP network that will be used. In [1], Syed and Ambore reported that the codec choice is the first factor in determining the quality of a VoIP call, and a codec with higher bit rate generally yields a better voice quality. They also argued that G.711 is the best available codec for voice encoding. In [5], Che and Cobley showed how VoIP performance can be affected by different routing protocols in the IPv4 technology. They simulated the RIP, OSPF, and EIGRP under three different scenarios using OPNET modeler. To study the route reconvergence, which is an important factor to VoIP user as it impacts both the latency and jitter, they configured a link to fail and autorecover. Their results show that OSPF is more consistent throughout the process, making it a better choice for VoIP solution compared to RIP and EIGRP. In [7], Singh et al. reported that signal quality in a VoIP solution depends on various factors such as networking conditions, coding processes, and speech content. In [8], Hussein and Jamwal studied the RIP, OSPF, and EIGRP protocols in a hybrid Ring and Star topology based on IPv4 technology using Riverbed Modeler Academic Edition 17.5. Their simulation results showed that EIGRP converges faster in large network (5-s) compared to OSPF (7.2-s) and RIP (14.15-s). To evaluate the performance of VoIP network, Ahmed et al. [9] used different metrics such as MOS, packet delay, end-to-end packet delay variation, IP traffic dropped, and Jitter. In [10], Besacier et al. focused on identifying the performance of audio and speech compression; they reported that MPEG transcoding impairs the speech recognition for low bitrates and sustains the performance of speech coders such as GSM and G.711. In [11], Sathu and Shah studied the performance of G.711.1, G.711.2, G.723.1, G.729.2, and G.729.3 codecs in VoIP, and reported G.711.1 as the preferred codecs to be used for both Windows and Linux operating systems. Their simulation results also revealed that Windows 7 generally yields a lesser delay when the G.729.2 and G.729.3 codecs are utilized.

1.4 Network Model

1.4.1 Network Topology

The topology considered in this chapter is a small enterprise network consisting of four CISCO 7200 routers (labeled as R1, R2, R3, and R4), two Nortel PS 6480 Ethernet switches (labeled as Switch 1 and Switch 2), and two IP phones (labeled as IP Phone1 and IP Phone2). All routers and switches are connected to each other via point-to-point (PPP) Digital Signal 3 (DS3) data links. The IP Phones are only supported via 10, 100, or 1000 BASE links; therefore, the highest available link 1000 BASE-X is used to connect them to switches. This topology is depicted in Fig. 1.1. It should be noted that the “Auto-Assign IP Addresses” option is used to assign the IPv4 address for all interfaces.

The considered topology [1] depicts a simple VoIP network exchanging voice packets to communicate between two IP phones, which are themselves connected via Ethernet switches and routers (as per Fig. 1.1).

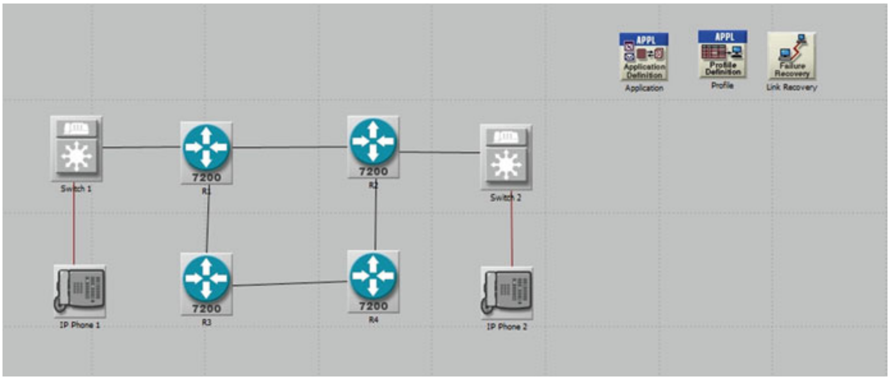

Fig. 1.1 Network topology

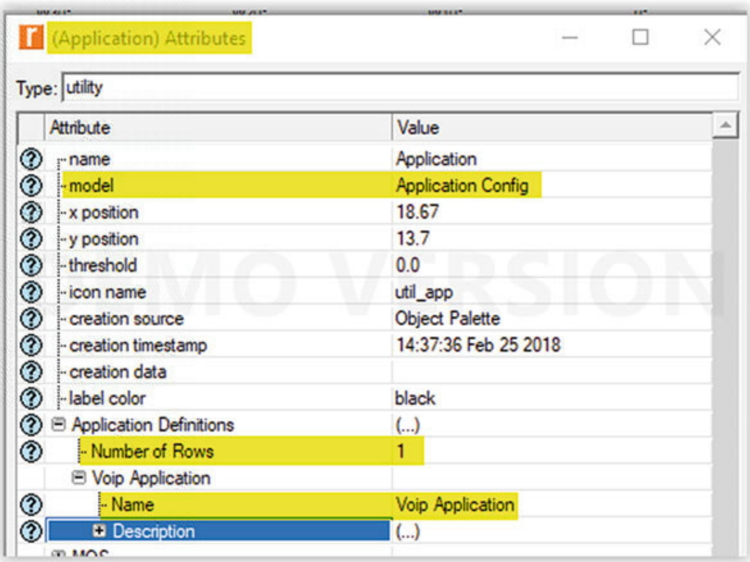

Fig. 1.2 Application configuration

1.4.2 Configuration Parameters

The application definition (Fig. 1.2), profile definition (Fig. 1.3), and failure recovery (Fig. 1.6) object are configured from the object palette to utilize the VoIP application.

Under the application attributes, the G.711 encoder scheme and PCM quality speech for voice are used for scenarios that are running G.711 (as shown in Fig. 1.4). On the other hand, the GSM encoder scheme and GSM quality speech for voice are

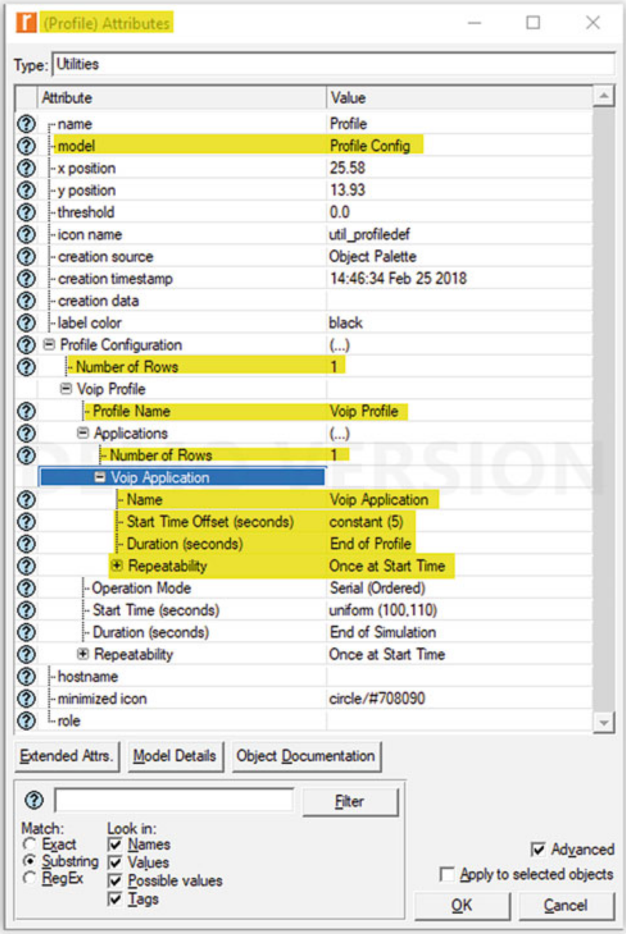

Fig. 1.3 Profile configuration

used for scenarios that are running GSM (as shown in Fig. 1.5). For the profile definition, the start time is set to 5-s offset for all scenarios (as shown in Fig. 1.6).

In order to simulate link failure and observe how the network recovers from it, the link between R1 and R2 is configured to process the failure recovery based on the information given in Table 1.2. The link fails and recovers after every 300-s. The simulation was running for 35 min. Figure 1.6 shows the Link Recovery configuration tab.

Before running our simulations, the DES statistics options and IP attributes were set for the appropriate protocol for each scenario. Depending on the considered scenario, the option “Sim Efficiency” was disabled for the RIP and OSPF so that the considered protocol can update its routing table when there is any change in the network such as link failure or recovery process.

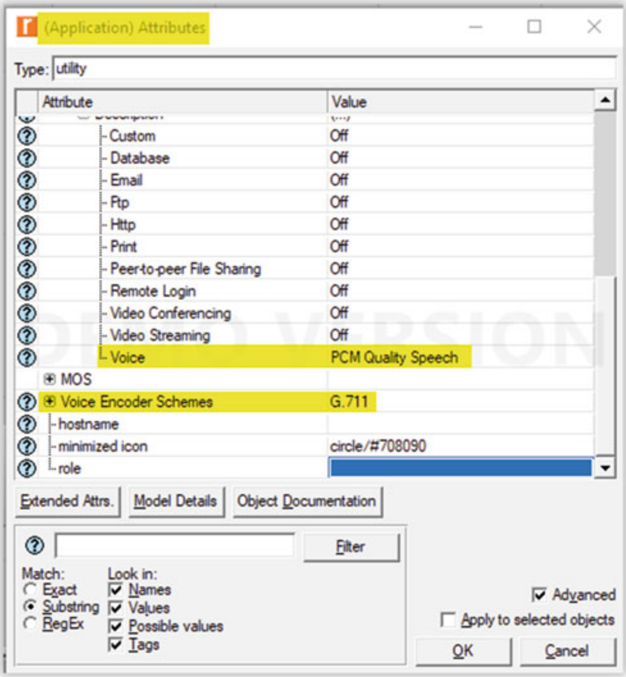

Fig. 1.4 G.711 setup

1.4.3 Routing Scenarios

For comparison purpose, four scenarios are configured with the network model shown in Fig. 1.1. Two of the scenarios are running the RIP protocol and the other two are running the OSPF protocol. One of the RIP (resp., OSPF) scenarios is running the G.711 codec and the other is running the GSM codec. Figure 1.7 depicts one of the RIP scenarios where all the network devices are configured to run the RIP, with the goal to compare the performance of the GSM and G.711 codecs. On the other hand, Fig. 1.8 depicts the same for one of the OSPF scenarios.

1.5 Simulation Results

In each graphic, the X-axis represents running time (minute) and the Y-axis represents the considered performance metrics being evaluated, which appear at the top (as a legend). For example, in Fig. 1.9, the Y-axis represents the *Voice Traffic Sent and Received* (in bytes/seconds).

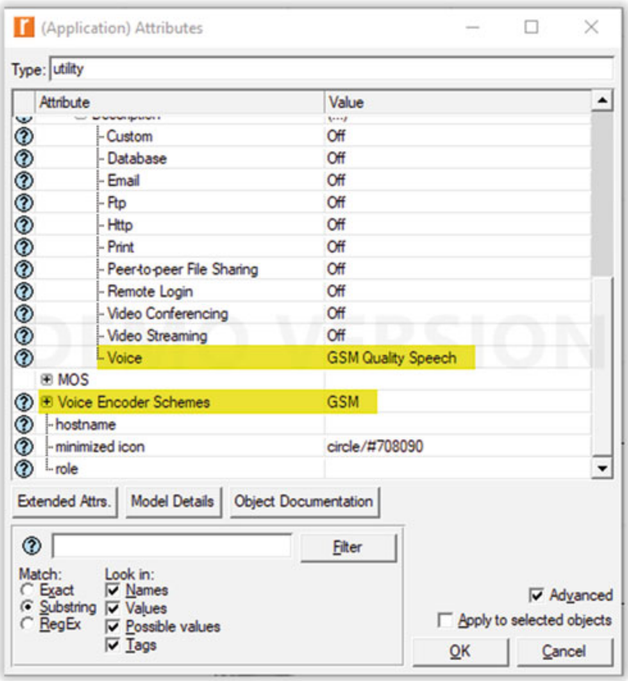

Fig. 1.5 GSM setup

1.5.1 Traffic Sent

This metric represents the total voice traffic sent from the source IP Phone to the destination IP Phone across the simulated network using the OSPF and RIP protocols. According to the results captured in Fig. 1.9, OSPF and RIP send about 6000 bytes/s using GSM and 30,000 bytes/s using G.711, which means that G.711 has sent 24,000 bytes more traffic per second than GSM at a given time.

1.5.2 Traffic Received

This metric represents the total voice traffic received from the source IP Phone to the destination IP Phone across the simulated network using the OSPF and RIP protocols. According to the results captured in Fig. 1.9, OSPF and RIP receive about 6000 bytes/s using GSM and about 30,000 bytes/s using G.711, meaning that G.711 receives 24,000 bytes more traffic per second than GSM at a given time, resulting in very low packet loss.

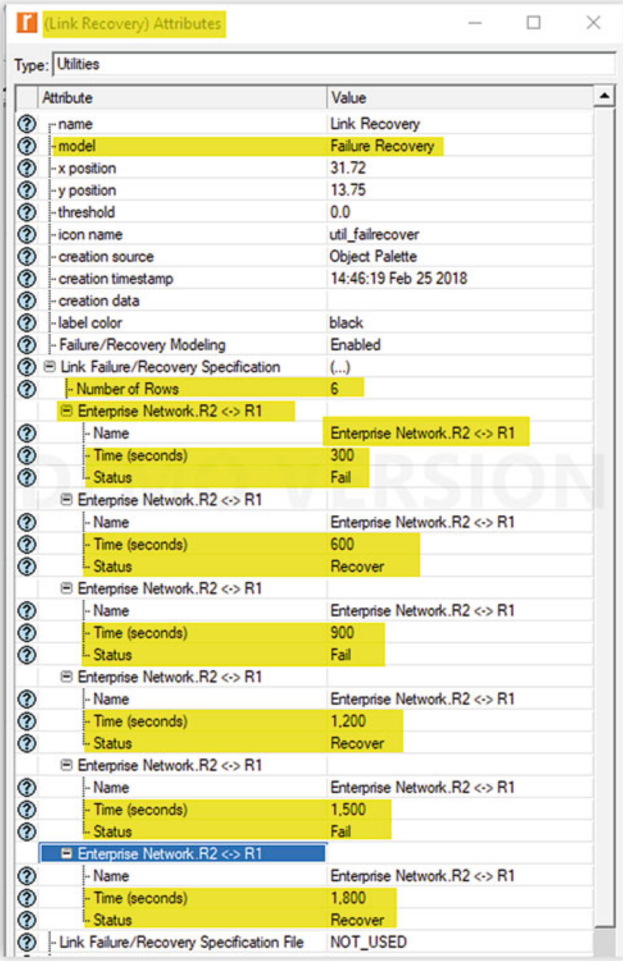

Fig. 1.6 Link recovery configuration

Table 1.2 Link recovery/failure

Time (s)	Status
300	Fail
600	Recover
900	Fail
1200	Recover
1500	Fail
1800	Recover

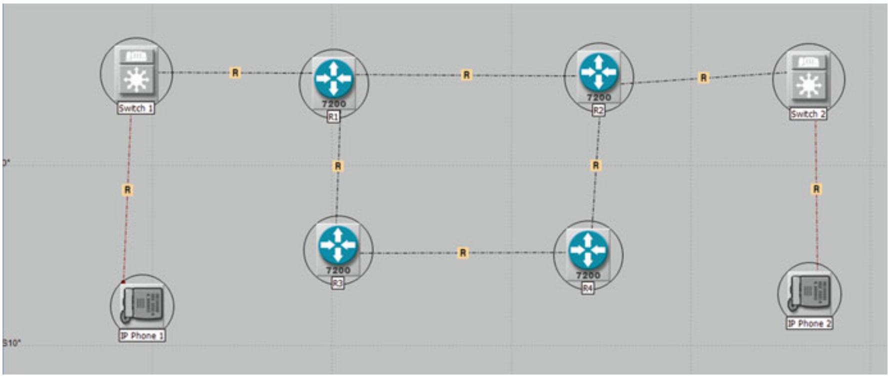

Fig. 1.7 RIP scenario

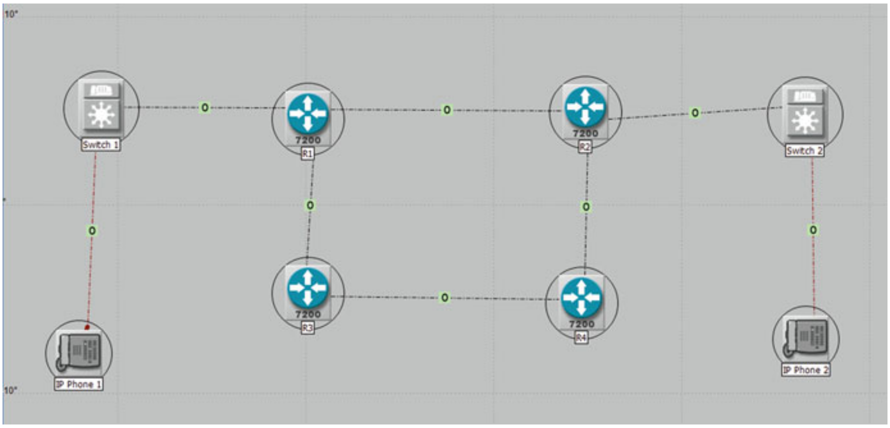

Fig. 1.8 OSPF scenario

1.5.3 Jitter

When packets are sent and received, there is an end-to-end delay variation between them. The variation in the arrival time of the packets at the receiver end leads to the jitter. The sender is expected to transmit each packet at a regular interval. Jitter can be caused by many factors such as congestion in the network, jitter buffer overload, timing drifting, or route changes. Ideally, a jitter of 0-ms is acceptable, but for higher-quality VoIP calls, jitter of 30-ms or less is also deemed acceptable [12].

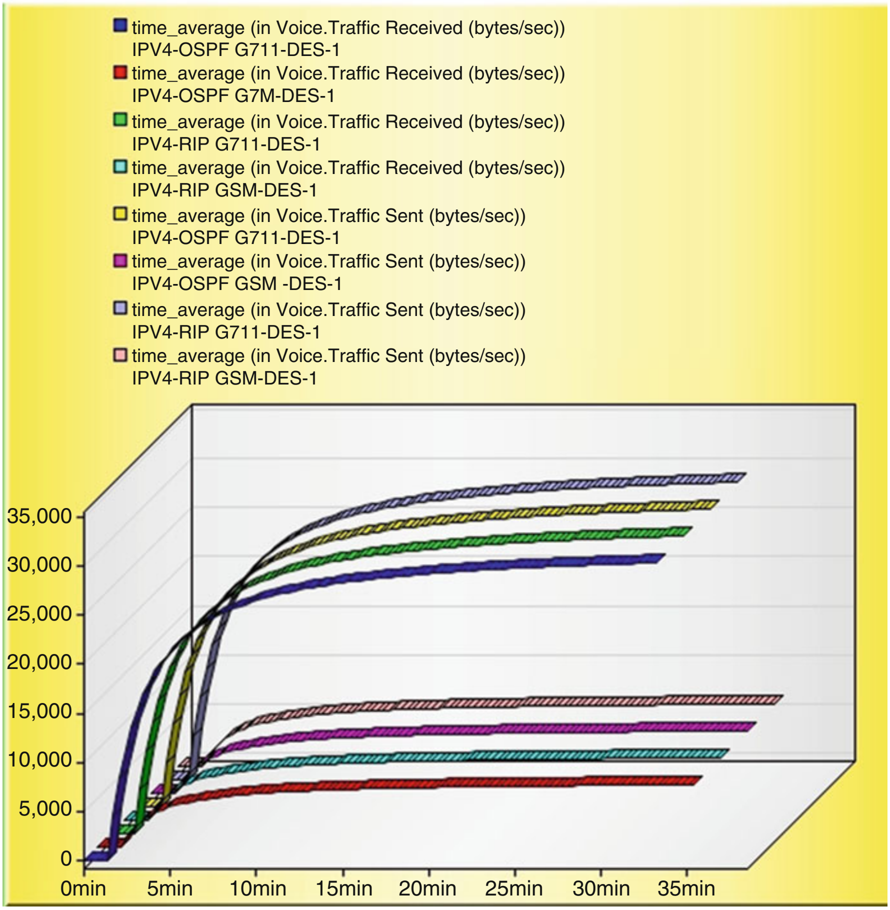

Fig. 1.9 Average voice traffic sent and received between GSM and G.711 using OSPF and RIP protocol

Based on the results captured in Fig. 1.10, OSPF and RIP have a similar performance for GSM and G.711 (jitter is less 0.00001-s), which means that a better quality of voice traffic is obtained since the lower the jitter, the better the quality of voice traffic.

1.5.4 End-to-End Delay

This metric represents the average time taken by voice packets to travel from source to destination. Delay can occur because of network congestion at the source or

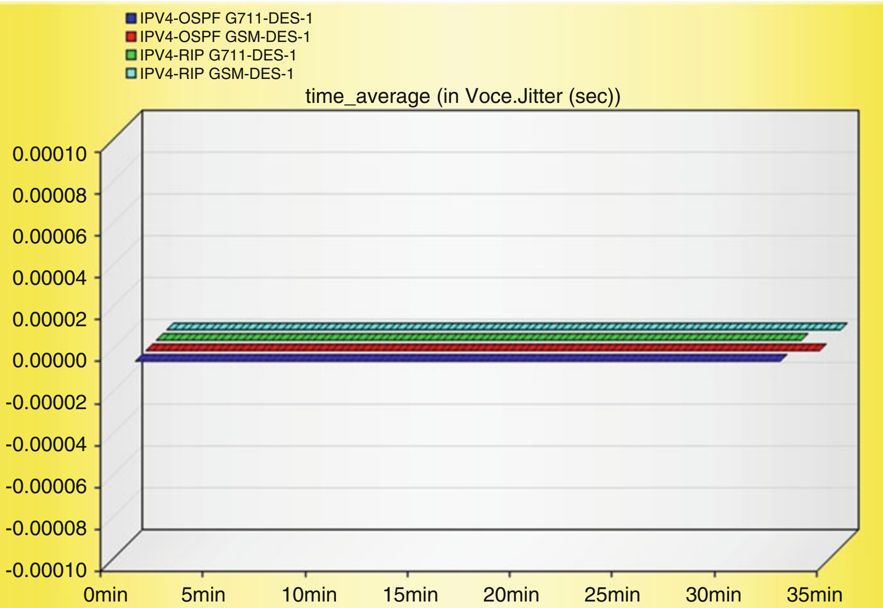

Fig. 1.10 Voice Jitter vs. time between GSM and G.711 using OSPF and RIP protocol

destination. For this metric, only the number of packets that have successfully been delivered to destination counts [13].

According to the results captured in Fig. 1.11, the end-to-end delay using GSM is 0.15-s and that using G.711 is 0.13-s for both OSPF and RIP, meaning that G.711 yields a lower end-to-end delay compared to GSM, resulting in a much faster voice encoding/decoding algorithm than that for GSM.

1.5.5 Packet Delay Variation

Packet delay variation (PDV) also known as IP Packet Delay variation is the difference in the end-to-end one-way delay between packets in transmission with any lost packets being ignored. The lower the packet delay variation, the better the performance is [13].

According to the results captured in Fig. 1.12, GSM has a PDV of about 0.000011-s whereas that of G.711 is about 0.000006-s for both OSPF and RIP,