Understanding Voice over IP

Introduction

For years, many different data networking protocols have existed, but now, data communications has firmly found its home in the form of IP, the Internet Protocol. On the other hand, even since telecommunications began, voice communications has rigidly stuck to a classic circuit switched networking environment.

Over the years, many technologies have been used for the integration of voice and data with some offering much better solutions than others, at least from a technical perspective. IP is far from the best technical solution for the support of voice, however there are many reasons why it is rapidly becoming the platform of choice for, not only data, but voice too.

This course looks at the subject of voice over IP, covering various methods of its implementation and looking at many of the relevant standards. We also cover many of the issues that need to be considered when deploying voice over IP and discuss how these may be dealt with.

In order to present the information in such a way that it is best understood, we make significant use of live demonstration equipment to provide a real-life and tangible perspective to the course. This course covers live demos of IP phones, IP PBX systems, gateways, voice compression systems and covers both the H.323 and SIP protocols. We also make use of protocol analysis equipment to show what is actually happening in the IP network.

Who should attend the course?

This course is aimed at voice and/or data professionals who wish to extend their knowledge and experience into the area of voice over IP. A detailed knowledge of neither voice nor data (IP) is required for a student to derive benefit from the course. A general understanding of voice and/or data (IP) will however be of help.

Course length

Two days.

Course Agenda

The following is an outline of the sections included in the course:

1. Introduction to voice over IP (VoIP)
2. Overview of the voice technologies relevant to voice over IP
3. Overview of the IP technologies relevant to voice over IP
4. The H.323 framework for voice over IP
5. Session Initiation Protocol - SIP for voice over IP
6. MGCP and Megaco / H.248 for voice over IP
7. Voice over IP applications
8. QoS requirements and solutions with voice over IP
9. Other considerations with voice over IP

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1 Introduction to voice over IP (VoIP)

This section gives a basic introduction to voice over IP, looking at some fundamental applications and business reasons as to why one might consider deploying it.

- Why VoIP? A view from a business perspective as to why voice over IP may be an appropriate technology to deploy in many voice related applications
- A look at voice over IP as deployed across the Internet or across a private IP network.
- An introduction to some of the standards for voice over IP:
    - H.323 and related protocols
    - Session Initiation Protocol (SIP)
    - Media Gateway Control Protocol (MGCP) and Megaco / H.248
- An introduction to some of the devices that may be used to implement a voice over IP network
    - IP phones
        - Power to the IP phone - powered Ethernet
    - Gateways
    - Call control systems (e.g. the Gatekeeper)
    - IP enabled voice switches (PBX and public network switches)

An introduction to some of the challenges of deploying voice over IP

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2 Overview of the voice technologies relevant to voice over IP

This section gives a simple overview of the voice related technologies that we see as relevant to voice over IP. It is not our intention to provide an in-depth voice technology course coverage as this would take too long. Having said that, the course tutor is very experienced in voice technology training and, if appropriate, can spend extra time with students if required to help them understand the important aspects (This typically would be done during a break so as not to impact others on the course).

- Interface types:
    - Foreign Exchange Station (FXS) and Foreign Exchange Office (FXO) interfaces
    - Ear & Mouth - E&M 2 wire and 4 wire
    - 2.048Mbit/s digital interface - G.732/G.704/G.703
    - Channel Associated Signalling (CAS).
    - Common Channel Signalling (CCS) with a summary of some of the key types including Q.931, QSIG, DPNSS, SS7
- Voice Coding:
    - G.711 Pulse Code Modulation (PCM) - 64kbit/s
    - Compressed voice:

- G.728 Low-Delay Code Excited Linear Prediction (CELP) - LD-CELP
- G.729 Conjugate Structured Algebraic CELP (CS-ACELP) - Also G.729a and b
- G.723.1 MP-MLQ and ACELP
- The use of Mean Opinion Score (MOS) to quantify the quality of voice communications

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3 Overview of the IP technologies relevant to voice over IP

This section gives a simple overview of the IP related protocols that are relevant to voice over IP. It is not our intention to provide an in-depth IP course coverage as this would take too long. Having said that, the course tutor is very experienced in IP technology training and, if appropriate, can spend extra time with students if required to help them understand the important aspects (This typically would be done during a break so as not to impact others on the course).

- The IP protocol and the key parts of the IP header relevant to voice over IP. We discuss this in relation to IP version 4 and IP version 6
- Overview of IP addressing and network and subnet addressing
- Other key protocols:
    - Transmission Control Protocol (TCP)
    - User DataGram Protocol (UDP)
    - Real-Time Protocol (RTP) and Real-Time Control Protocol (RTCP)

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4 The H.323 framework for voice over IP

H.323 is the oldest and most commonly used protocol suite used to deliver voice over IP. The reasons for this are manifold including the fact that the protocol specifications have been in place for a long time and others such as Microsoft's inclusion of H.323 in its Netmeeting software, an application residing on virtually every desktop computer in the world. The original version (V1) of H.323 released in 1996 was referred to as "Visual telephone systems and equipment for local area networks which provide a non-guaranteed quality of service". Its scope goes far beyond just voice over IP and in recognition of its expanded scope to cover not just LANs, it was renamed in version 2 as "Packet based multimedia communications systems". Since then, it has evolved further, adding various new features and improvements plus correcting issues that were present in its earlier versions.

- Overview of H.323
- H.323 architecture:
    - The H.323 Terminal
    - The H.323 Gateway
    - The H.323 Gatekeeper
    - The H.323 Multipoint Control Unit (MCU)

- H.225.0 - This deals with a number of aspects of establishing communication across the data network including Registration, Admissions & Status (RAS) operations, along with Q.931 signalling messages for call setup, call control, and communications between terminals, gateways, gatekeepers, and MCUs.
- H.245 - This specifies the syntax and semantics of terminal information messages as well as procedures to use them for in-band negotiation at the start of or during communication.
- H.323 examples

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5 Session Initiation Protocol - SIP for voice over IP

The Session Initiation Protocol (SIP) is a signalling protocol used for establishing sessions in an IP network. A session could be a simple two-way telephone call or it could be a collaborative multi-media conference session. It is basically a very simple protocol having been developed purely as a mechanism to establish sessions. It does not know about the details of a session, it just initiates, terminates and modifies sessions. In order to handle other aspects of end-to-end communications, SIP was designed to reuse many existing protocols and protocol design concepts.

- SIP architecture:
    - SIP user agents (user devices) - phones, workstations, PDAs etc.
    - SIP servers - Proxy server, redirect server, registrar server
    - SIP location servers
    - SIP gateways
- SIP functions:
    - Address resolution
    - Session setup
    - Media negotiation using the Session Description Protocol (SDP)
- SIP examples - We shall look at a number of examples of SIP's use including basic call setup, call forward plus others.

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6 MGCP and Megaco / H.248 for voice over IP

Media Gateway Control Protocol (MGCP) and Megaco / H.248 are control protocols designed to be used between a Media Gateway Controller or call agent and a media gateway. Megaco / H.248 is newer than MGCP, and is seen by many as MGCP's replacement. That said, many products already employ MGCP and therefore, for some time, both protocols will be used.

- Media Gateway Controllers
- Media Gateways
- Endpoints
- MGCP & Megaco / H.248 message examples

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7 Voice over IP applications

Voice over IP may be applied in so many ways that it makes it very difficult to cover all possible applications. This section aims at showing how voice over IP is, and will be in the future, used in various real-life applications. It also shows some of the additional benefits that integrating voice into an IP environment may offer over and above the classic way of supporting voice with a standard voice switch.

- PBX replacement applications - Full-blown IP telephony using IP phones, call control platforms, gateways etc.
- Toll bypass - Using voice over IP gateways to allow long-distance voice communications across existing data networks including private data networks and the Internet, thus bypassing the call charges normally associated with routing calls via the public telephone network.
- IP-enabled PBXs - Typically seen as an interim option between a traditional PBX based system and a full-blown IP solution. This option typically takes a PBX and adds IP capabilities so that PBX users can communicate with other IP telephony users.

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8 QoS requirements and solutions with voice over IP

Often banded about as the reason why voice over IP will not work successfully, quality of service is key to the operation of a real quality-grade voice over IP network. Many technologies and protocols exist to solve the quality of service issue, and here we intend to explain the key ones and de-mystify any uncertainties about how voice over IP may be deployed and operate very successfully.

- Quality of Service (QoS) issues:
    - Basic voice quality in terms of distortion, frequency reponse etc.
    - The causes of delay in a voice over IP system
    - A description and explanation of what levels of delay will actually impair a voice conversation and what levels will not be noticed
    - The effect of delay resulting in echo
    - The effect of transporting non-voice signals such as fax, modems, DTMF tones etc.
- Description of some protocols used to help deliver QoS guarantees:
    - IEEE 802.1p / IEEE 802.1Q
    - Asynchronous Transfer Mode (ATM)
    - Resource Reservation Protocol (RSVP)
    - Differentiated Services (Diffserv)
- Multi-Protocol Label Switching support of QoS

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9 Other considerations with voice over IP

We finish up with a look at a few potential issues with using voice over IP.

- Network Address Translation (NAT) issues
- Issues with voice over IP across Firewalls

End of Training Outline

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