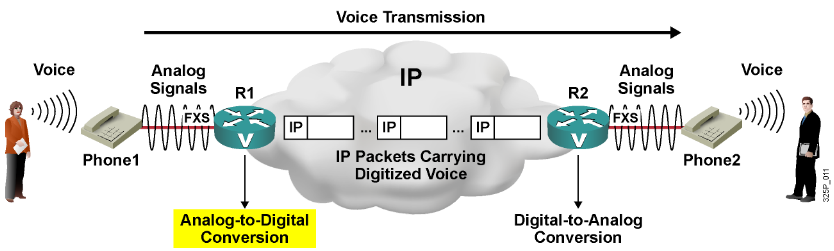
## Basic Voice Encoding: Converting Analog Signals to Digital Signals




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



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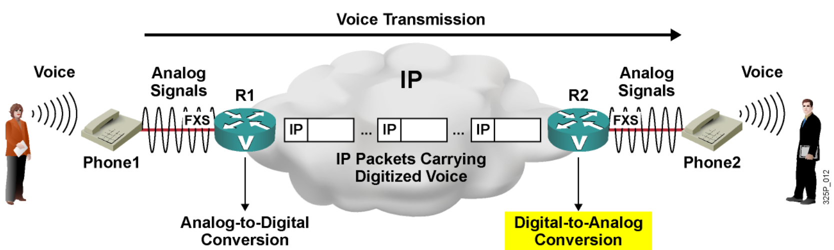
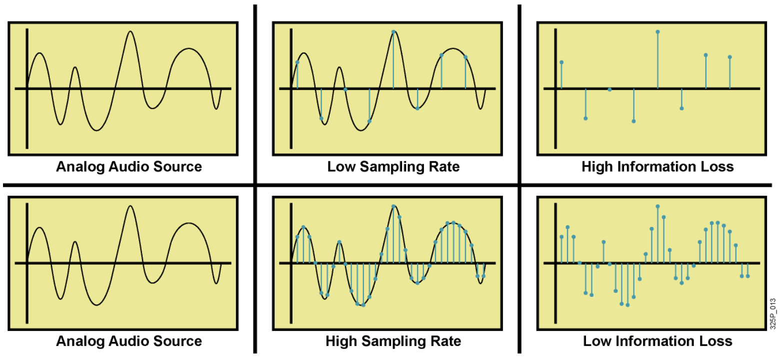



- Step 1: Decompress the samples.
- Step 2: Decode the samples into voltage amplitudes, rebuilding the PAM signal.
- Step 3: Reconstruct the analog signal from the PAM signals.

## Determining Sampling Rate with the Nyquist Theorem

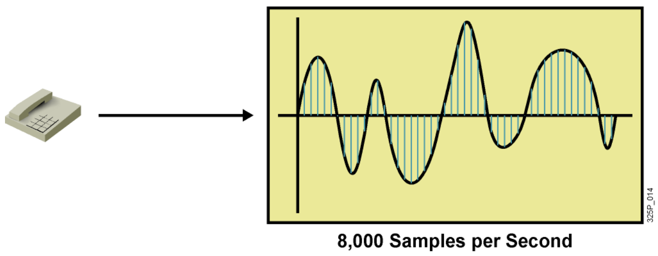
- The sampling rate affects the quality of the digitized signal.
- Applying the Nyquist theorem determines the minimum sampling rate of analog signals.
- Nyquist theorem requires that the sampling rate has to be at least twice the maximum frequency.





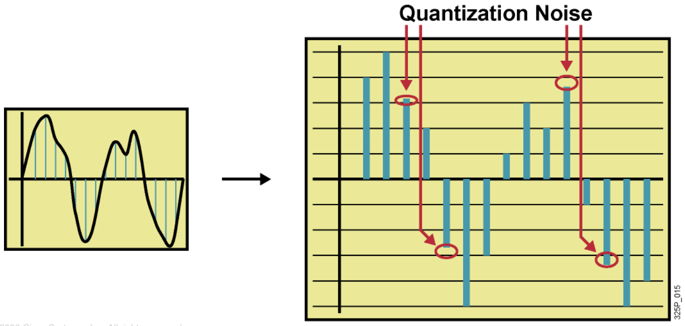

## Example: Setting the Correct Voice Sampling Rate

- 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.



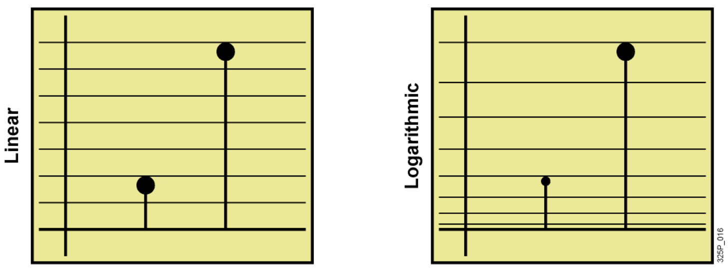
## Quantization

- 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

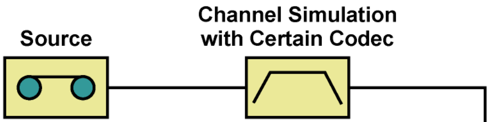
    Transmits only the difference between one sample and the next



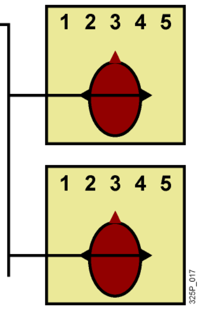
## Common Voice Codec Characteristics

| 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

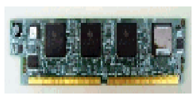


# A Closer Look at a DSP

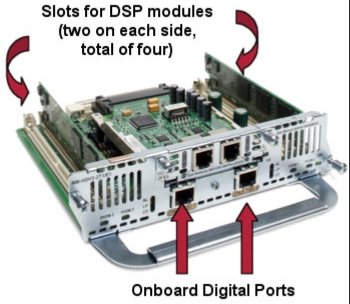
A DSP is a specialized processor used for telephony applications:

- 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

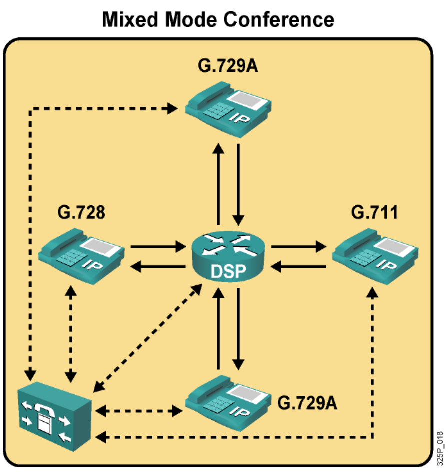


Voice Network Module



## 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.






## Example: DSP Used for Transcoding