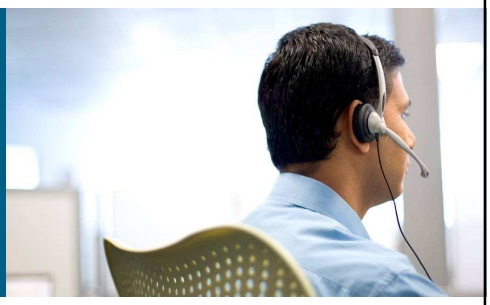
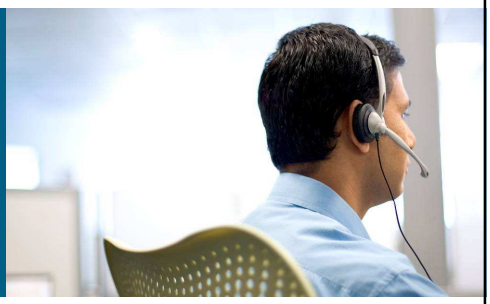
Optimizing Converged Cisco Networks (ONT)



Module 2: Cisco VoIP Implementations

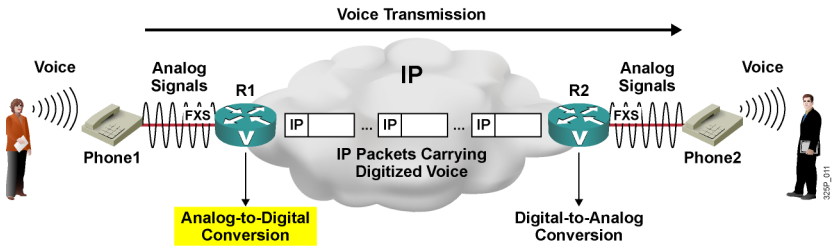
Lesson 2.2: Digitizing and Packetizing Voice



Objectives

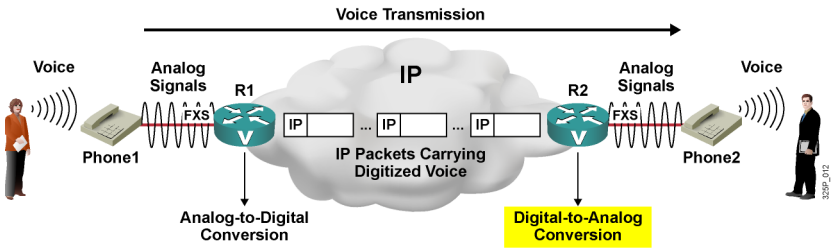
- Describe the process of analog to digital conversion.
- Describe the process of digital to analog conversion.
- Explain how sampling rates are determined using the Nyquist Theorem.
- Explain how quantization can lead to noise.
- Explain how MOS is used to judge voice quality.
- Describe the purpose of DSPs.

Basic Voice Encoding: Converting Analog Signals to Digital Signals



- Step 1: Sample the analog signal.
- Step 2: Quantize sample into a binary expression.
- Step 3: Compress the samples to reduce bandwidth.

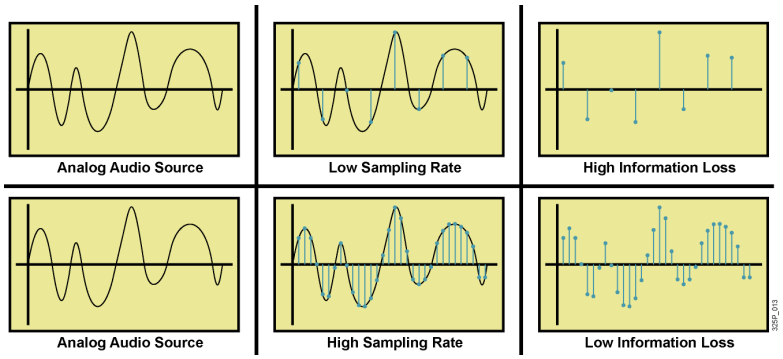
Basic Voice Encoding: Converting Digital Signals to Analog Signals



- Step 1: Decompress the samples.
- Step 2: Decode the samples into voltage amplitudes, rebuilding the PAM signal.
- Step 3: Reconstruct the analog signal from the PAM signals.

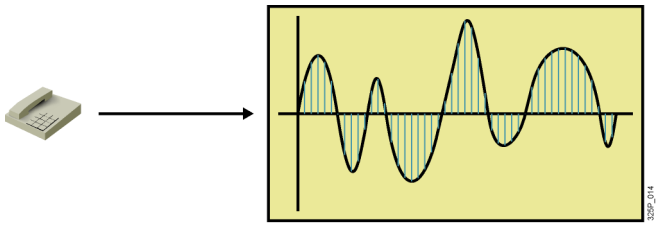
Determining Sampling Rate with the Nyquist Theorem

- The sampling rate affects the quality of the digitized signal.
- Applying the Nyquist theorem determines the minimum sampling rate of analog signals.
- Nyquist theorem requires that the sampling rate has to be at least twice the maximum frequency.



Example: Setting the Correct Voice Sampling Rate

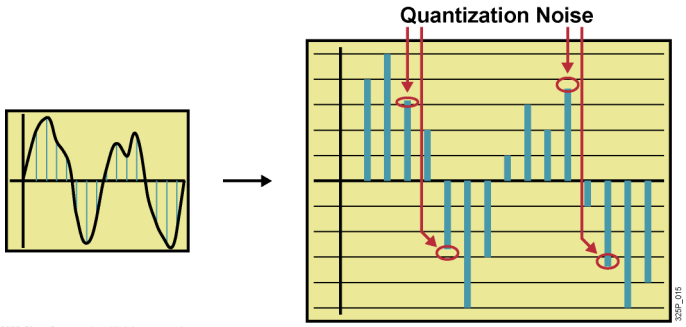
- Human speech uses 200–9000 Hz.
- Human ear can sense 20–20,000 Hz.
- Traditional telephony systems were designed for 300–3400 Hz.
- Sampling rate for digitizing voice was set to 8000 samples per second, allowing frequencies up to 4000 Hz.



8,000 Samples per Second

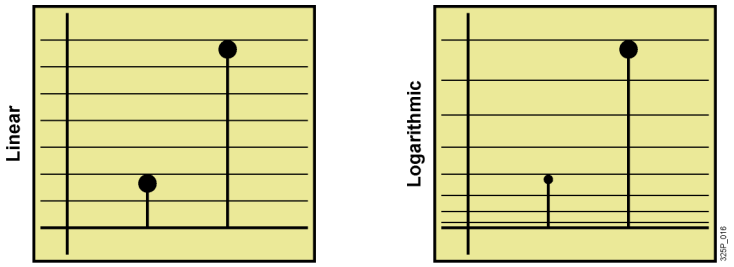
Quantization

- Quantization is the representation of amplitudes by a certain value (step).
- A scale with 256 steps is used for quantization.
- Samples are rounded up or down to the closer step.
- Rounding introduces inexactness (quantization noise).



Quantization Techniques

- Linear quantization:
  - Lower SNR on small signals (worse voice quality)
  - Higher SNR on large signals (better voice quality)
- Logarithmic quantization provides uniform SNR for all signals:
  - Provides higher granularity for lower signals
  - Corresponds to the logarithmic behavior of the human ear



Digital Voice Encoding

- Each sample is encoded using eight bits:
  - One polarity bit
  - Three segment bits
  - Four step bits
- Required bandwidth for one call is 64 kbps (8000 samples per second, 8 bits each).
- Circuit-based telephony networks use TDM to combine multiple 64-kbps channels (DS-0) to a single physical line.

Companding

- Companding — compressing and expanding
- There are two methods of companding:
  - Mu-law, used in Canada, U.S., and Japan
  - A-law, used in other countries
- Both methods use a quasi-logarithmic scale:
  - Logarithmic segment sizes
  - Linear step sizes (within a segment)
- Both methods have eight positive and eight negative segments, with 16 steps per segment.
- An international connection needs to use A-law; mu-to-A conversion is the responsibility of the mu-law country.

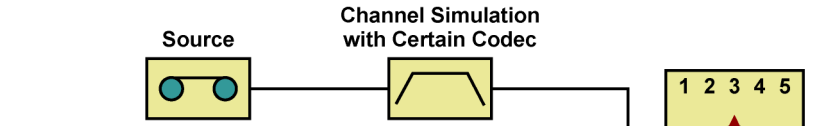
Coding

- Pulse Code Modulation (PCM)
  - Digital representation of analog signal
  - Signal is sampled regularly at uniform levels
  - Basic PCM samples voice 8000 times per second
  - Basis for the entire telephone system digital hierarchy
- Adaptive Differential Pulse Code Modulation
  - Replaces PCM
  - Transmits only the difference between one sample and the next

Common Voice Codec Characteristics

ITU-T Standard	Codec	Bit Rate (kbps)
G.711	PCM	64
G.726	ADPCM	16, 24, 32
G.728	LDCELP (Low Delay CELP)	16
G.729	CS-ACELP	8
G.729A	CS-ACELP, but with less computation	8

Mean Opinion Score



Rating	Speech Quality	Level of Distortion
5	Excellent	Imperceptible
4	Good	Just perceptible but not annoying
3	Fair	Perceptible and slightly annoying
2	Poor	Annoying but not objectionable
1	Unsatisfactory	Very annoying and objectionable

MOS of 4.0 = Toll Quality

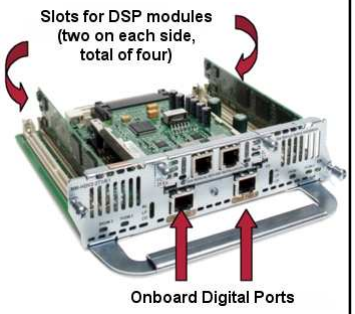
A Closer Look at a DSP

A DSP is a specialized processor used for telephony applications:

- Voice termination:
  - Works as a compander converting analog voice to digital format and back again
  - Provides echo cancellation, VAD, CNG, jitter removal, and other benefits
- Conferencing: Mixes incoming streams from multiple parties
- Transcoding: Translates between voice streams that use different, incompatible codecs

DSP Module

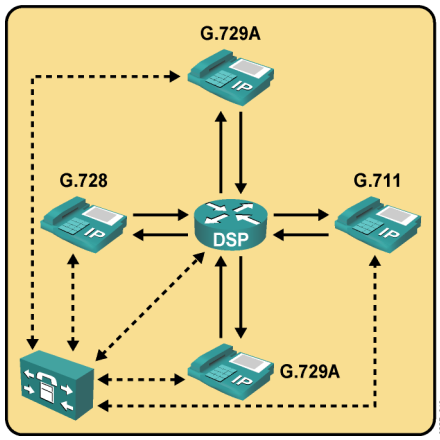
Voice Network Module



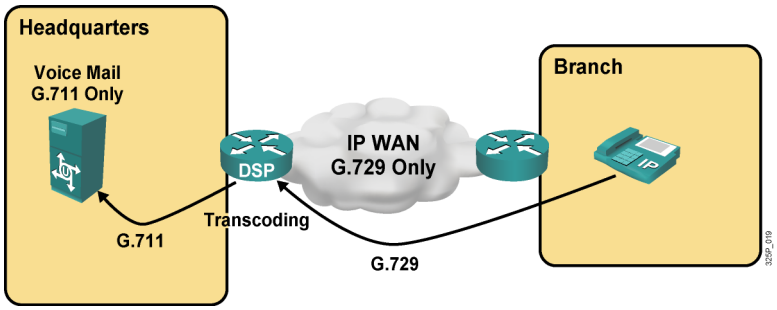
DSP Used for Conferencing

- DSPs can be used in single- or mixed-mode conferences:
  - Mixed mode supports different codecs.
  - Single mode demands that the same codec to be used by all participants.
- Mixed mode has fewer conferences per DSP.

Mixed Mode Conference



Example: DSP Used for Transcoding



Self Check

1. What sampling frequency is recommended by the Nyquist Theorem for reconstruction of a signal?
2. What is the Hz range for traditional telephone systems?
3. What is the implication of using 8 bits for quantization?
4. What is the purpose of logarithmic quantization?
5. What is MOS?

Summary

- Voice-enabled routers convert analog voice signals to digital format for encapsulation in IP packets and transport over IP networks. These packets are converted back to analog at the other end.
- Quantization is the process of selecting binary values to represent voltage levels of voice samples. Quantization errors arise when too few samples are taken.
- There are two methods of companding: Mu-law, used in Canada, U.S., and Japan, and A-law, used in other countries.
- The Mean Opinion Score (MOS) provides a numerical indication of the perceived quality of received media after compression and/or transmission.

Q and A

Resources

- Voice Codec Bandwidth Calculator (requires CCO login)

  http://tools.cisco.com/Support/VBC/do/CodecCalc1.do

- DSP Calculator

  http://www.cisco.com/cgi-bin/Support/DSP/dsp-calc.pl

- Free VoIP Quality Tester

  http://www.testyourvoip.com/