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Fonality Special Edition

Cloud VoIP For Dummies®

Learn to:

- Increase productivity with a cloud-based Unified Communications solution
- Save money by avoiding CAPEX
- Access the right person with the right information every time

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Talking Business

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Chapter 1

Introducing VoIP

In This Chapter

- Learning the basics of VoIP
- Getting the most out of your IP Bandwidth
- Familiarizing yourself with great VoIP features
- Collecting it all with Unified Communications (UC)

Cloud VoIP is an amazing technology that couldn't exist if it wasn't for Voice over Internet Protocol (or VoIP as it's commonly known). That breakthrough freed us from traditional telephony where a single phone line could only carry a single phone call.

VoIP calls are transmitted using the basic Internet Protocol (IP) architecture that makes up the fabric of the World Wide Web. The same IP infrastructure used for e-mails and IM chat sessions is now handling billions of phone calls globally every day. VoIP has evolved along with the Internet. It's grown more efficient, continually achieving higher quality standards while opening up telecom features to small and mid-size businesses (SMBs) that used to be the sole domain of Fortune 500 companies. This chapter covers the basics of VoIP and how its structure has enabled it to evolve and become the fastest growing technology of our time.

Understanding VoIP Benefits

VoIP calls look like traditional phone calls at the most basic level. There's a phone on each end and a network of some kind in the middle. A traditional telephony call uses the Public Switched Telephone Network (PSTN) that gives the call its own private bandwidth pipe from end to end. The 56 or 64 kbps allocation is established from the originating phone to the terminating phone for the life of the call, and no other call can traverse that bandwidth during that time.

At least, that's the way it worked for many years. It was a good system, but it didn't make the most of the infrastructure supporting the call. The pair of copper wires that each phone is plugged into can actually support 23 more calls, even with traditional telephony.

The bandwidth within the PSTN was being wasted as well. A 64 kbps pipeline was created every time a call was placed. It's kind of like someone giving you a private two-lane road whenever you and a friend want to drive somewhere. It'd be wonderful, but there's a lot of space in front and behind you on the lane that you simply aren't using.

Maximizing bandwidth

Even if you imagined one car on the road for every word spoken, there's still a lot of space left. Half of the time, one person is listening while the other one talks, so one of the lanes on your private highway is completely empty. Then add in the pauses between words and sentences while you think of what to say next. When you finish, you can see how you still have plenty of free space left on the road.

Figure 1-1 shows how the traditional telephony call has a rigid connection from end to end. The entire length and breadth of the call is using 64 kbps whether someone is speaking or not.

Even the VoIP packets save bandwidth

VoIP is built to sample the voice of the people speaking by using a software tool called a CODEC (short for Coder-Decoder). There are a variety of CODECs available, but they fall into two categories: the standard uncompressed CODEC, or the CODEC with compression. The most common CODEC with compression allows for an 8:1 compression ratio, allowing you to reduce your bandwidth overhead.

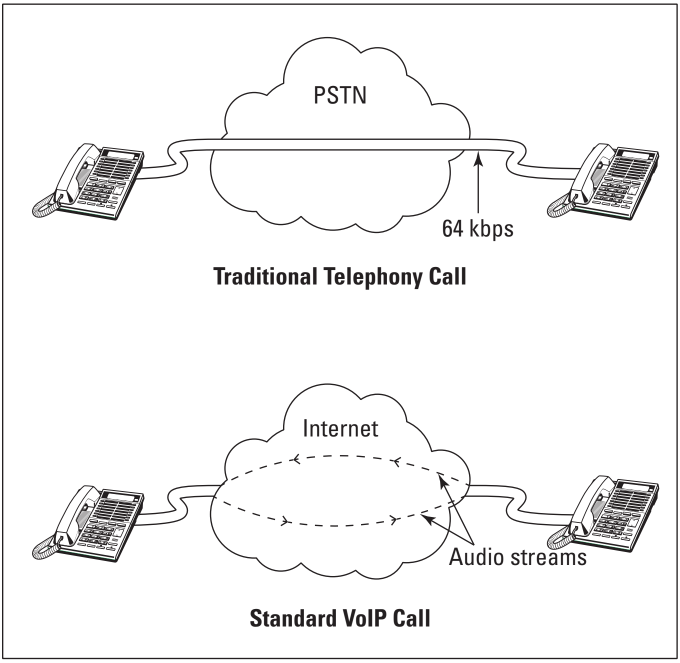

Figure 1-1: A TDM versus VoIP bandwidth comparison.

The standard VoIP call has dashed lines for the audio streams because it doesn't have any bandwidth reserved for its use only. The individual packets of audio pass through the Internet, surrounded by every other VoIP packet as well as e-mails, web browsing commands, and all other varieties of IP traffic.

The packetized nature of VoIP, along with the IP-based architecture in which it operates, gives it a tremendous advantage over traditional telephony. A small business that had 15 individual phone lines and an Internet connection for e-mail and web surfing can now combine all of these over a single Internet connection. The ability to avoid paying roughly $50 per line to simply have dial tone and Caller ID (not including local and long distance minutes) from a traditional local exchange carrier like Verizon or AT&T means an annual savings of up to $9,000.

A time-tested technology

VoIP may seem like it burst on the scene a few years ago, but long-distance carriers have been exploiting the efficiencies of it for decades before it was released to the public. The handwriting was on the wall in the late 1980s as the Internet began to evolve into more than a nationwide network of universities into a commercial and social environment.

The telecom networks had been using a transmission protocol called Signaling System Number 7 (colloquially known as SS7) since the late 1970s. They took much of the structure of this protocol and adapted it to an IP environment, hammering out the kinks of their new VoIP networks. The technology was leaked out to the public in the late 1990s when the Internet Engineering Task Force published the guidelines for the most common version of VoIP today, the Session Initiation Protocol put forth in RFC 2543.

This, and the release of the first open-source VoIP Private Branch Exchange, or PBX, software in the same year, allowed anyone to download and build their own very basic phone system. The next ten years saw an unprecedented growth in VoIP as it became available to everyone.

There were some bumps and bruises along the way, but the industry learned, grew, and elevated VoIP to the same consistent quality of a normal, legacy telephony call. By 2010, maintaining VoIP quality was easy as most of the long-distance networks were gearing much of their networks to VoIP, and eliminating the obstacles that it stumbled on along the way.

Every phone call is a little VoIP

Every long-distance carrier in the U.S. now allows its customers direct VoIP access into its networks. Even if a call comes in with traditional telephony, it probably won't avoid being converted to VoIP before it reaches the phone that's receiving it.

Long-distance carriers don't have direct connections to every telephone in America, much less to every phone in the world. They use a series of underlying carriers to terminate calls to, or from, every active phone on this planet. Many of these underlying carriers have other carriers they use as well, to the