# Voice Technologies for Data Engineers

## Introduction

Today's voice networks are rapidly being integrated into, what traditionally were separate, data networks. Voice over IP is the technology of choice rather than the traditional technologies such as analogue (analog) and TDM (Time Division Multiplexing).

The use of these technologies for the support of voice traffic is not always 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 voice 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 is 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 communication networks.

## Course Length

2 days

## Course Agenda

- Introduction to Voice Communication
- Digital Voice
- The Telephone
- The Telephone Exchange
- Digital Voice Transmission
- E1 Digital Voice Interface
- DS-1 (T1) Digital Voice Interface (optional)
- Signalling on Analogue Interfaces
- Signalling on Digital Interfaces
- Overview of Basic Rate ISDN
- Voice/Speech Compression
- Call Quality and Delay Impairments
- Echo and Echo Control
- Voice over IP

### 1 Introduction to Voice Communication

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
- Section summary and end-of-section review

## 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
- Section summary and end-of-section review

## 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 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 the analogue telephone - 2 to 4 wire hybrid, transmit/receive levels etc.
- Loop disconnect and DTMF signalling - Advantages and disadvantages
- Digital telephone set
- Section summary and end-of-section review

## 4 The Telephone Exchange

This section gives an overview of a TDM digital telephone exchange and looks at how it switches telephone calls, how it routes calls and how it interfaces to other equipments and networks including traditional telephone networks, ISDN and Voice over IP networks.

- Call routing in a digital PBX
- Principles of switching
- Interfaces available - analogue and digital
- Section summary and end-of-section review

## 5 Digital Voice Transmission

Historically, the predominant type of interface used for voice transport within a Wide Area Network today has been the TDM digital interface. There are two 'normal' types of digital interface used, 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.

- The Channel Bank
- An introduction to the E1 and DS-1/T1 digital voice interfaces
- Section summary and end-of-section review

## 6 E1 Digital Voice Interface

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
- Section summary and end-of-section review

## 7 DS-1 (T1) Digital Voice Interface (optional)

The operation of a DS-1 interface is somewhat different to an E1. This section looks at the DS-1 interface in depth.

- Electrical characteristics AMI - B8ZS
- Framing - D4 and Extended Superframe
- Channel Associated Signalling - Robbed bit signalling
- Common Channel Signalling
- DS-1 Alarms
- Section summary and end-of-section review

## 8 Signalling on Analogue Interfaces

Whilst most interfaces between telephone exchanges and other voice networking equipment are digital today, there is still a 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 (Main section)
- Section summary and end-of-section review

## 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)
- Section summary and end-of-section review

## 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
- Section summary and end-of-section review

## 11 Voice/Speech Compression

Historically, voice/speech compression has been one of the prime reasons for integrating a PBX network with a Wide Area Network and has been available on TDM systems for many years. Due to the way in which they operate, Voice over IP technologies are an ideal candidate for using voice/speech compression.

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
- Silence suppression - Speech Activity Detection (SAD), Voice Activity Detection (VAD)
- Mean Opinion Score (MOS)
- Section summary and end-of-section review

## 12 Call Quality and Delay 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 minimisation 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
- Packet Loss Concealment (PLC)
- Speech quality assessment
  - ITU-T G.113
  - The Equipment Impairment Factor Method
  - The E-Model
  - BT Perceptual Analysis Measurement System (PAMS)
  - Perceptual Speech Quality Measurement (PSQM) - P.861
  - PSQM+
  - Perceptual Evaluation of Speech Quality (PESQ) - P.862
  - Perceptual Objective Listening Quality Assessment (POLQA) - P.863
- Section summary and end-of-section review

## 13 Echo and Echo Control

Today, the use of such technologies as Voice over IP as well as techniques including voice compression, are resulting in more and more delay being imposed onto voice connections across 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.168 echo cancellers - Superceded G.165. Current version: G.168 (03/09)
- Section summary and end-of-section review

## 14 Voice over IP

Today, the most common way of implementing new voice networks is through the use of Voice over IP. As this course is primarily aimed at providing a foundation in voice technologies, this section simply provides an introduction to the subject of Voice over IP.

- Introduction to Voice over IP
- The H.323 framework for Voice over IP
  - H.225.0, H.245
- Session Initiation Protocol (SIP)
  - SIP trunking
- The support of voice in IP packets
  - IP / UDP / RTP
- Some potential issues of operating voice over IP
- Section summary and end-of-section review

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