

IP INTERCONNECTION FOR MANAGED VOIP

INTERCONNECTING NEXT GENERATION NETWORK SERVICE PROVIDERS

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EXECUTIVE SUMMARY

IP interconnection represents an opportunity for telecommunication service providers to reduce the five-year network costs of supporting voice interconnection with Incumbent Local Exchange Carriers by 98 percent. With greatly simplified networks and software-driven functionality-on-demand, IP Interconnection is the final piece needed to unlock significant economic and operational efficiencies as the PSTN evolves to an all-IP framework.

Conversely, without IP Interconnection the nation's PSTN will falter, as other means of communication supplant it with services that are inferior, but less costly to provide and less expensive for consumers. Quality, reliability, emergency response and service ubiquity will all be in jeopardy as traffic migrates from a PSTN, chained to the high costs of yesterday's technology, to "best effort" services operating under no such mandate for service excellence.

The nation's PSTN is the hallmark of service excellence because each participating service provider explicitly agrees to uphold the network operating characteristics necessary to maintain it when interconnecting to other providers. In this way calls that traverse three, four or even five service providers connecting from calling to called party can be assured to suffer no appreciable degradation in the process.

We are now at a crossroads where, for the first time, the ability of PSTN service providers to embrace new technology, and the economic and operational efficiencies that it brings, is being threatened by a gap in the regulatory framework necessary to efficiently deploy it. Whereas the rules governing how PSTN-participant service providers interconnect their networks have substantially, always kept pace with evolutions in technology, the FCC rules governing such interconnection are now sorely lagging and technologically obsolete. It is not surprising then, that in the absence of updated rules, incumbent LECs have simply refused to update their interconnections to other PSTN-participant service providers.

Without technologically updated interconnections, the benefits of IP technology are drastically reduced. Even if a service provider were to migrate its own network to IP-based technology, it must then convert all traffic to old

technology in order to pass that traffic to the incumbent LEC. The cost of that conversion equipment is not only needless; the equipment itself erodes service quality and eliminates the opportunity to extend innovative services across service provider networks. Further, the reduced cost and greater operational efficiency of fewer, larger and more technologically efficient network interconnections is lost.

Finally, evolutionary changes in the technology of the PSTN, throughout its history, have progressed steadily and are in evidence today. The PSTN is a network that ALREADY uses packet-switching as its exclusive signaling method, manifested in SS7. It ALREADY uses packet-transport protocols as its transport technology of choice, by way of SONET. And, it ALREADY incorporates IP-based softswitches as an efficient replacement for digital switching systems, as evidenced by deployment initiatives of the major incumbent carriers and, equally, the positioning of softswitches by the very manufacturers of the dominant digital switching systems in use today.

The FCC must evolve its interconnection rules in order to fulfill the technology-neutral directives of the Telecommunications Act and free PSTN-participant service providers to reduce costs and innovate while maintaining the service excellence that has always defined the PSTN. COMPTTEL is working hard for its membership to assist and compel the FCC to act upon its mandate.

BACKGROUND

THE PSTN – A STORY OF EVOLUTION

Understanding why IP Interconnection has now become a critical component of the evolving PSTN first requires an understanding of the significant ways in which the PSTN has historically embraced the use of emerging technologies, even packet-based technologies, in order to realize efficiency gains to the benefit of common carriage. Indeed, if we simply consider the past 50 years, the PSTN has evolved dramatically. Before 1957, the PSTN was entirely analog and circuit-switched. This configuration required one set of wires, transiting the distance between the calling and called party, to be dedicated to each telephone call. The inter-office circuits used to complete the connection between the parties were shared and “switched,” tearing down connections between the parties of terminated calls and connecting the parties of new calls.

TRANSPORT SYSTEMS

The device that eventually became the transistor was originally invented by AT&T Bell Labs (William Shockley, John Bardeen and William Brattain) in order to build equipment that could multiplex many calls onto one interoffice line, called a trunk line. This new interoffice transport technology was called the T-Carrier system and, in 1962, was deployed by AT&T to carry the digitally multiplexed calls inside of packets (frames) created using the now standard Digital Signaling technique and hierarchy that forms the basis of TDM voice communications. T-Carrier was gradually replaced as the main transport technology of choice by a new, more efficient transport system using optical fiber and a protocol suite named SONET (Synchronous Optical Networking) in the early 1990s.

SONET gained its footing in 1984, after the divestiture of AT&T, as an attempt to standardize the optical interface between equipment manufactured by different equipment vendors. Since all of AT&T’s equipment was manufactured by its own equipment arm (WECO), there was a need for an optical interconnection standard whereby AT&T’s equipment could interface to another vendor’s equipment, used by the new long distance competitors. This arrangement is known as a “mid-span meet” and is in common use today as an interconnection

method between networks of different carriers. SONET is a second-generation frame (i.e. packetized) transport technology that provides an improvement in operational and economic efficiency over the previous T-Carrier transport system.

SIGNALING SYSTEMS

The PSTN also has evolved its methods for signaling the initiation and termination of telephone calls in order to take advantage of emerging technologies that improve operational and economic efficiencies. The current “Signaling System 7” (SS7) is a “common channel” signaling system that uses a separate packet-switched network of special-purposed computers to place and terminate telephone calls. When an SS7-enabled central office switch needs to build a switched transmission path from its subscriber to a called party on a different central office switch, it relies on the SS7 network to signal the distant switch that an incoming call request is pending. In a successful call attempt, once the called party answers, the respective switches complete the transmission path between the parties.

The SS7 signaling system represented a substantial improvement over the signaling techniques used previously. Methods such as Dial-pulse, Dual-Tone Multi-Frequency, Multi-Frequency and ISDN “D” Channel signaling techniques were neither resilient nor efficient and SS7 was rapidly embraced by incumbents and competitors alike as a way to improve the behavior and resiliency of the PSTN.

In the mid-1990s, further technological evolution of SS7 made local number portability and, by extension, local exchange competition possible. By using SS7 signaling techniques and Advanced Intelligent Network architectures (built on the foundation of SS7) customers were able to port their telephone numbers from one carrier to another. SS7 packet-switched signaling methods were initially considered quite complex yet, today, it serves as the primary protocol and technology to communicate call initiation, identity, progress and termination across virtually every point of interconnection in use between domestic carrier networks.

TELEPHONE SWITCHING SYSTEMS

PSTN switching systems have also evolved dramatically within the past 50 years. From mechanical step-by-step and crossbar switching systems, the Stored Program Control switching systems introduced in the mid-1960s represented the dawn of computerization in the PSTN. The #1ESS was first installed by AT&T in Succasunna, N.J., in 1965 and provided “stored-program” convenience for routing, translations and dialed digits. But it did not provide a digital switching fabric for conversations. Conversations were still carried over reed relay physical switching mechanisms, which were very similar to older crossbar systems.

Digital switching was introduced by way of the Northern Telecom DMS-10, first installed in Fort White, Fla., in October 1977 as the first production digital Class-5 exchange. Advanced Intelligent Network technology evolved the versatility of digital switching systems by influencing call behavior based upon network conditions or subscriber preference. These parameters are communicated to the digital switch by special-purposed computers called Service Control Points using the PSTN’s SS7 packet-switched network.

Softswitches represent another fundamental evolution in the switching technology of the PSTN by bringing Internet Protocol to the switching, signaling and transport function. The major manufacturers of traditional digital switching systems, Lucent (with the LSS[®] or Lucent Soft Switch) and Nortel (with the Communications Server 2000[®]) began offering softswitches to their carrier customers, such as AT&T and Verizon, in the early 2000s as PSTN replacements for their #5ESS and DMS digital switching systems respectively. In 2003 EMBARQ began installing Nortel Communications Server 2000[®] softswitches as DMS replacements in their local exchange areas, including in cities such as Las Vegas.

Traditional digital switching systems connect subscriber lines to transport trunks that, in turn, interconnect numerous switches in the local exchange through a number of “trunk groups.” Switches allocate, and then dedicate subscriber lines and interoffice trunks to a telephone call for the entire duration of the call. Conversely, when a trunk or line is not in use, it sits as an idle, unutilized resource. A significant exercise in complexity, the traffic engineering of these trunk groups applies statistical probability of loss calculations to call traffic

forecasts in order to determine the size of each trunk group (in members) necessary to yield a “lost call” probability of 1 percent in the “busiest hour” of the affected switch’s operation. In order to meet this performance requirement a number of members of each trunk group necessarily remain idle a great deal of the time.

Softswitches use a completely different, and far more efficient, switching and transport framework based on Internet Protocol. First, instead of a “hard-wired” switching matrix where each line or trunk is physically represented, the Internet Protocol allows end points (subscribers), routes (trunk groups) and switches (soft switches) to be identified and individually addressed logically, over the shared physical facilities. In this way, switching complexity, as well as physical interconnectivity of switches, is greatly reduced; while ubiquitous interconnection is made possible with end points, routes and other switches existing simply as a stored list of unique addresses within each switch.

Further, using signaling methods such as Session Initiation Protocol (SIP), softswitches can allocate a portion of the bandwidth of the shared physical facility to a telephone call on an “as needed” basis, even freeing bandwidth when there is a pause in the conversation. In this way, the use of the physical facility is optimized.

These evolutionary changes in the composition of the PSTN exist today. The PSTN, therefore, is a network which ALREADY uses packet-switching as its exclusive signaling method, manifested in SS7. It ALREADY uses packet-transport protocols as its transport technology of choice, by way of SONET. Finally, it ALREADY incorporates IP-based softswitches as an efficient replacement for digital switching systems, as evidenced by deployment initiatives of the major incumbent carriers and, equally, the positioning of softswitches by the very manufacturers of the dominant digital switching systems in use today.

DRIVERS OF TECHNOLOGICAL EVOLUTION

ECONOMIC EFFICIENCIES

Carriers incur both capital costs and operational costs to support today's networks.

Capital costs are one-time costs incurred at the initial time of equipment deployment and also, on an incremental basis, as equipment is upgraded, additional user licenses are activated and additional features are added to the framework in question.

Operational costs increase and decrease on a gradual basis and are normally reflective of the growth (or shrinkage) of the carrier's network, customer base or geographic territory. The costs also may change as a result of gains in efficiency realized through the deployment of emerging technology.

All carriers are on a never-ending quest for emerging technology that has the capacity to improve the economic efficiencies of its business by reducing either capital or operational costs and, ideally, both. Internet Protocol, together with IP Interconnection, has the potential to accomplish both goals relative to voice and video telephony services traversing the PSTN.

OPERATIONAL EFFICIENCIES

Additionally, carriers continually seek operational efficiencies that can improve user-perceived service quality and/or convenience. Shared packet networks must be "managed" if they are to transport delay-sensitive information flows, such as voice/video telephony-over-IP services, at the level of service quality expected from the PSTN. This management is accomplished using various traffic queuing and shaping techniques that result in a deterministic behavior of the network as it pertains to these information flows. It is this management, then, that differentiates these services, allowing them to rise to the level of quality, and to be defined as, *telecommunication services*.

The term "Managed VoIP" shall be used in this white paper to draw a distinction between these services and other, so-called Over-The-Top (OTT) voice/video telephony-over-IP services that are not actively managed and, therefore, do not meet the quality standard of the PSTN. Almost always, however, these

efficiencies can only be realized with an end-to-end view of the service (as perceived by the user) and most often involve interactions between the networks of two or more carriers. Internet Protocol, together with IP Interconnection has the potential to accomplish all of these improvements in operational efficiency.

MARKET-DRIVEN INTEGRATION

Finally, additional economic and operational efficiencies, as well as additional revenue streams, can be garnered through enhanced interoperability with other, non-telecom services and service providers, including content and application service providers. The 3rd Generation Partnership Project, or 3GPP, purports to unite the telecommunications standards bodies as “Organizational Partners” and, as such has built a functional standard to allow access-agnostic networks to communicate with each other and Internet entities using a common control layer as shown in Fig. 1 below.

The IP Multimedia Subsystem achieves advanced interoperation of wireline, wireless and cable network providers, application providers and content providers, using Internet Protocol and SIP, as the main technologies to dramatically expand opportunities for explosive innovation.

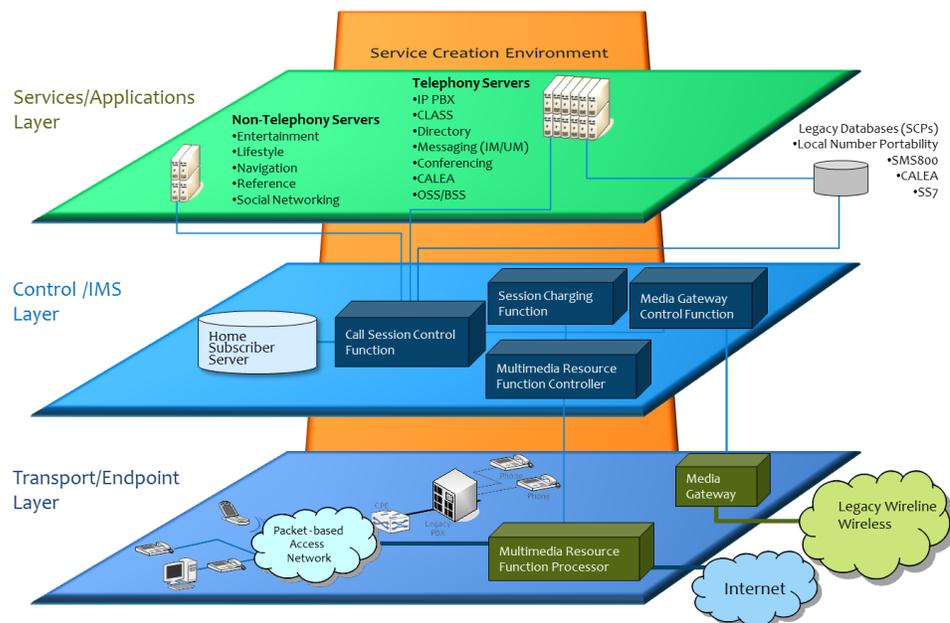


Figure 1. – The IP Multimedia Subsystem Functional Framework

SUMMARY

As evidenced above, technological evolution in the PSTN has always depended upon, and has historically enjoyed, support from a technology-neutral regulatory framework that is adaptable to different technology platforms. This is now expressed in the need for IP Interconnection rules. Like T-Carrier, SONET and SS7 before it, IP Interconnection is a critical necessity in order for the PSTN to take the next efficiency-advancing step in its continuing evolution.

WHAT IS IP INTERCONNECTION?

OVERVIEW

For the purpose of this white paper, IP Interconnection is defined as that which comprises the physical and logical interconnection of carrier networks required to initiate, terminate and/or exchange Managed VoIP services traffic and associated features and functions. Standards for this type of interconnection invoke the well-known, structured approach to computer-to-computer communications known as the ISO Open Systems Interconnection Reference Model (the “OSI Model”) as a means to explain individualized and composite functionality.

THE OSI MODEL

The OSI Model defines seven layers of interaction between a sending end point and a receiving end point, logically separated into different functions. Each functional layer of each side communicates with the equivalent functional layer on the opposite side using protocols and rules specific to and understood by that layer, and that layer alone.

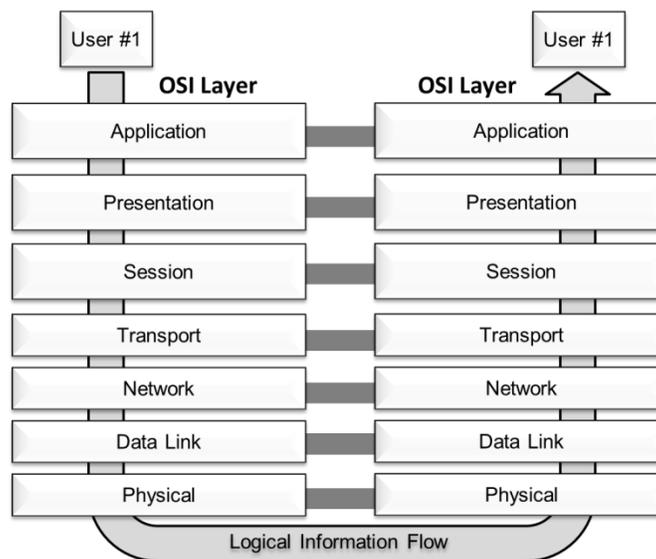


Figure 2. – OSI Reference Model

In order to understand IP Interconnection, one must first understand the concept of constructing a transmission unit (packet) by combining logical functions conducted by all layers, in a progressive manner, for transmission on one or more, physical interconnection link(s) between the parties' networks. Each layer of the receiving party's network then deconstructs the transmission unit, interpreting directives for it and passing the remainder upward to the higher levels for further processing.

IP Interconnection, then, is much more than just the physical continuity between two networks. It comprises the functional support for ALL layers of the OSI reference model according to standardized protocols and rules drafted to support the services in question, in this case, voice/video telephony-over-IP services traffic and associated features and functions. Fig. 3 below depicts the functions and flow of a Managed VoIP service within the context of the OSI reference model.

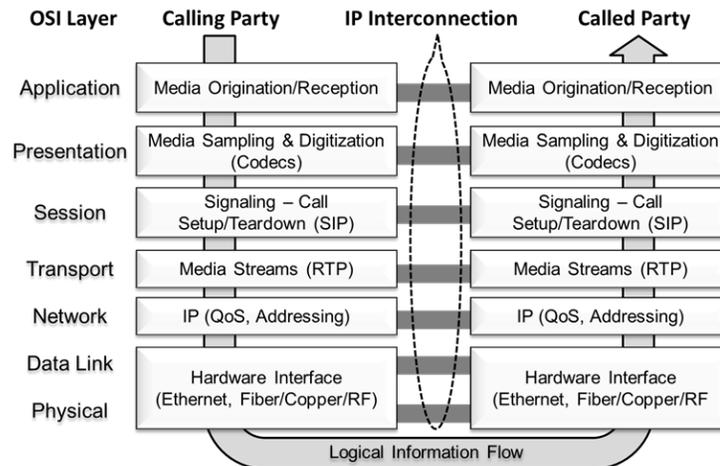


Figure 3. – OSI Reference Model overlay of IP-based voice/video telephony application

The same OSI Reference Model overlay, of course, can be applied to a TDM-based telephone call. The following figure depicts the functions and flow of a TDM-based voice/video telephony service within the context of the OSI reference model.

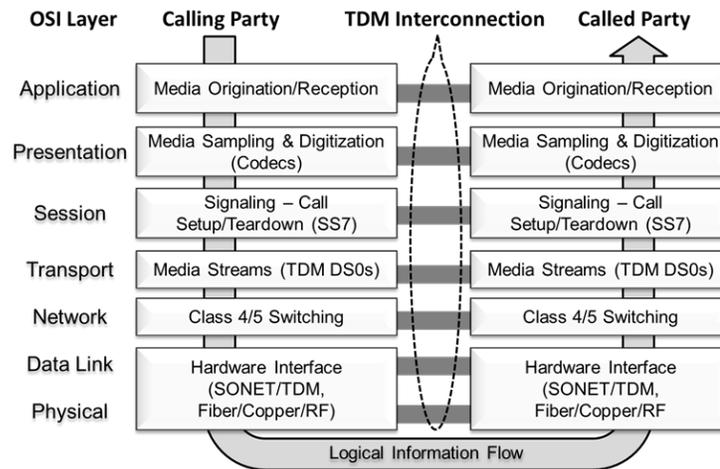


Figure 4. – OSI Reference Model overlay of TDM-based voice/video telephony application

As can be seen from a comparison of the OSI Reference Model overlays for IP-based and TDM-based voice/video telephony services, the same functions and framework exist for both. The difference is in the efficiency with which each function is performed, benefitting from advancements in technology.

TDM-TO-TDM INTERCONNECTION ARCHITECTURE RULES

TDM interconnection is well understood and has evolved to a relatively stable state. Whereas, in the past, advancements in the technology used at any layer would require those changes to be accommodated in the comparable layer of the connected party, those types of advancements have all but ceased in the TDM world. For instance, signaling conversion from SS7 to Multi-frequency (MF) has been performed for years. Likewise, Pulse Code Modulation (PCM) and Adaptive Differential PCM (ADPCM) are well-understood digitization techniques and transcoding capability between the associated codecs is widespread throughout TDM networks worldwide.

Because TDM interconnection architecture is so well understood, conformance of any new technology to this architecture will allow the new technology to

interoperate with TDM networks worldwide. However, though conformance may be possible, it may actually sacrifice the very economic and operational efficiencies that make the new technology attractive in the first place.

Such is the case with the technologies that comprise Managed VoIP. Further, because these new technologies introduce efficiency improvements that are orders of magnitude higher than TDM-based technologies, past PSTN methods of operation would indicate that a modification of the interconnection architecture – to accommodate these new technologies – is in order.

IP-TO-TDM INTERCONNECTION – WHAT DOES IT REQUIRE?

IP-to-TDM interconnection requires functional translation in the upper five layers of the OSI Reference Model. Fortunately, just like “SS7-to-MF” signaling conversion and “PCM-to-ADPCM” transcoding in the TDM world, these required translations are also well understood and, in fact, are conducted in networks today on a very broad scale.

For example, SIP is the most popular and, hence, the de facto standard intercarrier signaling method used today for Managed VoIP communication. SIP-to-SS7 interworking, therefore, is quite common and is accomplished by using encapsulation and translation techniques. It is the subject of an IETF Request For Comment (RFC) standard “RFC 3372 Session Initiation Protocol for Telephones (SIP-T): Context and Architectures”. Within the IP world, this allows a subset of SIP methods defined to conduct basic signaling functions such as call setup/teardown and certain mid-call functions to extend from IP networks into TDM networks and then to allow IP networks to accept directives, for these basic functions, from TDM networks.

Transcoding of voice/video media streams is also necessary for IP-to-TDM interconnection. The ITU-T has issued standards for almost all codecs in widespread use today. Though only a subset of those codecs is compatible with TDM networks, transcoding voice or video from one codec to another is easily accomplished through the use of a media gateway, a device built for that very purpose.

Network layer translations must also be performed in order to convert IP information flows to TDM bit streams. This is also a function accomplished by the media gateway. Fig. 5 depicts the interconnection of two networks, one IP-based and the other TDM-based.

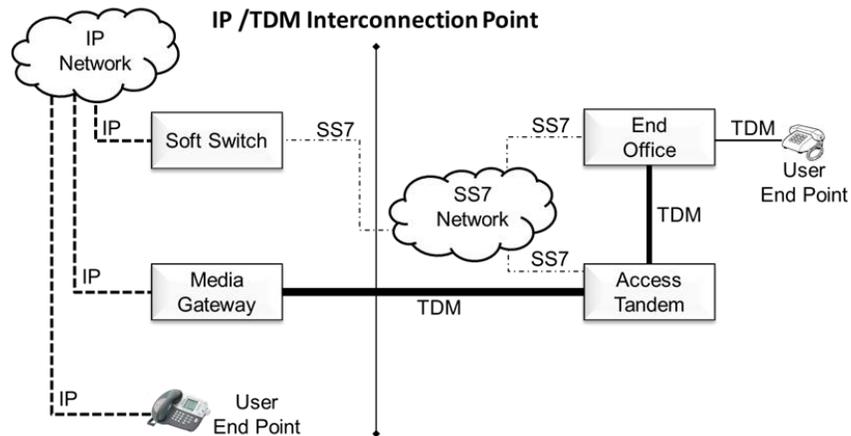


Figure 5. – IP-to-TDM interconnection

IP-TO-IP INTERCONNECTION – WHAT DOES IT REQUIRE?

IP-to-IP interconnection supports two distinct service configurations that possess different operational characteristics: First, when the interconnection supports a call wherein one or both of the end points (users) are connected using a TDM-based subscriber interface to the network (NID) and, second, a configuration where the entire NID-to-NID connection is IP-based.

Fig. 6 depicts the first type of service configuration for IP-to-IP Interconnection, wherein one user end point is IP-based and the other is TDM.

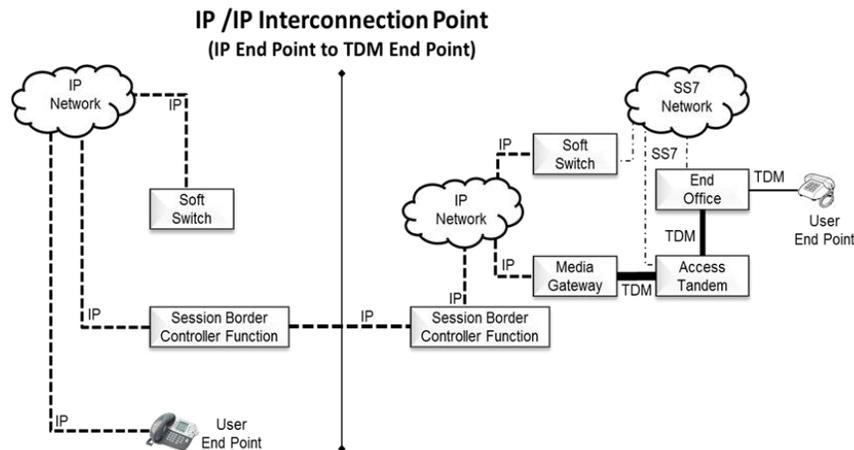


Figure 6. – IP Interconnection when one or both end points (users) are connected using a TDM-based NID

In this service configuration it is necessary for each carrier to exchange SIP signaling information in order to set up, tear down and control individual Managed VoIP calls. It is also necessary for interconnected carriers to exchange admission control, media sampling and digitization, media transport and QoS (quality of service) parameters in order to manage the Managed VoIP information flows to the performance specifications required for telecommunication services.

Additionally, the carrier whose subscriber is TDM-based must perform the functional translations required for IP-to-TDM interconnection (listed above), but within its own network. Care must be exercised on the part of the carrier who is performing IP-to-TDM transcoding, as a certain amount of quality degradation is inherent in the process. While the effect is minimal in a singular transcoding, it can become seriously disruptive to end-to-end call quality if such transcoding is performed multiple times while traversing multiple carrier networks.

Fig. 7 depicts the second type of IP Interconnection service configuration, wherein both end point NIDs are IP-based.

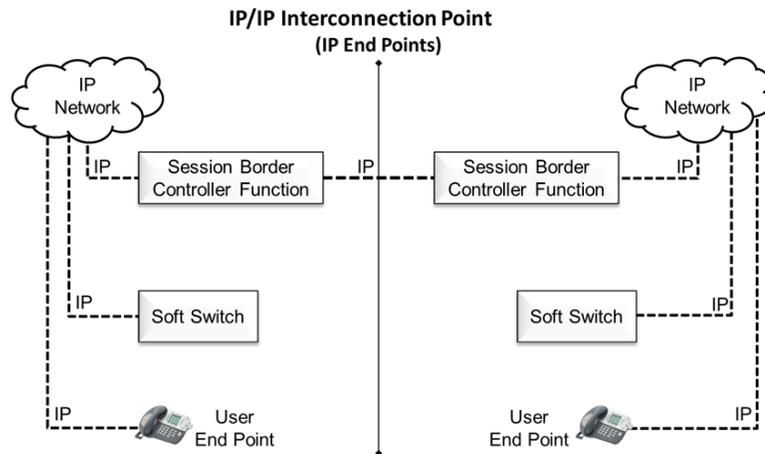


Figure 7. – IP Interconnection when the entire NID-to-NID connection is IP-based.

In this service configuration maximum efficiency is realized. The normal admission control, signaling, media sampling and digitization, media transport and QoS parameters are exchanged via the hardware interface so that calls may flow freely and efficiently across the interconnection point.

Using this configuration, SIP signaling provides both network carriers with expanded functionality that is not possible on a TDM network, using features integral to SIP known as SIP Request Methods. These signaling messages allow each carrier to query the connecting carrier as to the capabilities of their network. In this way, a wealth of additional functionality can be made available to each carrier’s subscribers for the duration of the call.

The following table outlines most popular SIP Request Methods and the IETF RFC standard within which each is defined. New SIP Request Methods are under constant development.

| SIP Requests | | |
|--------------|--|------------|
| Request name | Description | Defined in |
| INVITE | Indicates a client is being invited to participate in a call session. | RFC 3261 |
| ACK | Confirms that the client has received a final response to an INVITE request. | RFC 3261 |
| BYE | Terminates a call and can be sent by either the calling or the called party. | RFC 3261 |
| CANCEL | Cancels any pending request. | RFC 3261 |
| OPTIONS | Queries the capabilities of servers. | RFC 3261 |
| REGISTER | Registers the address listed in the To header field with a SIP server. | RFC 3261 |
| PRACK | Provisional acknowledgement. | RFC 3262 |
| SUBSCRIBE | Subscribes for an Event of Notification from the Notifier. | RFC 3265 |
| NOTIFY | Notify the subscriber of a new Event. | RFC 3265 |
| PUBLISH | Publishes an event to the Server. | RFC 3903 |
| INFO | Sends mid-session information that does not modify the session state. | RFC 6086 |
| REFER | Asks recipient to issue SIP request (call transfer.) | RFC 3515 |
| MESSAGE | Transports instant messages using SIP. | RFC 3428 |
| UPDATE | Modifies the state of a session without changing the state of the dialog. | RFC 3311 |

Table 1. – SIP Request Methods

WHAT IS WRONG WITH IP-TO-TDM INTERCONNECTION?

As mentioned above, IP-to-TDM interconnection requires functional translation in each of the upper five layers of the OSI Reference Model. Though these translations are well understood, they are costly and restrictive. For example, media transcoding adapts the actual voice or video content from its native form of digitization to one compatible with the interconnected network. This transcoding degrades the content and adds cost to network. A media gateway must be deployed to accomplish this function and the media gateway must not only provide the media transcoding but must also provide conversion for the layer-1 physical interface to the TDM network.

As mentioned above, transcoding degrades the quality of voice/video content. If it is done more than once to the same media stream, the resulting call quality degrades exponentially. For example, if a customer of a competitive carrier whose customers are served with an IP network places a call to a Verizon TDM customer through an IP-to-TDM interconnection, only one transcoding would take place. The call quality would be slightly degraded by the transcoding but, in the absence of any other network anomaly, it should meet acceptability

standards. If, however, the call was destined to be received by a Verizon VoIP customer and it was passed through the same IP-to-TDM interconnection point the call would require two transcodings and the call quality could be degraded to the extent it would no longer meet acceptability standards. If the call were to pass through the Verizon Access Tandem network and be further connected to a PIC-selected interexchange carrier who also used IP for transport functionality, the call would undergo a third transcoding, which may very well render it unintelligible.

This scenario is becoming increasingly common, where a competitive local exchange carrier customer PIC-selects an alternative interexchange company (that uses IP transport) and calls a VoIP subscriber of a third carrier. IP Interconnection, of course, resolves the quality issue by preserving the original digitized media content (voice/video) in its native state.

At an average cost of \$85 per voice port, the media gateway function itself represents more than \$50,000 in additional capital cost for each DS-3 connected from the media gateway to the TDM network. Further, because this incremental cost is on the TDM-side of the media gateway it subsumes the same traffic engineering complexity and financial inefficiencies as the TDM network for these “trunk groups.” Because separate physical trunk groups must be built to each and every access tandem (or, alternatively, directly to end offices) in a served LATA, the complexity and inefficiency of such interconnection manifests itself not only in the port costs of the media gateway, but also in the operational cost of the TDM facilities interconnecting the media gateway with the various access tandems and end offices within the LATA; even at TELRIC prices, easily more than \$150 per year per voice port.

Finally, additional costs are not only incurred on the IP carrier-side of the interconnection point, the TDM carrier must also invest in the same number of TDM ports on its interconnected switches. To the extent that the call is placed between two IP-based customers the entirety of these expenses is wasted. As the PSTN evolves to embrace Managed VoIP services, the financial and operational waste caused by IP-to-TDM interconnection will continue to grow if not supplanted by IP Interconnection.

WHAT ARE THE FINANCIAL AND OPERATIONAL BENEFITS OF IP INTERCONNECTION?

IP Interconnection will provide a significant reduction in the number of interconnection ports and facilities required to comprise ubiquitous interconnection. While TDM interconnection requires the 20th century template for a physical appearance of trunk groups to each interconnected switch within the LATA, IP Interconnection uses a common physical connection with unique IP-based addressing of the individual interconnected switches. Further, the unique addressing per switch would not be required, to the extent the receiving network carrier wished to keep its network topology private. All that is actually required is the dialed telephone number of the called party.

To exemplify the potential cost savings of IP Interconnection, assume a LATA within which an IP-based CLEC operates contains four access tandems. The CLEC must build a minimum of three trunk groups per tandem for a total of twelve trunk groups. Assume the CLEC has 20,000 subscribers in the LATA and that their 1,000-minute/month/subscriber call volume is evenly distributed across the 12 trunk groups. In order to maintain a P-grade of service of .01 during busy hour each trunk group would require 314 members. At a cost of \$85 per port the total capital cost of media gateway functionality incurred by this CLEC would be more than \$320,000. At \$150/facility/year, the TELRIC-based interconnection facility would comprise a continuing operational cost of more than \$565,000 per year. The ILEC also would incur the additional capital TDM port costs of approximately \$20 per port, or approximately \$75,000. That brings a five-year support cost of \$3.22 million for IP-to-TDM Interconnection.

With IP port capital costs at approximately \$5,000 per 10Gb port and the cost of the single, redundant interconnection facility at approximately \$1,000 per month, the comparable five-year support cost of IP Interconnection with no media gateway function would be approximately \$65,000, or 2 percent of the cost of IP-to-TDM Interconnection.

These savings are augmented by the aforementioned drastic reduction in environmental requirements. With an equipment footprint reduction on the order of 7:1, and a power consumption reduction that could be as much as 4:1, the environmental costs per location produce significant savings.

The cost savings of IP Interconnection, however, are not limited to a LATA by LATA grooming exercise. Because IP addressing structures allow a table-driven software switch matrix to replace a physical switch matrix, the number of Points of Interconnection (POIs) can be drastically reduced. Whereas IP-to-TDM interconnection requires three trunk groups per interconnected switch, IP Interconnection could be achieved with as few as three POIs to serve the entire domestic U.S. With the emergence of VoIP and a subsequent migration to IP Interconnection millions of unnecessary TDM ports and billions of dollars in capital and operational expenses would, eventually, be saved. This reduction in the number of locations (POIs), which could be as high as 30:1 for a national provider, would also bring environmental costs down by several orders of magnitude.

In addition to cost savings, IP Interconnection will drastically improve call quality of service for those calls that are increasingly originated on an IP network. As mentioned previously, transcoding a media stream from its original digitized form causes degradation. Multiple transcodings exacerbate the problem. IP Interconnection will allow a media stream to remain in its original digitized form from its point of origin to its destination. Assuming the carriers adhere to the previously mentioned network QoS standards and rules (at layer-3), the call quality will suffer no degradation. Further, these Managed VoIP services can take advantage of high-definition codecs that provide voice quality far in excess of the current TDM norms. Indeed CD-quality (or better) audio support may spur future innovative services for deployment on the PSTN.

IP Interconnection also will reduce the operational complexity and cost of network support. Table-driven systems capable of remote configuration, diagnostics and support will allow the replacement of thinly distributed support staff with a robust, centralized operation. With software-driven switching supplanting physical switching devices the ecological cost, in terms of power consumption and facility structure support, will also be drastically reduced.

Finally, the simplified structure enabled by IP Interconnection will allow for flatter, resilient networks wherein route redundancy and business continuity for all operational aspects of common carriage can be accommodated at a fraction of what is now spent by carriers for those functions.

CONCLUSION

There are many reasons that all participants in the PSTN should be moving, quickly, to supplant TDM interconnection with IP Interconnection and Managed VoIP services. The following table will serve to summarize the major points:

| Issue | TDM Interconnection | IP Interconnection |
|----------------------|---|--|
| Advanced Services | Reduces interoperability to lowest common denominator. No ability to access advanced applications or services or to extend those services end-to-end. | Support of SIP signaling and packetized voice gives full and open access to the most robust application and service creation environment available today. Advanced services can be extended end-to-end, across carrier boundaries. |
| Cost Efficiency | Physically oriented – requires many ports and locations to support segregated, partially filled “pipes” of transport. | Logically oriented – dynamically allocated “talk paths.” Bandwidth use is flexible and multipurposed, gaining scale-efficiencies by lowering the number of “pipes” required to move traffic. |
| Network Topology | Many Points of Interface (POIs) are required – at least one per LATA. | Packetized traffic transport is easily consolidated for exchange at fewer POIs. Technology would not preclude as few as three POPs to serve the entire domestic market. |
| Environmental Impact | Multiple locations filled with millions of physical ports consuming large equipment footprints and power capacity. | Few locations required, using logical ports to supplant physical plant. Drastic reduction in locations, equipment footprint and power capacity. |

Table 2. – Interconnection Comparison Table

The need for IP Interconnection, supportive of Managed VoIP services, becomes more urgent each day. To the extent carriers are asked to continue transcoding such services from their native format in order to conform with TDM Interconnection, needless capital and operational expenses are incurred by all carriers. This money could be otherwise spent to deploy broadband facilities, and would serve to further reduce the cost of these emerging technologies as they become more main-stream.

There is no technological development necessary for IP Interconnection. It is a well-understood and widely deployed concept in all other instances but the PSTN. If the PSTN is to continue its evolution as a framework for increasingly efficient voice/video telephony services, IP Interconnection is a critical and irreplicable component.

ABOUT COMPTEL

Based in Washington, D.C., COMPTEL is the leading industry association representing competitive communications service providers and their supplier partners. COMPTEL members are entrepreneurial companies driving technological innovation and creating economic growth through competitive voice, video, and data offerings and the development and deployment of next-generation, IP-based networks and services.

COMPTEL advances its members' interests through trade shows, networking, education, and policy advocacy before Congress, the Federal Communications Commission and the courts. COMPTEL works to ensure that competitive communications providers can continue to offer lower prices, better service, and greater innovation to consumers. For more information, please call (202) 296-6650 or visit www.comptel.org.

ABOUT ETC GROUP, LLC

ETC Group, LLC is a business management and engineering consulting company with significant experience in the management, operation and deployment of a wide range of business models using emerging technologies to support the successful operations of telecommunications and other broadband service providers.

Our team of professionals brings a wide and deep base of knowledge and best practices, derived through more than 200 years of combined, first-hand operating experience in both incumbent and competitive telecommunications companies as well as Internet and application service providers. If you would like to discuss ways in which ETC Group can help your organization, please call (724) 396-0432 or visit our website at www.etcgroup.net.