Voice Over Cable

Telephony and other applications

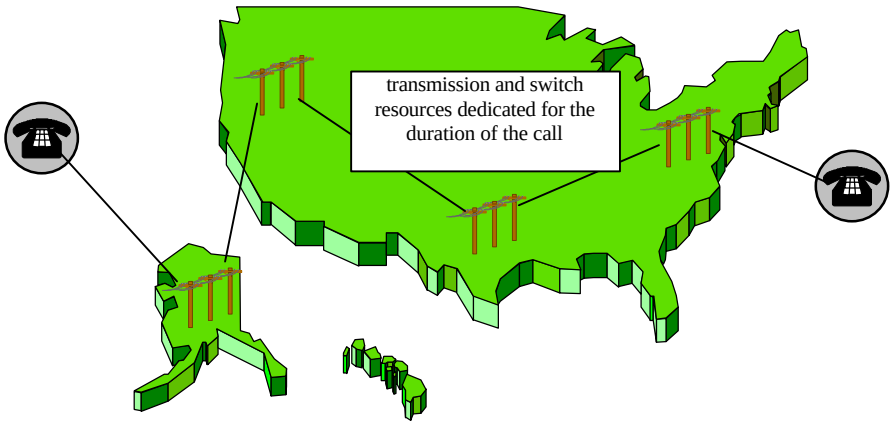
Summary—*A convergence of affordable technology, relaxed governmental regulations, societal comfort with computers, infrastructure growth, and international standards for multimedia over packet-switched networks has created a surge of interest in the transport of digitized voice over traditional data networks. In order to transmit toll-quality voice, with the clarity of sound customers expect from local telephone companies, the network must support an adequate quality of service (QoS). Unless transmission quality can be assured, voice streams on data networks may sound uneven, patchy, or even unintelligible.*

Com21's ComUNITY Access® System supports guaranteed QoS and allows cable operators to offer up to 16 different operator-defined levels of service. Com21 architecture was designed in advance to support toll-quality voice for residential and corporate applications.

Voice over IP (VoIP) and Toll-Quality Voice (TQV)

Introduction

Historically, the transmission of voice streams has been the sole domain of the telephone companies, or *telcos*, using circuit-switched networks. In a circuit-switched network, transmission and switch resources are dedicated for the entire duration of each individual call, guaranteeing voice quality. In the event that the offered load on a circuit-switched network exceeds its capacity, new calls are blocked. The "Sorry, all circuits are busy" message is often heard on high call-volume holidays.

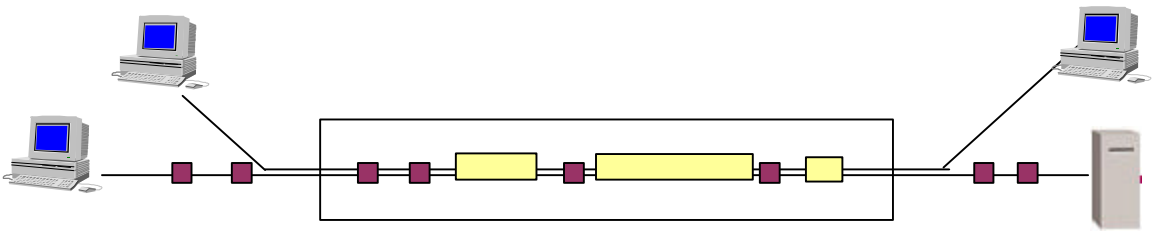


Voice call over circuit-switched telephone network

Digital data streams have, in the past, commonly used circuit switching. More recently, digital services have moved to using packet-switched networks such as IP, Frame Relay, and ATM. Packet-switched networks still often use circuit-switched telephone lines for network access links. Modems (MOdulator/DEModulators) convert digital computer data into analog signals that are carried on the

telephone network and then converted back into digital data at the other end of the modem connection. Common usage of the word *modem* has grown to encompass any technology that connects a remote computer to a network access point—hence digital *ISDN modems* and *cable modems*.

Packet-switched networks utilize bandwidth capacity far more efficiently than do circuit-switched networks. In packet-switched networks, multiple users share all common resources, including transmission and switching capabilities, while in circuit-switched networks, a pair of users ties up a circuit and switch connections regardless of whether or not they are actually transmitting information. Although bandwidth is more efficiently utilized in packet switching, end-to-end performance is less controlled due to statistical variations, creating the potential for packet losses and added latency from switching delays and jitter smoothing techniques.



Data streams sharing a packet-switched network link

Within a digital network, chunks of data (packets) are routed or dynamically switched via data pipes. To illustrate this concept, picture a packet of data as an automobile or truck, and data pipes as roadways. Data packets can be large or small (e.g. buses vs. motorcycles); data streams between a source-destination pair can consist of single packets or multiple packets (e.g. a single commuter vs. a convoy). Data pipes can be large and high-speed (e.g. freeways), or small (e.g. country roads). As the traffic load on a packet-switched network increases, so does the congestion; packet delays increase, packet arrival times become more variable (jitter increases), and data throughput decreases—think about what happens to your commute during rush-hour. The effect of unregulated network congestion on voice streams results in inconsistent and potentially degraded sound quality.

Data networks such as IP and Frame Relay were originally developed for the purpose of transporting large volumes of delay-tolerant data with classless, best-effort service. In general, typical data traffic needs only to be delivered without errors. Performance is measured in terms of throughput; there are no delay or jitter requirements or service provisions. This is the sort of traffic for which these networks are optimized.

ATM (Asynchronous Transfer Mode) is a form of packet switching in which all data is transmitted as short, fixed-length 53 byte packets, commonly called *cells*. This short, fixed-length cell size has multiple benefits. Processing time at switches is decreased since the switch is not required to parse variable length headers, and the large, variable delays caused when very long packets are intermixed with short packets (mixing trailer trucks with motorcycles) are avoided. A second, key differentiating feature of ATM is its built-in support for several classes of QoS. ATM was designed to transport not only data, but also digitized voice, video, and other real-time traffic, trading a small amount of cell overhead in return for short delivery times and the ability to support both guaranteed and best-effort QoS levels.

Activity is underway to rectify IP's limitations. For example, current generation IP (IPv4) can support real-time traffic adequately if bandwidth capacity is over-provisioned, and next generation IP (IPv6) will

include support for prioritized packet flows. But, for the moment, ATM is the only major packet-switched protocol to provide inherent support for voice or video traffic.

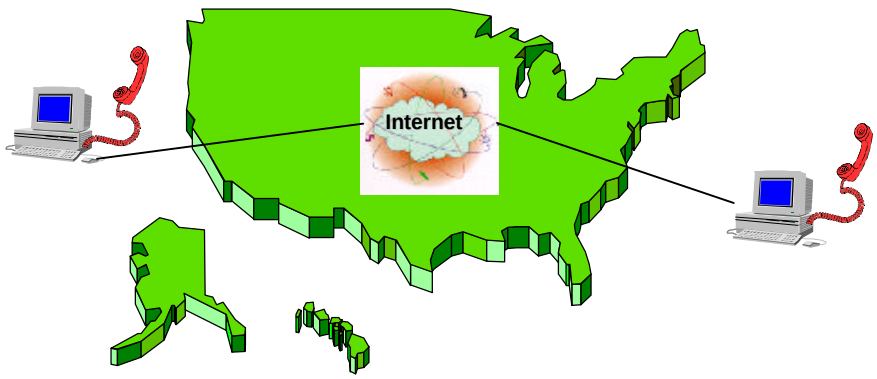
The Rise of Voice over IP

A confluence of affordable technology (such as sub-$1000 multimedia computers), easy-to-use application interfaces (like Web browsers), and a growing amount of useful network-based content has fueled explosive Internet growth over the past few years. IP networking is reaching the critical mass necessary to spawn a new generation of applications that will bring network access into the mainstream, much in the way telephones and television entered our lives in earlier years. In fact, some analysts believe that the Internet now motivates computer purchases. "The number one driving factor in consumer PC purchases is access to the Internet," according to Kiran Narsu, area director and senior analyst at Giga Information Group of Cambridge, Massachusetts.

One Internet application gaining momentum on the acceptance curve is Internet telephony, also known as *IP telephony* or *Voice over IP* (VoIP). A major facilitator for the spread of VoIP is the International Telecommunication Union (ITU) Recommendation H.323 for transmitting multimedia in packet-switched networks. H.323 is a protocol for the transmission of voice, video, and data across a packet-switched network like IP that does not guarantee quality of service. It consists of four parts: video codec (COder/ DECoder) requirements, audio codec requirements, data applications, and system control. While H.323 does not provide QoS guarantees *per se*, it does specify sequence, timestamp, and reliability mechanisms critical to providing real-time voice. A number of VoIP product vendors are planning to conform to the H.323 standard.

Architecture

Computer enthusiasts have been experimenting with computer-to-computer voice communication for years. The only equipment required is microphone/speaker-equipped computers, compatible software, and a network connection. After that, voilá!—free phone calls to any friends anywhere in the world who happen to have compatible software and who happen to be logged on the Internet when you want to talk to them.

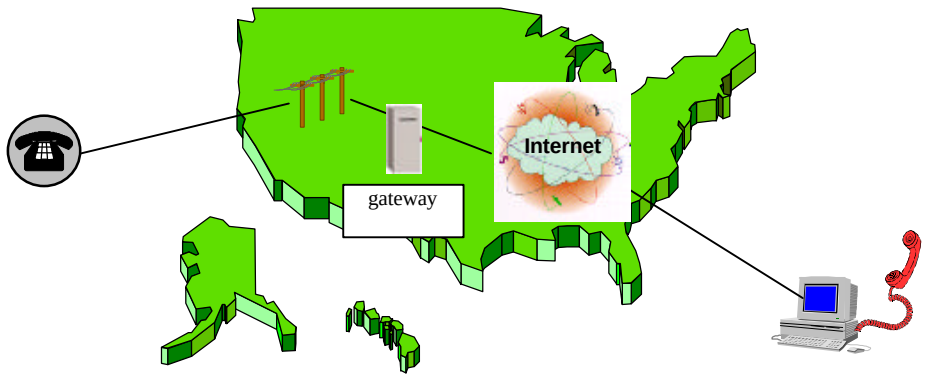


PC-to-PC telephony

Popular software/service packages, some of which include proprietary handsets or plug-in cards that allow you to use a regular telephone handset, include the Intel Phone, NetSpeak's WebPhone, IRISphone, FreeTel, and VocalTec's Internet Phone. Sound quality of calls may vary dramatically, depending on network latency and packet loss, i.e. depending on congestion levels of the portion of the Internet over which a call passes. Software factors, such as compression ratios or echo cancellation can also affect sound quality. PC-to-PC telephony users seldom have complaints, though, since free long-distance service may have its costs, after all.

Another application of voice-over-packet-network is PC-to-telephone telephony. Companies including Vienna Systems and Net2Phone have created gateways that connect an IP connection to the Public Switched Telephone Network (PSTN). This allows calls to travel from a computer, over the Internet to the gateway, and then from the gateway to a telephone over normal telephone lines. This type of equipment is typically sold to companies that offer discount long-distance service to subscribers. Long-distance companies can use their own long-haul infrastructure and bypass local-loop charges (typically two to three cents per minute in the US), and other companies can bypass some fees to long-distance companies by sending traffic as far as possible over the Internet.

Telephone-to-telephone Internet telephony and telephone-to-PC telephony are other services that typically use gateways or some other "black box" to interconnect regular telephone users via the Internet. Aplio, for example, offers its Aplio/Phone which can enable telephone-to-telephone service, as long as each side of the call has an Aplio/Phone and Internet service. Most telephone-to-telephone services use the gateway method.



Telephone-to-computer connectivity over packet-switched network via Internet telephony gateway

Call quality becomes an important issue in the telephone-to-telephone and PC-to-telephone scenarios, since subscribers are often paying for the service. Service for paying customers also has additional complexities such as billing techniques and the need to provide revenue-generating value-added services such as Caller-ID and call-waiting to compete with telco offerings.

Performance Issues

A packet-switched network can transport any form of digitized information. Digitized voice, typically broken down into small packets, has latency, jitter, and packet loss requirements in order to sound "toll-

quality"—i.e. of regular telephone quality. Since IP packetized voice is typically digitized and compressed, correlated packet losses can cause severe degradation in the reconstructed voice stream. Uneven delays (jitter) will cause ebbs and flows in the call quality. A 200ms or greater delay often results in a perceptible pause.

At the local loop portion of an Internet telephone call, call processing hardware such as modems, echo cancellers, or VoIP gateways can impose up to hundreds of milliseconds of delay; phone-to-phone connections require a gateway at either end of the telephone. Beyond the gateways, the network itself imposes delays, which depend on route, location, time-of-day, and many other factors. Typical round-trip packet delay on the Internet can range from 60-200ms in the United States, and from 150ms and up between the United States and Europe or Asia, not including gateway or modem processing delays.

Today's IP networks lack effective mechanisms to guarantee bandwidth or delay/throughput/loss characteristics to data or voice streams. IP, as it exists today, can often support good-to-fair service. But, widespread toll-quality service over today's IP networks is unlikely [3]. Massive infrastructure upgrades, now in progress, will eventually provide large amounts of bandwidth on major pipes, so that QoS may be a moot point for traffic that only travels those routes. (Network planners are aiming at gigabit-per-second backbone capacities, versus megabit-per-second backbone capacities common today). For the near future, however, data rates to the network edges will continue to be low, thus requiring QoS provisions to guarantee acceptable voice quality.

Value-added telephony services, such as Caller-ID and messaging impose additional and more stringent latency requirements (as low as 20ms.) on the local-loop portion of the network.

Opportunities

The short term lure of IP telephony is obvious—low cost telephone service, especially for international calls. But why have major carriers of traditional circuit-switched telephony, such as AT&T and Deutsche Telekom, also joined the rush?

One answer is telephony revenue and market share. Industry analysts have forecasted tremendous growth in the area of digital network-based telephony. Estimates range from a conservative $2B worldwide service and equipment market by the year 2004 to the more aggressive $3B (and 10% worldwide market share) by 2001. The current IP telephony market is estimated at $200M and 0.2% penetration of the global market today. Telcos appear to be joining the Internet telephony fray in fear that if they don't, someone else will.

The appeal of IP telephony extends beyond the inexpensive long-distance phase, though. People are willing to tolerate inconsistent call quality as long as the service is inexpensive and/or the application is for personal, non-lifeline use. Analysts agree, though, that before long, competition will drive down circuit-switched telephone rates; in addition, groups such as the America's Carriers Telecommunication Association (ACTA) are already lobbying for Internet telephony regulation and the elimination of ISP/Internet telephony exemptions from the FCC's Universal Service tariff contributions. (In the United States, Universal Service exemption was granted to ISPs since they only covered computer-to-computer communication. Now, however, Internet telephony also permits telephone-to-telephone communication).

When the inexpensive IP telephony phase ends, providers need to be positioned to compete with circuit-switched telcos offering toll-quality voice *plus* value-added services over a robust, reliable, disaster-proof, fault-tolerant infrastructure. The biggest opportunities will come from being best-positioned to offer integrated services. The ability to provide voice, video, and data services on the same, interoperable network will become mandatory.

Integrated Services

The concept of integrated services (voice, video, and data on the same channel) has been around for a long time. Up until now, though, integrated services haven't gained much popularity because of problems with bandwidth capacity, connectivity, critical mass (not enough) and prices (too high)—look at the problems ISDN (Integrated Services Digital Network) has faced in the United States over the two decades since it was originally proposed.

Now, however, technology has brought us to the point where support for data, video and voice in a single network are feasible. Standards, affordable computing power, and high-speed, high-bandwidth infrastructure are in place. Packet voice/video committees are defining requirements for quality support and interoperability of IP, Frame Relay, and ATM networks. Protocols for supporting the service guarantees necessary for quality packet voice, such as ATM, are available. Even network usage patterns have reached critical mass; 23% of all U.S. households now have Internet access, 60% of which use the Internet every day. Affordable, high-speed access to the home and business is quickly becoming a reality with the advent of cable modems as an alternative to costly private lines or ISDN. The final barrier to widespread proliferation of integrated services appears to be end-to-end QoS guarantees.

Applications

What can be accomplished with an integrated services network? Besides the obvious benefit of price competition between voice, video, and data carriers, there are numerous residential and business applications for unified services. Some examples include:

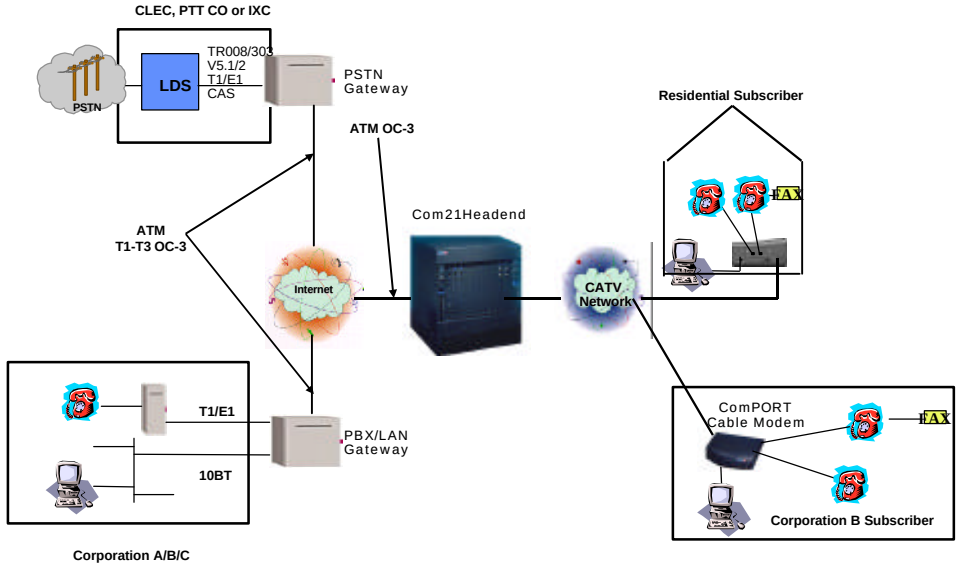
- voice-enhanced web sites with e-commerce or help-desk applications
- business or residential PSTN telephony
- PBX connectivity
- remote call centers
- fax
- video-conferencing

Integrated services can enhance remote collaborative efforts or telecommuting by providing services such as videoconferencing, combined voice and text/graphics calls, file sharing, and toll-quality PBX or PSTN connectivity, all without tying up a telephone line or causing multiple monthly bills. Commercial and e-commerce endeavors could offer more personalized, convenient service by allowing potential customers to click a button on a web site and immediately be connected to a sales representative (who could be working from a home office anywhere in the world). Global toll-free numbers are another application. Finally, as real-time services gain a foothold on the residential markets, home entertainment applications such as on-line gaming and pay-per-use music and video will proliferate.

Com21 Cable Voice Solutions

Cable data access provides a high-speed, high-bandwidth pipeline into subscribers' homes. Currently deployed VoIP equipment already allows PC-to-telephone and PC-to-PC services via HFC access. The next step is to enable telephone-to-telephone IP-quality and *toll-quality* voice via HFC access.

The Com21 ComUNITY Access® System consists of ComPORT cable modems, the fully-featured NMAPS network management software, and a ComCONTROLLER headend channel switch. It is well-positioned to implement toll-quality voice over HFC because its ATM-based architecture supports QoS provisioning *today*, and allows cable operators to offer up to 16 different operator-defined levels of service (with corresponding pricing tiers). The ComUNITY Access System can support Voice over IP (VoIP) as well as Voice over ATM (VoATM), since it functions at a Layer 2 level.



Com21 Voice Over Cable Solution

Also, ComPORT cable modems contain Application Interface Module (AIM) expansion slots, for value-added feature modules. Com21's telephony AIM, currently under development, could be added to a subscriber's modem without having to replace already-deployed equipment. The telephony AIM will use a separate virtual circuit to dynamically assign the necessary bandwidth capacity to voice calls. ComCONTROLLER headend channel equipment will also provide a –48V battery-backed power option for increased system reliability. In addition, the ComUNITY Access System's support of VLANs (Virtual Local Area Networks) can be extended to cable voice users.

Com21will leverage its architecture to support toll-quality voice for residential and corporate applications. The availability of high-speed access and configurable levels of service to the home create an unparalleled opportunity for cable operators to participate in the lucrative voice-application markets.

References and Suggested Reading

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[5] The Global Internet: Parts 1 and 2, *IEEE Comm. Magazine*, May and June 1997.

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[7] Hyperlink to http://www.forrester.com for market analyses (sample keywords: Voice over IP, Telco, Telephone).