Making The Right Choice For Your MDU
Should It Be POTS Or VoIP?

Advantages and disadvantages of the prevalent services being offered today.

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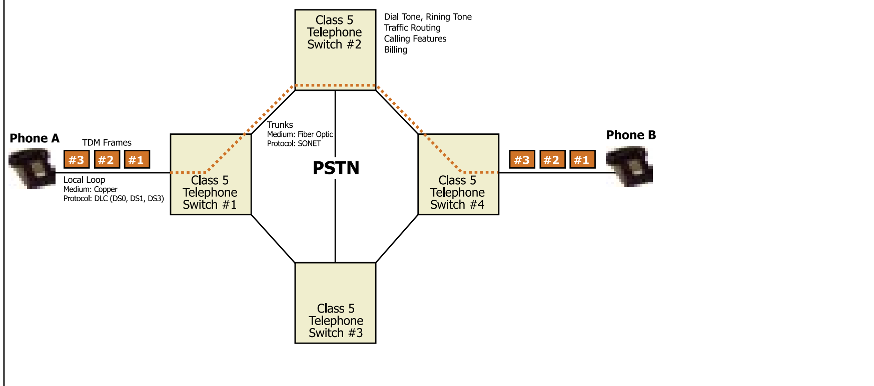
The purpose of this article is not to advocate any one specific method for deploying telephony in MDU (Multi-dwelling Units) properties, but rather to discuss some of the advantages and disadvantages of the prevalent services offered today. Voice services today can generally be classified under two major groups: POTS (Plain Old Telephone Service) and IP Telephony (Internet Protocol Telephony). However, due to certain technical and network limitations, today the closest one can get to IP Telephony is VoIP (Voice over Internet Protocol) which can be considered a hybrid of POTS and IP Telephony.

Figure 1: POTS over PSTN

What We Have Today



The Culmination of Alexander Graham Bell's Experimental Telephone of 1876.

- Transmission path is nailed-down end-to-end before the call.
- Transmission protocol is the synchronous TDM.
- Switch only talks and hears in TDM.
- The PSTN is owned by Baby Bells (heavy tolls apply).

POTS = Plain Old Telephone Service
PSTN = Public Switched Telephone Network
TDM = Time Division Multiplexing
DS0 = Digital Signal Level 0 = 1 Voice Channel
SONET = Synchronous Optical Network

POTS Is Ubiquitous

POTS is the culmination of Alexander Graham Bell's experimental telephone of 1876, intended to transport only voice signals, which evolved during the last 120 years into our most ubiquitous means of telecommunication. During the last 30 years, it has also been adapted to keep pace with changing applications such as transporting data and video signals. Today, POTS is considered a nickname or epithet for reliable, high quality, and feature-rich voice service — service which is also striving to deliver data and video services with low to moderate bandwidth. In the US alone, the industry, dominated by the incumbent telephone providers, generates over $70B in revenue annually.

IP Telephony began in 1995 as the result of work done by hobbyists. The impetus behind the effort was to utilize the practically free Internet network for transporting voice signals, and therefore significantly reducing the cost of transporting voice communications over the private networks owned and operated by telephone companies.

In both POTS and IP Telephony, the human voice, an analog signal, is sampled and digitized into a series of zeros and ones. However, the two primary technological differences between POTS and IP Telephony, and hence the difference in the quality of services provided, are in the transmission protocols and the network topology utilized for transporting these digital signals.

Let's first consider the transmission protocols. In the case of POTS, the digitized signal is "framed" in a fixed size according to a certain arrangement known as TDM (Time Division Multiplexing). The smallest frame size — known as DS0 (Digital Signal Level Zero) — is equivalent to one voice channel, and contains 64 thousand bits per second (64kbs). The next frame size in this hierarchy of digital signals is DS1 (Digital Signal Level One) at 1.544 megabits per second. It contains 24 DS0 frames, and consequently can carry 24 voice channels. T1 or T1 Facility is the common industry misnomer for DS1. In the IP Telephony world, the digitized signal is "packetized" according to Internet Protocol.

The IP packets, also known as datagrams, are not required to be of the same size and can vary in the amount of bits of data they hold. However, the IP packets are required to have a "packet header," typically consisting of 20 bytes, that includes, among other information, the size of the datagram (header+data), source address, destination address, and the number of hops/links the packet can be routed over.

The second fundamental difference between POTS and IP Telephony is the network topology over which the TDM-based frames and IP-based packets are transmitted. In the case of the POTS, the physical path is owned by the telephone company and is part of the regulated Public Switched Telephone Network (PSTN). As depicted in *Figure 1*, Phone A is connected, via the local loop, to a Class 5 telephone switch located in a central office. The central offices are typically connected to one another via redundant paths called trunks.

The primary role of the Class 5 telephone switch is to determine and establish the end-to-end path necessary to complete the call from Phone A to Phone B — not unlike a railway switching control that establishes the traffic paths for trains. In addition to "switching" the paths to connect two end users, the Class 5 switch provides all the call-set up functionalities such as dial tone and ringing tone, and stores all call records such as time and duration for billing purposes. All the calling features, such as Caller ID with name, Call Waiting, and Three-way Calling are also provided by the switch.

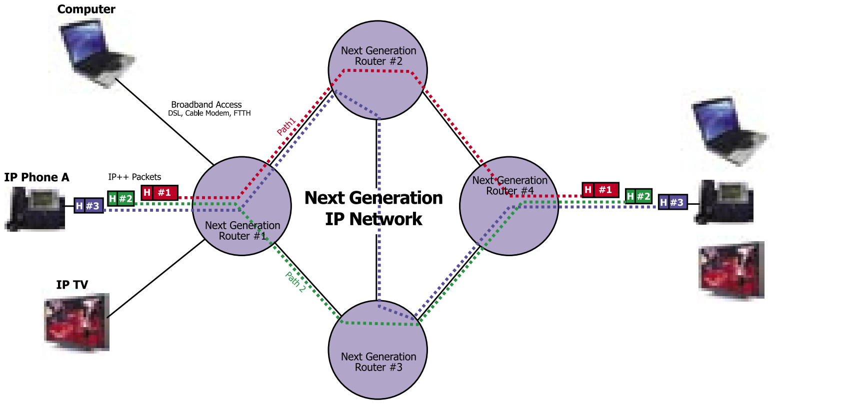
As shown in *Figure 1*, when Phone A calls Phone B, switch #1 establishes the end-to-end path, in this case via Trunks to switches #2 and #4, between the two Phones. This path will remain dedicated to the call and the TDM frames will traverse along this path from Phone A to Phone B.

Four significant issues are worthy of reiteration. First, the path between Telephone A and B is actually "nailed-down" by the switching mechanism before the flow of conversation has started. The path remains dedicated during the calling period to those two subscribers and only to those two subscribers -— the path cannot be shared by other subscribers. Second, the TDM-frames are transmitted sequentially — the first frame leaving Phone A arrives first at Phone B. Third, since it is not unusual for a central office to serve over 100,000 subscribers, the trunks connecting the central offices employ very robust and reliable transmission networks, predominantly utilizing optical fiber paths and self-healing SONET (Synchronous Optical Network) technology, which are costly to deploy and operate.

Fourth, the multi-million dollar Class 5 switch is rather obdurate in adapting to new transmission protocols. Namely, it will only "hear and talk" in a TDM-based language. In summary then, POTS services are delivered by a circuit-switching method where individual bits of a voice signal are sent on a synchronous basis along a nailed-down, end-to-end circuit between a pair of end users.

Figure 2: IP Telephony over Global Information Infrastructure

What We Will have in the Future



Leonard Kleinrock at MIT published the first paper on packet switching theory in July 1961 maintaining that packet switching is a more efficient method of transmitting digital signals compared to the circuit switching.

- IP Packets are spewed out before the path is established.
- Transmission path is established one hop at a time.
- Transmission protocol is the NON-synchronous TCP/IP.
- Router only talks and hears in TCP/IP.
- The Global Information Infrastructure is free.
- One Infrastructure for all IP-based services.

IP = Internet Protocol
TCP = Transmission Control Protocol
H = Header
IP++ = Next Generation of IP Protocol

IP: Many Users Share Path

By contrast, the network topology of IP Telephony, as the name implies, is based on utilizing Internet Protocol where the same physical path can be shared among many different users. In essence, IP packets from various sources destined for various destinations are transmitted without the need to establish a dedicated and nailed-down path.

The path is established by the router one hop at a time by reading the information in the header that contains the end destination and is "torn-down" immediately after the IP packets are delivered to the next router, which is responsible for establishing the next link of the transmission path.

This sharing of the transmission path among many end-users via an open-architecture networking is the underlying technical idea of packet switching. In fact, the beginning of the Internet, which is based on packet switching, can arguably be traced back to Leonard Kleinrock at MIT. He published the first paper on packet switching theory in July 1961, and the first book on the subject in 1964, maintaining that packet switching is a more efficient method of transmitting digital signals than the circuit switching used in telephone networks — then the only method for federating networks.

Since IP packet switching was first developed for connecting computers, whose natural voice, unlike human's analog voice, is digital, it is not surprising that IP packet switching was quickly adopted as the transmission protocol of choice for the Internet.

By the end of 1969, four host computers at UCLA, Stanford, UC Santa Barbara, and the University of Utah were connected together and the budding Internet was off the ground.

The rules for a new version of the protocol which would meet the needs of an open-architecture network, eventually to be called TCP/IP (Transmission Control Protocol/Internet Protocol) was established by Bob Kahn of DARPA (Defense Advanced Research Projects Agency) in 1972 and it has since replaced most other wide-area computer networking protocols and has helped IP to become the dominant transmission method used in the Global Information Infrastructure.

As shown in *Figure 2*, in the ideal world of IP Telephony of the future, the voice signal is translated into a series of IP packets at Phone A and handed off to router #1. The router, which is receiving IP packets from many different sources (telephone, computer, and even television) will then read the destination information for each packet and will establish a hop/link on a "best effort" basis to the next router.

Typically, the IP packets generated at Phone A will take several different paths to go from router to router to reach the final destination. Referring to *Figure 2* for example, it is possible for the sound "h" of the caller's "Hello" at Phone A to traverse via Path #1 and arrive at Phone B after the sound "o" which was transmitted via Path #2.

Obviously, some queuing and synchronization must be exercised at Phone B to make sure that all the sounds are reproduced in the sequence they were spoken.

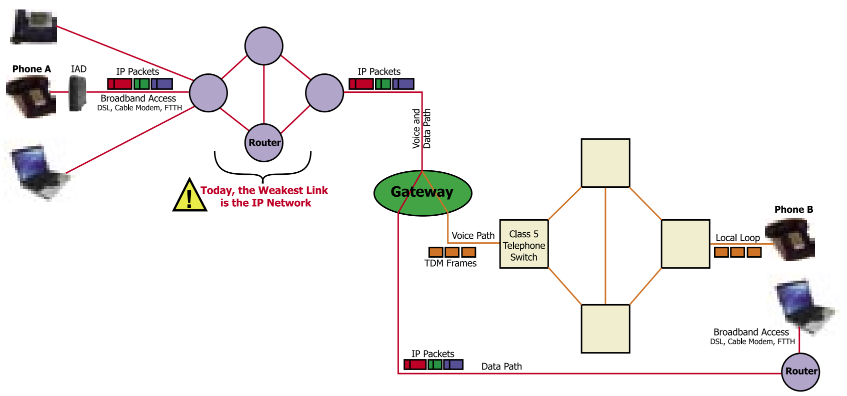
Additionally, many of the functions provided by the Class 5 telephone switch, such as dial tone, ringing tone, calling features (caller ID, Call Waiting, …), and billing information are to be emulated and recorded by the telephone set itself and/or the routing equipment handling the IP packets.

The obvious advantage however, is that the same transmission equipment and path can be used to not only send the voice signals, but also to send the data/video signals — one unified IP-based network serving all three IP-based services.

However, we are some years away from having access to an ideal IP network. In the meantime, a large-scale deployment of IP Telephony is hindered by two rather persistently challenging and stubbornly costly impediments. First, is the fact that not all the telephone users today have access to a reliable and inexpensive broadband IP network. Second, is that today's IP routing technology is not robust enough, and the existing IP networks are not resilient enough, to provide telephone services equal to POTS, which has become the benchmark of quality and reliability against which all new telephone services are compared. The middle ground for the moment is considered to be VoIP, where the idea is to transmit voice signals in IP packets as far as possible before translating them back to TDM frames for delivery to the called party.

Figure 3: Voice Over IP (VoIP)

A Hybrid of POTS and IP Telephony



VoIP Began in 1995 as the Result of Work Done by some Hobbyists in Israel.

- Some network utilization for both Voice and Data.
- The Global Information Infrastructure is not Global yet.
- Next Generation Router Technology is underdeveloped.
- IP++ Protocol is underdeveloped.
- Gateway translation is underdeveloped.
- The Network Topology is as strong as its weakest link.

IAD = Integrated Access Device
IP = Internet Protocol
TDM = Time Division Multiplexing

Referring to *Figure 3*, let's assume that a caller at Phone A, who is subscribed to VoIP, wants to call a person at Phone B who is served by POTS. Because the only path to Phone B is via the Class 5 switch and the local loop owned by the telephone company, the IP packets generated at Phone A and traversing toward Phone B on an IP-based network must be at some point "translated" by a Gateway to TDM frames before being routed via the voice path shown on the figure to the Class 5 switch for delivery to Phone B. Because of this hybrid nature of the VoIP topology, the VoIP service is only as good as the weakest link in the system.

The Network — Weak Link

Today, the weakest link is indeed the IP network itself. Although impressive advances have been made in the last few years in the areas of router technology, software efficiency, and gateway functionalities, the fact is that today's IP networks were primarily designed and utilized for data-centric applications and are not yet optimized for more demanding and delay-sensitive voice and video services. It is due to these technological shortcomings that VoIP services today are typically inferior to POTS services.

For example, the service may not be available during the utility power outage; telephone dial tone may not be present at all the phone jacks on the premise, but only at those jacks equipped with certain external devices such as IADs (Integrated Access Device); the name and address of a caller often cannot be displayed at the 911 emergency dispatch center, potentially defeating the purpose of the call; it is likely that a subscriber in, say, Michigan be assigned a phone number with Florida's area code — an inconvenient arrangement for the subscriber, and an expensive proposition for a friend trying to call the subscriber from the same neighborhood; certain calling features (caller ID with name, speed dialing, *69,…) may not be available; quality of voice is typically not at par with regular phone service — echo and intermittently dropped syllables occur; privacy of the conversation is questionable and may easily be compromised.

Debate Over The Future

There is no doubt that today POTS is the prevalent service of choice among telephone users. But recent advances in VoIP technology have given rise to impassioned debates among the ardent patrons of POTS and VoIP as to which technology will become the dominant service of the future.

A more practical debate, however, for those lacking clairvoyance and not hampered by technological prejudice, is how to financially profit from providing telephone service today, using existing and/or approaching technologies. The fact remains that, given the choice of several services, it is the telephone subscriber who will ultimately decide on what service he or she will sign up for.

And that decision is based on (a) the quality and reliability of service and (b) the price of service. In other words, the key element in providing profitable telephone service is flexibility vis-a-vis what is required by the end users.

It is likely that a VoIP service priced 10-15% below POTS offerings would be attractive on a college campus where subscribers — short on money but computer savvy — are willing to forgo ease of use and quality of service in exchange for lower prices.

It is unlikely however, that residents of luxury apartment complexes or condominium owners would sacrifice service quality and reliability for marginal savings on the monthly fees.

And the flexibility in providing POTS and/or VoIP is dependent on the choice of equipment and network topology used for delivering these services.

Choosing The Equipment

As for the choice of equipment, it is desirable that the equipment be capable of generating the voice signals both in TDM-based frames and IP-based packets. The equipment also should be capable of seamlessly interfacing with both the PSTN and Internet networks.

As for the network topology, many VoIP shortcomings can be avoided by using an IP network designed and managed to handle not only data packets, but also voice packets. This in turn requires using the latest routing technology and advanced IP routing protocols. But in doing so, the transmission cost will be increased and the network reach will be curtailed from a global Internet network to a semi-private network.

For this reason, most of the initial successful deployments of VoIP have been in a corporate WAN (Wide Area Network) environment utilizing a "managed IP network" and not in the mass residential market. ❖

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