POZNAN UNIVERSITY OF TECHNOLOGY

EUROPEAN CREDIT TRANSFER AND ACCUMULATION SYSTEM (ECTS)

COURSE DESCRIPTION CARD - SYLLABUS

Course name

IP Telephony [S2EiT1-SSiU-TelIP]

Course

Field of study

Electronics and Telecommunications

Year/Semester

2/3

Area of study (specialization)

Networks, Systems and Services

Profile of study

general academic

Level of study

second-cycle

Course offered in

polish

Form of study

full-time

Requirements

elective

Number of hours

Lecture

15

Laboratory classes

30

Other (e.g. online)

0

Tutorials

0

Projects/seminars

0

Number of credit points

4,00

Coordinators

prof. dr hab. inż. Grzegorz Danilewicz
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Lecturers

Prerequisites

The student knows the rules of information transfer in networks, knows the basics of analog to digital signal conversion and vice versa, knows signalling functions in telecommunications networks, knows the basics of network protocols from the TCP / IP protocol stack. Can obtain information from the literature, databases and other sources in Polish or English, interpret them, draw conclusions and justify opinions. Can communicate in Polish or English in a professional environment. The student can prepare a well-documented study of problems in the field of electronics and telecommunications in Polish or English. Can prepare an oral presentation in Polish or English on specific issues in the field of electronics and telecommunications. Student can educate herself/himself. Student can configure devices and run a local computer network. Can use applications analyzing traffic in LAN networks and applications that enable safe data transmission. Student knows the limitations of her/his own knowledge and skills, understands the need for further training.

Course objective

Presentation of the concept of using packet-switched networks (including IP-based) for the implementation of multimedia services, mainly audio and video. Indication of the similarities and differences in Internet telephony systems over previous solutions, such as mobile telephony, analog and ISDN. Presentation of the issues related to ensuring the quality of service (QoS) for real-time services implemented in packet switched networks.

Course-related learning outcomes

Knowledge:
1. Student has a knowledge of the equipment performing signaling functions and data transfer in packet switching networks used to provide multimedia services, knows the signaling systems used in networks based on IP protocol that provide establish, maintenance and disconnection of communication sessions to support real-time services.
2. Student has knowledge about the functioning of packet switching networks in practical applications for implementing multimedia services, knows the important parameters for assessing the quality of service in circuit switching and packet switching networks.
3. Student has the necessary knowledge to determine the functionality of the devices that need and / or can be used to create packet switching networks used to provide multimedia services, knows services and equipment to design a VoIP telephony network at least for a small business.

Skills:
1. Student is able to collect and analyze technical information needed for VoIP network design, is able to present these issues in the form of short paper and presentation (in Polish or English), and participate in the discussion.
2. Student can use the knowledge base accumulating norms and standards for telecommunications.
3. Student can practically implement the selected tasks for building a VoIP network.

Social competences:
1. Student understands the importance of communication for the development of individuals and societies, understands the evolutionary development of networks and telecommunications systems.

Methods for verifying learning outcomes and assessment criteria

Learning outcomes presented above are verified as follows:

Learning outcomes presented above are verified as follows:
In the laboratory, a grade is based on: the basis of preliminary questions, answers to questions about the material from the previous laboratory, the basis of written reports of laboratory and the tests.
The final grade is the result of component grades, with each component grade being positive. The rating scale from 2 (insufficient – negative) to 5 (very good) is used for component grades and for the final grade.
In respect of lectures, a grade is based on test and/or oral examination. The test is composed of 10-15 closed and open questions. A positive test grade is issued when the number of points exceeds 50%. The rating scale from 2 (insufficient – negative) to 5 (very good) is used for an exam grade. It is allowed to lower the threshold by a maximum of 10%.

Programme content

Laboratory: Installation of a virtual machine in the Windows environment and installation of the Linux operating system in a virtual machine, installation of the Asterisk environment, configuration of the Asterisk environment, configuration of VoIP phones and software on PCs, making voice calls using the Asterisk environment, configuration of IVR machines in the Asterisk environment, analysis SIP signaling.
Lectures: Introduction to the Internet telephony. Methods for switching signals (messages, circuits, channels, packets, datagrams, cells). The importance of signaling in telecommunication networks. Fundamentals of Voice over IP network solutions based on the H.323 protocol family. Functions of H.323 devices in the domain. Signaling protocols in the system based on the H.323 protocol family. Fundamentals of VoIP network solutions based on SIP. Device features in VoIP network based on SIP protocol. SIP signaling procedures. Cooperation of solutions based on H.323 and SIP. Related and new solutions in packet switching networks for the implementation of multimedia services. QoS parameters for VoIP.

Teaching methods

Laboratory experiments, lecture using a whiteboard and/or projector, seminar lecture with small discussion.

Bibliography

Basic
1. Marek Bromirski „Telefonia VoIP”, Wydawnictwo BTC, Warszawa 2006.
2. International Telecommunication Union (ITU-T), Packet-based multimedia communications systems, H.323 Recommendation.
2. J. Rosenberg et. al. SIP: Session Initiation Protocol, RFC 3261.
Additional
1. Samrat Ganguly, Sedeept Bhatnagar: VoIP. Wireless, P2P and New Enterprise Voice over IP, Wiley, 2008
2. Olivier Hersent, Jean-Pierre Petit, David Gurle: IP Telephony, Wiley, 2005
3. Olivier Hersent, Jean-Pierre Petit, David Gurle: Beyond VoIP Protocols, Wiley, 2005
4. Sivannarayana Nagireddi: VoIP Voice and Fax Signal Processing, Wiley, 2008

Breakdown of average student's workload

	Hours	ECTS
Total workload	100	4,00
Classes requiring direct contact with the teacher	58	2,00
Student's own work (literature studies, preparation for laboratory classes/ tutorials, preparation for tests/exam, project preparation)	42	2,00