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MAB: **Voice over IP**

Online course specification

Target audience:
Those responsible for the design or maintenance of Voice over IP systems.

Course aim:
This course introduces the principles and operation of telephony services that operate over Internet Protocol (IP) networks - Voice over IP (VoIP).

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

Pre-requisites:
An understanding of the operation and use of the Internet protocols. It is recommended that the PTT course "IP networks" is studied before attempting this course. An appreciation of the techniques used to digitally encode speech, and the differences between a circuit switched and a packet switched telephone service would also be useful. It is recommended that the PTT course "Media encoding and transmission" is also studied before this course.

Course structure:
The course consists of the following 7 modules:
1. Course introduction
2. VoIP packet transport
3. Voice encoding
4. VoIP systems
5. Session Initiation Protocol
6. H.323 VoIP systems
7. Applications of VoIP

Module 1: Course introduction

Module aim: To summarise the aims of each module and introduce the navigation and learning facilities provided by the course.

Module 2: VoIP packet transport

Module aim: Describe the techniques used to transport voice samples over an IP network.

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

- describe the advantages of a Voice over IP (VoIP) telephony service over that provided over a circuit-switched network.
- explain the basic principles of carrying voice samples in IP packets.
- describe the effects of the network impairments delay, jitter and packet loss on a telephone call.
- identify the main causes of delay in a packet switched network.

- describe how the real-time transport protocol (RTP) minimises the effects of network impairments.
- describe the basic principles and benefit of packet loss concealment.
- explain that the amount of delay introduced by a VoIP system depends on the method of voice compression and the size of jitter reduction buffers.
- indicate the target quality of service for an IP network carrying VoIP traffic in terms of parameters such as maximum delay, jitter and packet loss.

Module 3: Voice encoding

Module aim: Describe the factors that affect the bandwidth requirement of a 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 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 Class of Service affects the total bandwidth requirement of a Voice over IP system.

Module 4: VoIP systems

Module aim: Describe the components, protocols and operation of VoIP systems.

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

- describe the role of the processes involved in making a telephone call inc. routing, signalling, and call control.
- explain the role of the various standards bodies in developing protocols for VoIP systems.
- compare the main features of VoIP systems using SIP and those conforming to H.323.
- 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 gateways in a VoIP system.
- describe the requirements for a VoIP system in terms of service parameters including grade of service, availability and post-dial delay.
- describe the role and the relationship between the DiffServ, RSVP and MPLS protocols in providing an improved quality of service for VoIP calls.
- describe the role of, and the methods used for, admission control.

Module 5: Session Initiation Protocol

Module aim: Describe the components, protocols and operation of signalling systems using the Session Initiation Protocol (SIP).

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

- describe the role of the functional components of a VoIP system using SIP including user agent, proxy server, location and redirect servers.
- describe the role of a DNS server in a SIP system.
- describe the role of the Session Description Protocol (SDP).
- describe the format of SIP messages inc. INVITE and responses inc."200 OK".
- describe the role of SIP header fields inc. "To:" and "Via" and media attributes.
- describe the role and sequence of signalling messages during call set-up and release.
- describe the role and operation of various SIP facilities including authentication, call forwarding, call-on-hold and call forking.

Module 6: H.323 VoIP systems

Module aim: Describe the components, protocols and operation of signalling systems conforming to the ITU H.323 family of recommendations.

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

- describe the role of the various protocols involved in providing an H.323 audiovisual service inc. H.225, H.245, RTP, IP, TCP and UDP.
- explain that various types of encoder can be used in an H.323 system giving examples of their bit rate requirements.
- describe the role of the components of an H.323 system inc. gateways and gatekeepers and multipoint control unit (MCU).
- describe the mandatory and optional functions of a gatekeeper inc. registration, address translation and call authorisation.
- describe and compare the direct and gatekeeper routed signalling modes.
- describe the role and sequence of signalling messages involved in call set-up in terms of admission control, call signalling, capability negotiation and media channel initiation.
- describe methods of reducing post-dial delay inc. “Early H.245”, “fast connect” and H.245 tunneling.
- describe the role and sequence of signalling messages occurring during call forwarding in an H.323 system.

Module 7: Applications of VoIP

Module aim: To introduce examples of the systems that provide Voice over IP and video conferencing services for the general public and businesses.

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

- 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 a 4G (LTE) mobile network.
- describe the provision of a hosted (“cloud-based”) VoIP service.

- describe the provision of a VoIP facility using an on-site IP-based Private Branch eXchange (PBX) and SIP trunking.
- explain the barriers to VoIP provision for businesses with reference to NATS and firewalls, and give examples of solutions to those barriers.
- describe the components and basic operation of an H.323 based video conferencing system.

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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