Exam 642-845

ONT

Mega Guide

Prepare With Confidence

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.

PrepLogic

Be Prepared. Be Confident. Get Certified.

Sean Wilkins - Author
Gene Bagwell - Technical Editor

PrepLogic, Inc. • 1300 N. Westshore Boulevard, Suite # 125 • Tampa, Florida 33607
Phone: (813) 769-0920 (800) 418-6789 • Fax: (813) 769-0926 • Website: www.preplogic.com

Domain 1 - Describe Cisco VoIP Implementations

Advantages of VoIP Over Traditional Switches

Voice over IP networks have many advantages over traditional circuit switched voice networks. These advantages include improved bandwidth utilization, consolidated network expenses, and unified messaging integration.

Traditional circuit switched networks utilize a complete 64-kpbs channel; within T carrier networks this is called a DS0. Within a T-1 there are 24 DS0 channels equaling 1.544 Mbps. Packet based voice like VoIP allows more efficient use of this space through compression. This compression allows a number of channels of voice to be transported over a single DS0. The highest level of compression that is currently used is 8:1; meaning a single DS0 can transport 8 channels of voice.

Up until the creation of VoIP, voice and data networks were separate. VoIP enables these two networks to be consolidated into one. Cost is the obvious advantage of having a consolidated network. Having to design and maintain two separate networks is cost prohibitive when it can be done with only one.

VoIP enables unified messaging integration. A phone can become not only a way to talk to someone but also the ability to integrate with video service, database services and several other technologies which enable easier communications throughout a company.

VoIP Components

Like normal telephony systems, VoIP has a number of components which are used to communicate in various ways. These components include phones, gateways, gatekeepers, call agents, Multipoint Control Units (MCU), Digital Signal Processors (DSP) and various application and database servers.

Phones	Just like normal telephony systems, phones are used to communicate via voice. However, with VoIP and Video over IP these phones are extended to not only provide audio communication but also provide video communications.
Gateways	A gateway is a device which is used to communicate between two different types of network. In VoIP applications, this can be used for multiple things including connecting analog phone systems to VoIP systems or connecting from a PBX into a VoIP network. i.e. Using a standard phone over IP (Vonage).
Gatekeepers	Gatekeepers are used for two purposes, to perform some call routing and call lookups (IP to Phone Number), and to perform Call Admission Control (CAC).
Call Agents	Call Agents are used when a central control model is used; this essentially means that call administration is done via a central location. This includes call routing, translations, call setup and teardown among other things.
Multipoint Control Units (MCU)	MCU's are used in conference communications, specifically the MCU is used to combine multiple streams of traffic and join them together seamlessly.

Table continued on next page

Digital Signal Processors (DSP)	Within VoIP, The DSP is used heavily. The DSP's in a VoIP network are used to convert analog signals to digital signals and compress them.
Application and Database Servers	Several different servers can be used in a VoIP implementation; this includes application servers for Call logging and configuration management among others. Database servers can also be integrated with VoIP, this enables XML- based information to be available at the phone level.

Analog & Digital Interfaces

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.

Digital Interface

Within VoIP there are a couple of different digital interfaces; Basic Rate Interface (BRI), T1 and E1 being the main ones. A BRI is used for small office connectivity and provides two channels of voice (64 kbps each), which are called B-Channels and an independent signaling channel (16-kbps), known as the D-Channel. T1 signaling often gets confused because there are two different ways to signal with a T1 interface. The two different ways are Channel-Associated Signaling (CAS) and Common Channel Signaling (CCS). CAS utilizes what is called robbed-bit signaling, this is because T1's are divided into frames which fit into either a Superframe (SF – 12 Frames) or an Extended Superframe (ESF – 24 Frames). Each frame includes 24 timeslots which are used for the 24 T1 channels and each frame includes 8 bits of each channel plus a framing bit. In CAS signaling the 6^{th} and 12^{th} channels have their low order bit "robbed" for use in signaling. CCS signals a completely different way, it utilizes one full 64-kbps channel for signaling and leaves the other 23 channels for traffic; this configuration is typically called a Primary Rate Interface (PRI). Having defined this, BRI is considered a CCS interface utilizing one full 16-kbps signaling channel. E1's all operate the same in a CCS configuration (sometimes they can be incorrectly called CAS). An E1 interface includes a total of 32 channels, with 30 being used for traffic. E1's are split similar to T1's except their framing is a little different. E1's are split into multiframes which include 16 contiguous frames and each frame includes 8-bits of each channel. The 1^{st} and 17^{th} channels are used for frame synchronization and signaling, accordingly. There are no framing bits within an E1 like T1's have.

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. Packetization has four different phases; sampling, quantization, encoding and compression. As shown in the following figure, packetization takes an analog waveform and converts it into a stream of digital 1's and 0's.

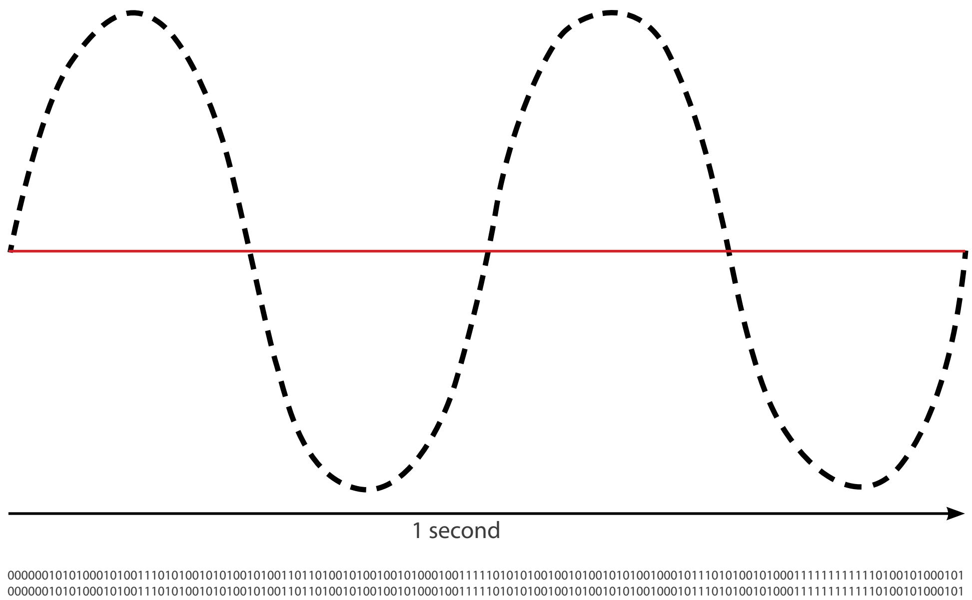

Figure 1 - Packetization

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 which is also called Pulse Amplitude Modulation (PAM); sampling takes slices of the analog wave at consistent intervals. Within sampling, the Nyquist 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 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).

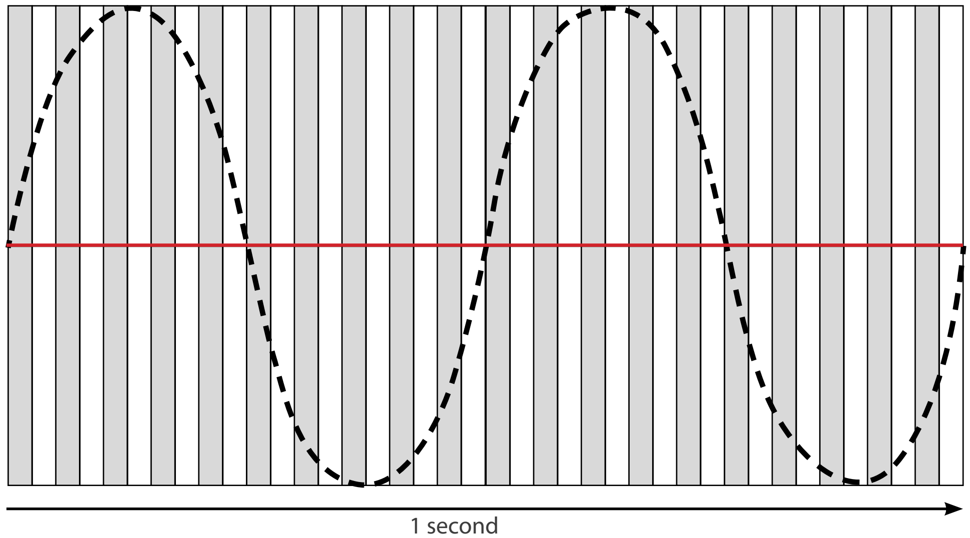

Figure 2 - Separating the Analog Signal

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

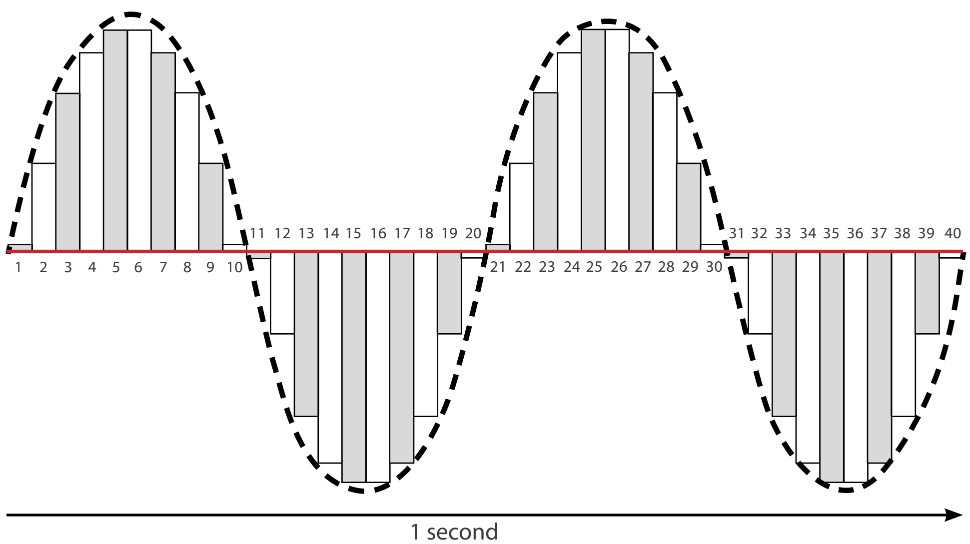

Figure 3 - Creating Samples

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 like 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. The range of numbers that can be assigned is from -127 to +127, as seen from the following figure each sample is given a number.

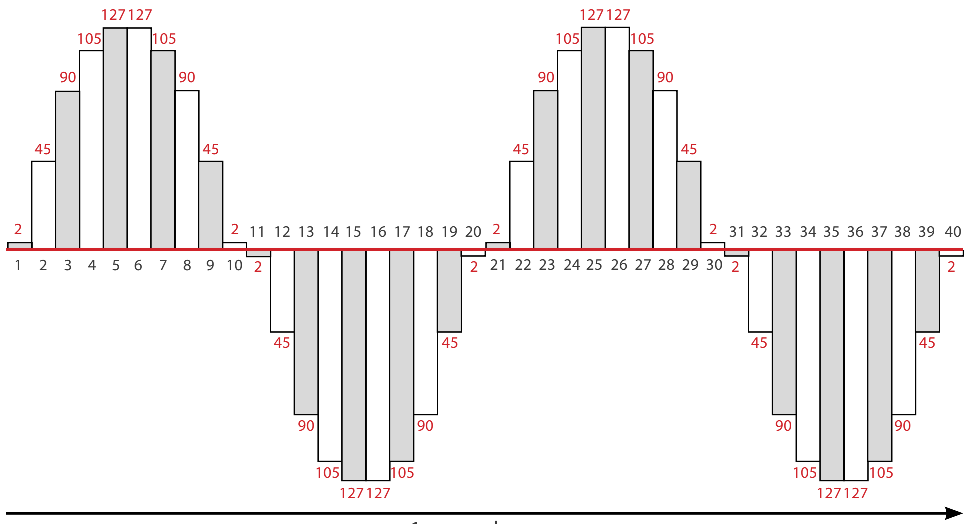

Figure 4 - Quantization

There are two main algorithms which are used to calculate these numbers, μ-law and a-law. 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. It is also standard in μ-law countries to convert to a-law in order to communicate with a-law countries.