Exam 642-436

Cisco: CVOICE

Mega Guide

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This PrepLogic Mega Guide was written by certified subject matter experts and published authors to provide you accurate, in-depth exam coverage. All exam objectives are covered in detail, giving you the knowledge and confidence you need to pass your exam.

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Domain 1 - Describe the Components of a Gateway

Describe the Function of Gateways

The gateway concept is easy for almost everyone as it is simply a translator between one type of language or interface and another. The gateway is responsible for translating the information from the one to the other and vise versa. When talking specifically about an IP voice gateway the relationship is typically from an analog trunk/port or digital trunk/port to a packet based media. This media can be anything which can support IP from an Ethernet connection to a T1 which has IP running over HDLC/PPP. A simple example of this would be from a standard Foreign Exchange Station (FXS) analog port (what would connect to a typical analog telephone) to an Internet connection (Think Vonage). The IP gateway is responsible for taking the traffic from the analog FXS port and encoding it and transmitting it over the Internet connection running IP and in the reverse direction decoding the traffic from the Internet connection and formatting it over the FXS port.

The voice gateway also carries some other duties past simple translation; these involve support for various call control protocols, call setup and teardown, call hold, call transfer, DTMF relay among other duties. On Cisco equipment the call control protocols typically used are Media Gateway Control Protocol (MGCP), H.323, and Session Initiation Protocol (SIP) as well as various other protocols which are supplementary. Included with the support for these call control protocols is the responsibility of encoding and decoding a variety of different codecs which are used to efficiently send this traffic over a digital medium.

Describe DSP Functionality

As stated above one of the duties of the voice gateway is to translate the information from one type of media to the other. The Digital Signal Processors (DSP) job is to take an analog signal and translate it to a digital signal using a number of codecs into an IP stream. Part of this job is terminating the signal, sampling it and using the codecs to packetize it for transmission. This process also happens in reverse, as an IP voice stream comes in the traffic is terminated, decoded and translated onto the line. The gateway also has the capability to translate from one type of IP voice stream to another. The process of speaking between two different voice streams (Different call codecs, bandwidths, sampling rates…) is called transcoding and is also handled by the DSP inside the gateway. The DSP also has the ability to support voice conferencing which allows multiple callers to communicate over a common line.

Traffic Packetization

In order for an analog voice signal to be transmitted across a digital network, it must first be digitized. This is done through a process known as packetization. As shown in the following figure, packetization takes an analog waveform and converts it into a stream of digital 1's and 0's.

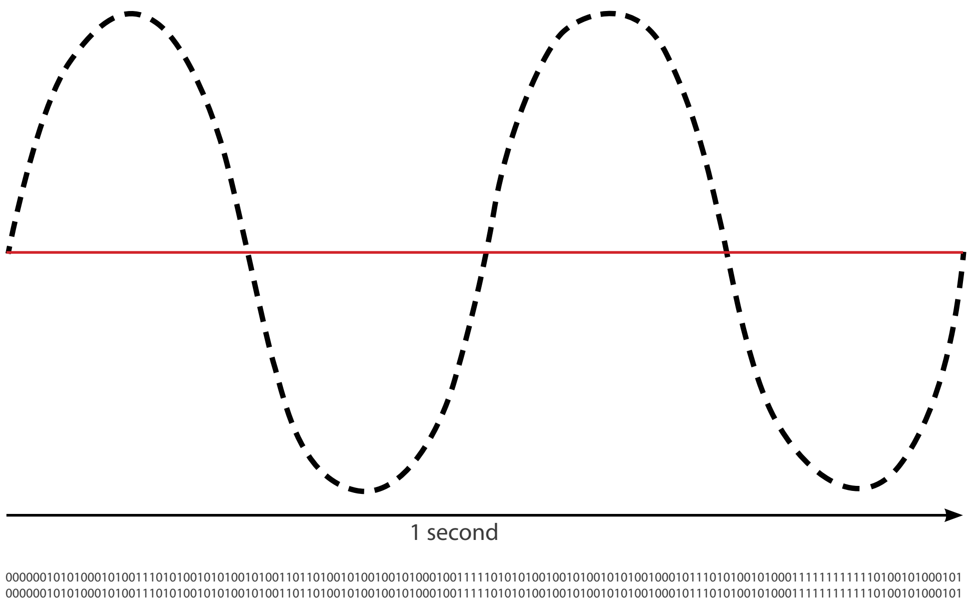

Figure 1 - Packetization

Sampling
The first step in digitizing a signal is to turn the analog wave into something that can be digitized. This is done through sampling. Sampling takes slices of the analog wave at consistent intervals. Within sampling, the Nyquist–Shannon sampling theorem is followed. This theorem states that in order to adequately digitally represent an analog signal, the analog signal must be sampled at a rate of twice the highest analog frequency. Within voice networks the frequency ranges from 300 to 3400 Hz is transmitted, because of simplicity it was decided to sample from 0 to 4000 Hz over digital lines. When following the Nyquist–Shannon sampling theorem, this means that this signal must be sampled at 8000 Hz which translates to 8000 samples per second.

In order to demonstrate this process the following figures have been created. Instead of showing 8000 Hz sampling the following examples show 40 Hz sampling, to keep it simple.

First an analog signal must be separated into pieces, because 40 Hz sampling is being shown, this analog signal is split into 40 different pieces (samples).

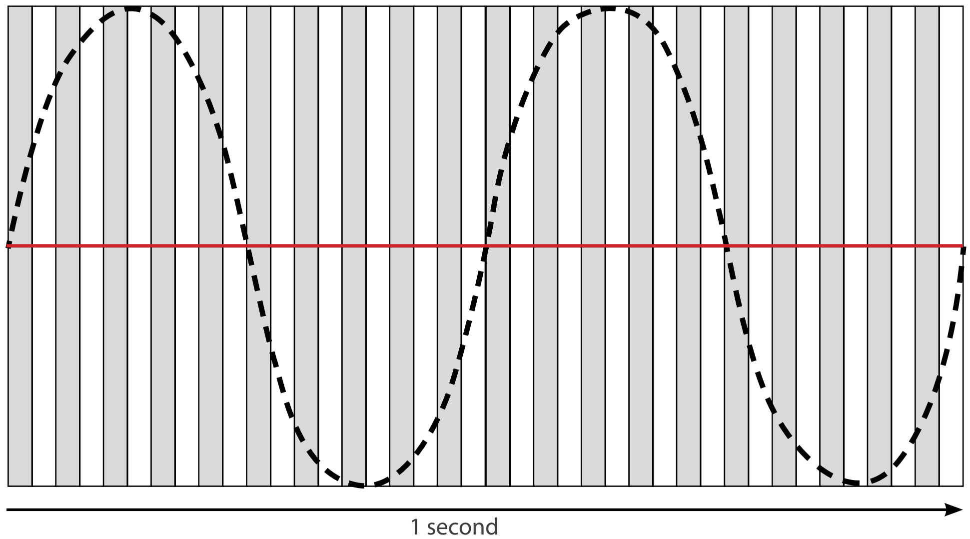

Figure 2 - Separating the Analog Signal

From these, pieces or samples are taken which best represent the analog signal; this is shown in the following figure:

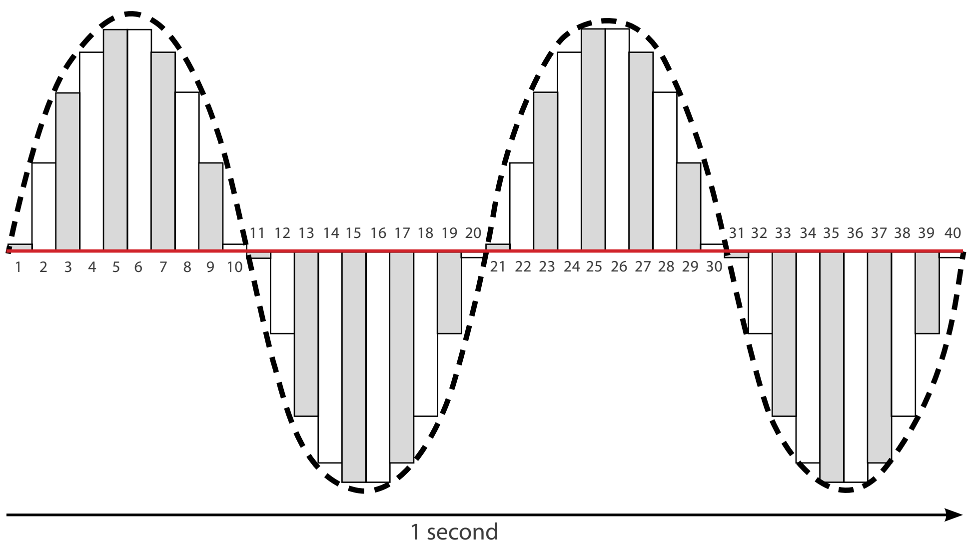

Figure 3 - Creating Samples

From this figure it is seen that the digital samples represent a signal similar to the analog signal being converted. Obviously, the more samples that are taken the more likely the original analog signal the digital representation will be.

Quantization

What this stage of processing does is it calculates a mathematical value for each sample taken, this is also called companding. For the purposes of this example we will describe Pulse Code Modulation (PCM) which is also referred to as G.711. With standard 64 kbps PCM the range of numbers that can be assigned is from -127 to +127 which are from the 8 bits used to record the signal, as seen from the following figure each sample is given a number. When the whole signal is given uniform translation regardless of the level of the signal it is called uniform (or linear) quantization. Uniform quantization results in low level signals having a higher Signal-to-Noise (SNR) ratio then higher level signals and because most signals are lower in nature this is rather inefficient. This problem is remedied with two different companding algorithms, μ-law and a-law.

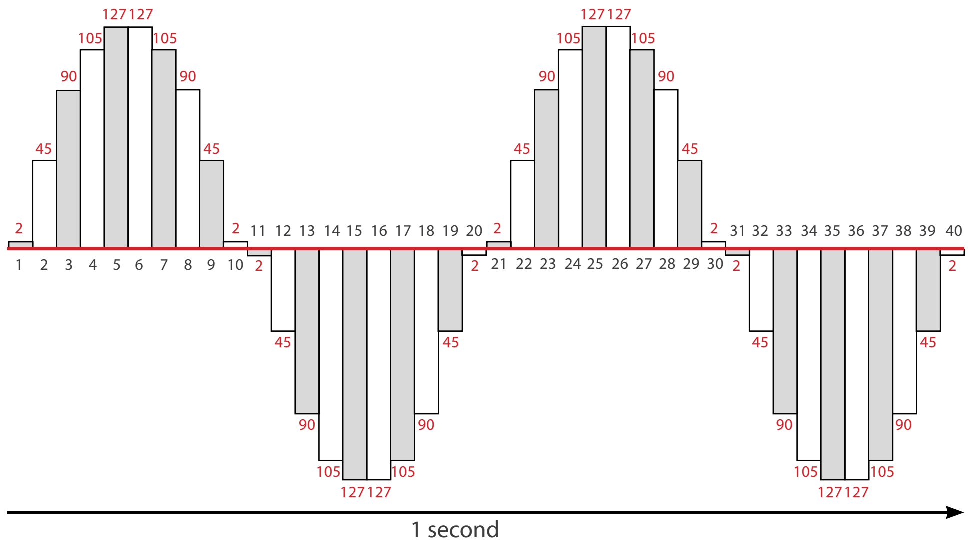

Figure 4 - Quantization

μ-law (mu-law) is the formal standard in North America and in Japan; a-law is what is used in the rest of the international community. These two algorithms work by taking a 14-bit (μ-law) or 13-bit (a-law) PCM sample and mapping it logarithmically to an 8 bit sample. Put simpler, this means that a larger signal is compressed down (14 or 13 bit) to fit in an 8-bit space and in order to remedy the linear quantization problem both μ-law and a-law encode lower level signals at smaller step intervals and higher level signals at higher step intervals. Both of these algorithms effectively increase the Signal to Noise (SNR) ratio of the signal. It is also standard in μ-law countries to convert to a-law in order to communicate with a-law countries.

In order to make these numbers into a stream which can be transmitted digitally as binary, encoding is needed. With PCM, the encoding process takes each number and converts it into a 7-bit binary number with the 1^{st} bit being used to denote the sign (or *polarity*) with 1 meaning negative and 0 meaning positive, the 2^{nd}, 3^{rd}, and 4^{th} bits signifying the *segment*, and the 5^{th}, 6^{th}, 7^{th} and 8^{th} bits signifying the *step*. Once the signal is converted to binary it is run through a digital to digital conversion process which shapes the signal for transmission. The figure on the next page shows this process:

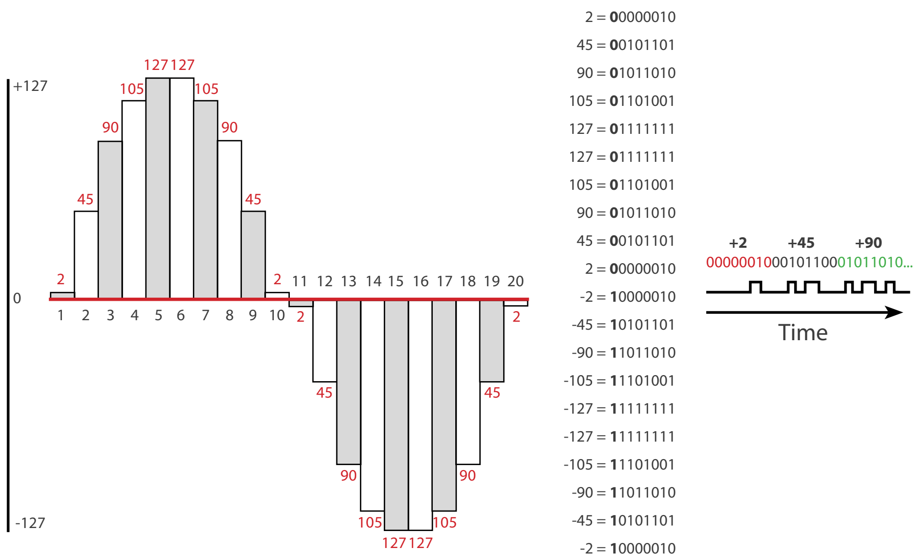

Figure 5 - Binary and Digital-to-Digital Encoding

Depending on the codec which is used on the signal this quantization phase operates in different ways in order to achieve bandwidth savings.

Describe the Different Types of Voice Ports and Their Usage

In order to communicate with any network, interfaces must be used. Within a VoIP network there are two different groups of interfaces; analog and digital.

Analog Interfaces

Inside voice networks there are three different types of analog interfaces; Foreign Exchange Office (FXO), Foreign Exchange Station (FXS) and Earth and Magneto (E&M). FXO and FXS interfaces are used with one another, the FXO interface is connected to the telephony switch and the FXS interface connected to the telephone equipment (phone). When a call comes in, the FXO interface is alerted via ring voltage from the switch then the FXO interface tries to transport the signal to the FXS. The FXS is responsible for receiving the signal from the FXO and providing battery, dial tone and other signaling to the telephone equipment. The E&M interfaces are typically used to connect Private Branch Exchanges (PBX) which exists inside offices. The PBX is essentially a small telephony switch that allows different features to be used inside an office environment; these types of features include extensions, forwarding, and conferencing among others. There are five different types of E&M interface; types I through V (1 through 5). The details of each interface are beyond the scope of this guide but types I and V are the most common; Type I is typical in North America and Type V is typical outside North America.

Analog Line and Trunk Signaling

There are five different types of line and trunk signaling: Loop-start, Ground-start, E&M wink-start, E&M immediate-start and E&M delay-start.