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Course Outline for CNT 7301

VOIP: CISCO & ASTERISK IP PHONES

Effective: Spring 2015

I. CATALOG DESCRIPTION:
CNT 7301 — VOIP: CISCO & ASTERISK IP PHONES — 4.00 units

VoIP (Voice over Internet Protocol) offers a cost-effective alternative to plain old telephone service. This class covers VoIP planning and configuration basics and objectives for the Cisco and Astarisk certification tests. Extensive use is made of open source VoIP systems as a mechanism for practice implementation of VoIP in a network environment. This class is for all business, SOHO and computer users interested in using this technology, and will serve as a practical hands-on guide to the purchase and setup of hardware and software for Internet phones and the broadband Internet services required to support them, as well as covering objectives for certification exams.

3.00 Units Lecture 1.00 Units Lab

Strongly Recommended
CIS 50 - Intro to Computing Info Tech

Grading Methods:
Letter or P/NP

Discipline:

	MIN
Lecture Hours:	54.00
Lab Hours:	54.00
Total Hours:	108.00

II. NUMBER OF TIMES COURSE MAY BE TAKEN FOR CREDIT: 1

III. PREREQUISITE AND/OR ADVISORY SKILLS:

Before entering this course, it is strongly recommended that the student should be able to:

A. CIS50

IV. MEASURABLE OBJECTIVES:

Upon completion of this course, the student should be able to:

A. outline the key features, advantages and uses of VoIP B;
B. install and configure a VoIP service
C. describe and evaluate VoIP software and hardware solutions;
D. identify and discuss the advantages and limitations of closed source and open source VoIP technologies;
E. demonstrate an understanding of tradition Telephone;
F. demonstrate an understanding of IP Telephony;
G. outline the steps necessary to plan and design a VoIP installation for a home or SOHO business;
H. demonstrate an understanding of Windows, Linux and Apple OS X desktop environments as they relate to VoIP;
I. demonstrate the ability to configure system and network settings for VoIP;
J. discuss and evaluate security and authentication methods;
K. demonstrate an understanding of TCP-IP basics related to VoIP on LANs and WANs;
L. demonstrate the use of Wireshark and other network monitoring tools in evaluating VoIP;
M. describe and demonstrate troubleshooting methods for VoIP.

V. CONTENT:
A. Telephony Fundamentals:
    1. History
    2. Development
    3. Analog
    4. Digital
    5. PSTN
B. Voice Over IP
    1. IP
    2. VoIP Signaling
    3. VoIP Conversation

4. Features
C. Broadband phone services
   1. Infrastructure
   2. Costs
   3. Virtual / real phone numbers
   4. Features
D. Reliability / Availability
   1. QOS
   2. 911
   3. E911
   4. Backup power
   5. Features
   6. TCO
E. VoIP equipment
   1. Hardware
   2. Software
   3. ATAs
   4. SIP providers
F. VoIP PBX
   1. Cisco
   2. Astarisk
   3. Open Source
   4. Other vendors
G. Planning home VoIP
   1. Requirements
   2. Plan
   3. RFP
   4. RFQ
   5. Installation / Configuration
   6. Security / Troubleshooting
H. SOHO PBX
   1. Requirements
   2. Plan
   3. RFP
   4. RFQ
   5. Installation / Configuration
   6. Security / Troubleshooting
   7. Cisco call manager / options
   8. Astarisk
I. VoIP on Wireless
   1. Cordless sets
   2. LAN wireless
   3. Security
   4. Bandwidth
   5. Configuration / Troubleshooting
   6. Bridging
J. Additional services
   1. Chat
   2. client-server
   3. Skype
   4. Google Talk
   5. Gizmo Project
K. Future of Telephone
   1. Network convergence
   2. Video/VoIP/Data convergence
   3. Costs/TCO
   4. Features

VI. METHODS OF INSTRUCTION:
A. **Lecture** -
B. **Demonstration** -
C. **Research** -
D. **Lab** -
E. Assigned reading
F. **Discussion** -

VII. TYPICAL ASSIGNMENTS:

A. Reading / listening to presentations and readings
1. Presentations and lectures
a. Example: Lecture on Cisco Call Manager
2. Selected current online readings
a. Example: Read Astarisk configuration tutorial, at
www.astarisk.org

B. Access relevant material and read
1. Students use search engines to find readings relevant for each module
a. Example: Find resources describing effects of delay and jitter on VoIP, select 3 to read

C. Online flash based training
Example: Complete Skillsoft training module for network security.

D. Write reports
Example: “Discuss how SIP manages call setup and teardown.”

VIII. EVALUATION:

A. **Methods**

B. **Frequency**

1. Frequency:
    a. 6-10 module assignments
    b. Weekly discussion of group work
    c. 6-10 module quizzes
    d. 6-10 labs
    e. 1 final project
2. Typical quiz question:
    a. What is the difference between Gizmo Project and Skype?
    b. Describe the operation of SIP
3. Final exam

IX. TYPICAL TEXTS:

1. Flanagan , William *VoIP and Unified Communications: Internet Telephony and the Future Voice Network.* 1 ed., Wiley, 2012.
2. Association of Computing Machinery ACM.org student membership

X. OTHER MATERIALS REQUIRED OF STUDENTS:

A. Students require access to a computer connected to the Internet, with word processing and browser software, and an email address.
B. Association of Computing Machinery ACM.org student membership