Understanding Voice Technologies & Networking

Introduction

The integration of voice with data is not new. For many years, voice has been integrated with data in networks based on multiplexer technology. However, today, new technologies are being developed to make even more effective use of the available bandwidth as well as matching today's new networking applications. Terms such as VTOA (Voice and Telephony over ATM), VoFR (Voice over Frame Relay) and VoIP (Voice over IP) are rapidly becoming more common. Most people working in the datacommunications and telecommunications industries today will most likely not have escaped the hype particularly around Voice over IP.

The use of these technologies for the support of voice traffic is, unfortunately, not as simple as many would like to think. It creates many challenges for those responsible for network engineering. For example, delay, echo, issues with voice compression and incorrect voice signal levels can all easily result in poor communication even to the extent of complete communication failure. This course addresses all of these issues and many more, providing an insight into what issues exist and how they may be resolved.

Who should attend the course?

This course was originally aimed at individuals from a data background who need to develop a greater understanding of voice technologies, from the basics through to the practical "in the field" issues. It has however also been found to be suitable for those who already have experience in voice networking, yet wish to consolidate their knowledge and understand many of the new issues facing voice in today's communication networks.

Course Agenda

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

1. Introduction to Voice Communications
2. Digital Voice
3. The Telephone
4. The Telephone Exchange
5. Digital Voice Transmission
6. E1 Digital Voice Interface
7. DS-1 (T1) Digital Voice Interface
8. Signalling on Analogue Interfaces
9. Signalling on Digital Interfaces
10. Overview of Basic Rate ISDN
11. Voice/Speech Compression
12. Speech Impairments
13. Echo and Echo Control
14. Voice and Data Integration
15. Voice over Frame Relay (VoFR)
16. Voice and Telephony over ATM (VTOA)
17. Voice over IP (VoIP)

Course Length

Three days.

Course Section Descriptions

1 Introduction to Voice

The objective of this section is to give a brief introduction to the true fundamentals of voice in terms of how speech is generated and the importance of certain aspects of speech when applied to networking communications.

- How speech is generated and the sounds that make up speech
- The importance of certain frequencies and levels
- Review of dBs

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2 Digital Voice

Today, voice is usually integrated into networks in a digital format. However, to be compatible with the human user, the signals must still be in an analogue form at some point, notably the telephone set. This section looks at how voice is converted from an analogue signal into a digital format and looks at some of the issues involved in its conversion.

- Comparison of analogue and digital voice
- PCM encoding: Analogue - Sample - PAM - Quantise - Compand - PCM (G.711)
- A law and mu law companding - Where should A law and mu law be used?
- Quantisation distortion
- Power of the digital signal

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3 The Telephone

The telephone is the fundamental building block of a voice network. While it was originally designed in 1874 and patented in 1876, its principles have not changed greatly since then. In fact, many of the techniques used in these early telephones are still used extensively today. The telephone also has attributes that make it one of the greatest causes of problems in networks today, that of echo. This section goes into the operation of the basic 2 wire telephone set, and to give a complete picture, also looks at other proprietary types of telephone, and finally looking at the operation of the Basic Rate ISDN set.

- Basic operation of telephone - 2 to 4 wire hybrid, transmit/receive levels etc
- Loop disconnect and DTMF signalling - Advantages and disadvantages
- Digital telephone set

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4 The Telephone Exchange

This section gives an overview of a modern digital telephone exchange and looks at how it switches telephone calls, how it routes calls and how it interfaces to other equipment such as multiplexers.

- Call routing in a digital PBX
- Principles of switching
- Interfaces available – analogue and digital

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5 Digital Voice Transmission

Today, practically all voice is carried across networks in a digital form. This section serves as an introduction to the next two, looking at the basis of the support of multiple PCM voice channels across standardised primary rate digital interfaces.

- The Channel Bank
- An introduction to the E1 and DS-1/T1 digital voice interfaces

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6 E1 Digital Voice Interface

The predominant type of interface used for voice transport within a Wide Area Network today is the digital interface. There are two main types of digital interface available on a PBX, those of DS-1 (T1) and E1. DS-1 is used in North America and in some cases in Japan, while E1 is the standard used elsewhere. This section looks at the E1 interface in depth.

- Electrical characteristics - G.703 - HDB3
- Framing - G.704
- Channel Associated Signalling - Timeslot 16
- Common Channel Signalling
- E1 Alarms

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7 DS-1 Digital Voice Interface

The operation of a DS-1 interface is somewhat different to an E1, and a detailed understanding of the differences is key for those involved with networks using both. This sections looks at the same areas as with E1s, however, as we shall see, the techniques used are quite different in many respects.

- Electrical characteristics AMI - B8ZS
- Framing - D4 and Extended Superframe
- Channel Associated Signalling - Robbed bit signalling
- Common Channel Signalling
- DS-1 Alarms

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8 Signalling on Analogue Interfaces

Whilst most interfaces between telephone exchanges and other voice networking equipment are digital today, there is still a significant demand for analogue interfaces. Typically, analogue interfaces are used where only a small number of voice channels are required whereas digital interfaces are used where a larger number are required. This section discusses a number of different analogue interface types, focussing on a particularly common type known as the E & M (Ear and Mouth) interface.

- Loop Disconnect
- E & M – Various types (Main section)

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9 Signalling on Digital Interfaces

Digital signalling systems are used on digital interfaces to carry such information as call setup details in addition to more advanced messages for activating PBX features across a network. It is important to understand the differences between the two main types of digital signalling i.e. Channel Associated Signalling (CAS) and Common Channel Signalling (CCS), since they are quite different. CAS tends to be easier to deal with, while CCS is more common and, typically, provides more feature capability. This section looks at some examples of CAS yet concentrates on CCS with prime focus on signalling protocols including Q.931, QSIG and Euro-ISDN.

- Channel Associated Signalling
- Common Channel Signalling
- "Standard" and proprietary CCS protocols - ISDN/Q.931, Euro ISDN, QSIG, DPNSS, CCITT#7 (CCS7, SS7)

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10 Overview of Basic Rate ISDN

While much of this course is orientated towards primary rate interfaces, this section gives an introduction to Basic Rate ISDN and looks at its operation and some of its unique features and capabilities.

- Physical Structure - Multiple Devices
- Basic Rate Frame Structure
- Bearer Services, Teleservices and Supplementary Services
- Circuit Mode, Packet Mode and Frame Mode Services

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11 Voice/Speech Compression

Voice compression is one of the prime reasons for integrating a PBX network with a Wide Area Network. Voice compression has been available on TDMs for many years. Over time, more and more compression techniques have been developed to improve the efficiency of the bandwidth used. Key to the use of voice compression are the effects it has upon the quality of the voice being carried both in terms of distortion and, in some cases, the additional delay added. Furthermore, when we try to support voice band data such as modem traffic, fax traffic or possibly in-band signalling, a number of issues arise. An appreciation of these factors is key to the successful implementation of voice compression.

- What is voice compression and why do it?
- A look at various different types of speech coder – Waveform, Source and Hybrid coders
- ADPCM - G.726 (16, 24, 32 & 40kbit/s)
- Embedded ADPCM (E-ADPCM) – G.727 (16, 24, 32 & 40kbit/s)
- LD-CELP - G.728 (16kbit/s)
- CS-ACELP - G.729 (8kbit/s)
- ACELP/MP-MLQ – G.723.1 (5.3k and 6.3kbit/s)
- 7kHz ADPCM - G.722 (64kbit/s)
- Other compression systems
- Support of voice band data on voice compression circuits

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12 Speech Impairments

Essential to the successful support of voice in any network whether it be a traditional TDM (Time Division Multiplex) environment or a packet based network such as voice over IP, is the minimization of speech impairments. This section looks at some of the key potential impairments including delay and distortion.

- End-to-end delay including packetisation delay, transmitter delay and packet jitter delay
- What delays can we tolerate? - ITU-T G.114
- Distortion - What can we tolerate? - ITU-T G.113

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13 Echo and Echo Control

Today, the use of such technologies as ATM and Frame Relay, as well as techniques such as voice compression, are resulting in more and more delay being imposed onto voice connections across our networks. The combination of signal reflections and delay create the effect of echo. In practice, we can tolerate a certain amount of echo, although only very little, and steps should be taken to eliminate it. The objective of this section is to look at various causes of echo and to discuss how it can be removed. Both echo suppressors and echo cancellers are discussed, although the primary focus is on echo cancellers.

- Delay and what we can tolerate
- Causes of echo
- Echo Suppressors
- Echo Cancellers
- G.165 vs. G.168 echo cancellers

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14 Voice and Data Integration

This section investigates a number of potential ways that voice and data may be integrated and some of the key aspects that need to be considered in each case.

- The transport of voice across a data network when supporting analogue interfaces and digital interfaces with CAS and CCS signalling types
- The interworking of DS-1 and E1
- Using the "data network" to support voice switching

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15 Voice over Frame Relay (VoFR)

This section looks at how voice is supported on Frame Relay as defined by the Frame Relay Forum FRF.11 Implementation Agreement.

- Introduction to Frame Relay
- How voice samples are carried in frames
- The use of Virtual Circuits for voice, data and voice/data
- Multiplexing of multiple channels within a frame
- The support of signalling: DTMF, CAS and CCS
- The support of fax traffic
- FRF.12 – Frame Relay Fragmentation

HN Networks • Endellion • Money Row Green • Holyport • Berkshire • SL6 2NA
Tel: 01628 622187 • Fax: 01628 672813 • WWW: http://www.hn-networks.co.uk

16 Voice and Telephony over ATM (VTOA)

In this section we look at methods used to carry voice over ATM. The ATM Forum has produced a number of standards for VTOA which we shall look at. In addition we will also look at the specification for a new AAL, AAL2 for the support of voice trunking including compressed speech support. We shall look at the issues that surround some of these techniques along with many of the benefits that can be derived from them.

- Introduction to ATM
- Transport of a complete E1/DS-1 digital voice stream - Circuit Emulation
- Support of a single voice channel in a single Virtual Circuit
- Support of multiple voice channels in a single Virtual Circuit
- Silence Suppression (Speech Activity Detection, Voice Activity Detection)
- af-vtoa-0078.000: Circuit Emulation Service Interoperability Specification V2
- af-vtoa-0083.000: Voice and Telephony Over ATM to the Desktop Specification
- af-vtoa-0085.000: Specifications of (DBCES) Dynamic Bandwidth Utilisation - In 64kbit/s Timeslot Trunking Over ATM - Using CES
- af-vtoa-0089.000: Voice and Telephony Over ATM - ATM Trunking using AAL1 for Narrowband Services Version 1
- ITU-T I.363.2: Voice Trunking on ATM including compressed speech support

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17 Voice over IP (VoIP)

Today the topic of supporting voice over, traditionally, data based networks is one of the hottest around. There is a huge move towards the support of voice over IP, both across Local Area Networks (LANs) and across Wide Area Networks (WANs). This section looks at how voice is supported over IP. It also discusses the idea of running voice over the Internet.

- Introduction to Voice over IP
- A look at H.323 – Terminal equipment, Gatekeepers and Gateways
- A look at H.225.0 and H.245
- How speech, DTMF, signalling etc.. is carried in IP packets
- What are the issues of supporting voice over IP?

End of Training Outline

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