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VAB: **Voice over IP**

Online course specification

Target audience:
Those responsible for the design and maintenance of private or public telephony systems.

Course aim:
This course describes the components, protocols and operation of telephony systems that operate over Internet Protocol (IP) networks with emphasis on the Session Initiation Protocol (SIP).

Course level: Advanced
An explanation of PTT course levels is given at the end of this document

Pre-requisites:
An understanding of the basic principles of telephony. It is recommended that the PTT online course "Introduction to telephony" is studied before attempting this "Voice over IP" course.
Prior knowledge of Ethernet and IP networks will also be beneficial. The PTT online courses "Ethernet networks" and "Internet protocols" cover these topics.

Course structure:
The course consists of the following modules:

1. Voice encoding and transport
2. VoIP systems
3. Session initiation protocol
4. VoIP configuration and security
5. VoIP service provision

Module 1: Voice encoding and transport

Module aim: This module describes the factors that affect the bandwidth requirement of a voice over IP (VoIP) system.

After completing this module, a trainee will be able to:

- describe the role of the various stages of processing of speech in a VoIP system including PCM encoding, echo cancellation, compression and silence suppression.
- describe the characteristics of typical encoder types in terms of output bit rate, sample size and mean opinion score (MOS).
- describe the benefits and characteristics of an "HD Voice" service.
- describe how the real-time transport protocol (RTP) minimises the effects of network impairments.
- describe the basic principles and benefit of packet loss concealment and silence suppression.
- describe the role of the various overheads in an IP packet carrying voice samples including those relating to the PPP, RTP, UDP and IP protocols.
- describe the relationship between the number of voice samples in each IP packet and bandwidth requirement and delay.

- compare the bandwidth requirements of various types of encoder used in VoIP systems.
- explain how the required Grade of Service affects the total bandwidth requirement of a VoIP system.

Module 2: VoIP systems

Module aim: This module describes the components, protocols, and operation of VoIP systems.

After completing this module, a trainee will be able to:

- explain that in a voice over IP system signalling and encoded media may take separate paths between calling and called telephone.
- explain that various signalling protocols exist and compare the main attributes and applications of the session initiation protocol (SIP) and H.323 protocols.
- describe the role of a call server in a VoIP system in terms of the provision of registration, authentication, address translation, routing, and admission control services.
- describe the role of a user agent, proxy server, register, redirect server, location server, and PSTN gateway in a VoIP system using SIP.
- describe the role of the various protocols and equipment involved in providing an H.323 audiovisual service.
- describe the role of the components of a private telephony system including IP-PBX, voice and PSTN gateways, and a SIP trunk.
- describe the role of, and the methods used for, admission control.

Module 3: Session Initiation Protocol

Module aim: This module describes the role and format of, and facilities provided by, the various signalling messages employed in systems using the Session Initiation Protocol (SIP).

After completing this module, a trainee will be able to:

- describe the role of a DNS server in a SIP system.
- describe and compare the role of SIP request, response, provisional response, and acknowledgement messages giving examples.
- describe the role of the Session Description Protocol (SDP).
- describe the role and format of SIP messages inc. INVITE and responses inc."200 OK".
- describe the role of the various SIP header fields and media attributes.
- explain how the route taken by SIP messages are defined with reference to Via, Contact and “Record route” message headers.
- describe the role and sequence of signalling messages during call set-up and release.
- describe the role, applications, and operation of a redirect server.
- describe the role and operation of various SIP facilities including authentication, call forwarding, call-on-hold and call forking.

Module 4: VoIP configuration and security

Module aim: This module explains the measures taken to overcome the barriers to VoIP service provision and to provide secure communications.

After completing this module, a trainee will be able to:

- explain the relevance of IP addressing and port numbers in the provision of a VoIP service.

- explain the barriers to VoIP provision for businesses with reference to network address translation (NAT) and firewalls.
- explain the use of STUN and application layer gateways to overcome the barrier to VoIP presented by NAT.
- explain the use of port forwarding to overcome the barrier to VoIP presented by firewalls.
- describe and compare the operation and benefits of symmetric and asymmetric encryption of encoded voice.
- explain how digital certificates provide trusted authentication of devices using VoIP services with reference to digital signatures and certification authorities.
- describe the role of, and relationship between, the Transport Layer Security (TLS) protocol and the Secure Real-time Transport Protocol (SRTP).

Module 5: VoIP service provision

Module aim: This module describes the systems that provide Voice over IP services for the general public and businesses with emphasis on ensuring an acceptable quality of service.

After completing this module, a trainee will be able to:

- describe how networks employing the Internet protocol may introduce delays which impair voice over services.
- explain that certain types of IP network can provide quality of service guarantees by giving priority to voice over IP packets.
- Give examples of the types of protocol that can provide a differentiated class of service and their use on local area networks, leased lines, and wide area networks.
- describe how call control and quality of service enforcement is delivered in a network that provides fixed line, public telephony services using voice over IP techniques.
- describe how call control and quality of service enforcement for a voice over IP service is provided by 4G (LTE) and 5G mobile networks.
- describe the provision and benefits of a hosted ("cloud-based") VoIP service.
- explain the use of a SIP trunk in providing access to an Internet telephony service provider's system.
- compare methods of providing a SIP trunk with reference to the characteristics that will influence the choice of SIP trunk for a given application.
- describe the role of, and facilities provided by, a session border controller (SBC).
- explain how an SBC overcomes problems associated with network address translation for voice calls and improves the security of the hosting private network.
- explain how cloud-based telephony services can be integrated with existing private SIP or H.323 systems and provide alternative routes for external calls.

Course access requirements:
To access the course, a computer running a browser such as Google Chrome, Safari etc is required. The computer should have Internet access. A screen resolution of at least 1024x768 is necessary.

Learning facilities:
This online course employs interactive simulations, hypertext links to an online glossary and multiple-choice question sessions to fully involve the trainee in the learning experience. Each module provides revision links to previously studied, relevant topics. A record of progress and level of achievement is recorded for each trainee. Once studied as a structured, assessed course, the content can be browsed for revision or reference.

PTT course levels:
PTT online courses are categorised by one of three levels according to the depth of treatment they provide:

1. **Introductory:**

   PTT Introductory courses are designed for those with no previous experience or knowledge of telecommunications. These courses provide an overview of telecommunications or discuss the fundamentals of electronic communications. The study of general science at secondary (high) school is a typical pre-requisite for PTT Introductory courses.
   PTT Introductory courses are suitable for those joining the telecommunications sector particularly those in an apprenticeship programme.

2. **Intermediate:**

   PTT Intermediate courses are designed for technicians and engineers requiring an understanding of a certain aspect of telecommunications. Those planning to study an Intermediate course should have an understanding of the basic principles of electronic communications.
   The depth of treatment provided by Intermediate courses is typically equivalent to level 3 of a UK national vocational qualification (NVQ). PTT Intermediate courses can be used to support the attainment of a Communications Technology NVQ at level 3.

3. **Advanced:**

   PTT Advanced courses are designed for those who require an in-depth treatment of a certain aspect of telecommunications. Such courses are suitable for system designers as well as those who will be responsible for the maintenance of the system described in the course.
   Those planning to study a PTT Advanced course should have a background in telecommunications, and an understanding of telecommunications fundamentals and the principles of the type of telecommunications system described in the course.

PTT
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