Circuit Extension over IP: The Evolutionary Approach to Transporting Voice and Legacy Data over IP Networks

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Introduction

High-speed IP-based networks are the latest innovation in the world of communications. The capacity of these networks is increasing at a prodigious rate, fueled by the popularity of the Internet and decreasing costs associated with the technology. Worldwide data traffic volume has already surpassed that of the telephone network, and for many applications, the pricing of IP traffic has dropped below the tariffs associated with traditional TDM service. For this reason, significant effort is being expended on VoIP technologies.

Inherent in all forms of VoIP is revolutionary change, whereby much of the existing telephony infrastructure will be replaced by novel IP-based mechanisms. Despite the hype, this effort has been more protracted and less successful than initially expected. Today's telephony technology, both those portions that VoIP aims to replace and those to which VoIP must interface, is extremely complex. Revolutionary implementations of its hundreds of features and thousands of variations cannot be expected to be developed in a short time frame.

There is, however, an alternative method of exploiting IP networks for telephony service that is evolutionary rather than revolutionary. This method uses IP networks as a drop-in replacement for native TDM networks. It seamlessly interfaces to all existing equipment, such as legacy PBXs and switches, and inherently provides all the hundreds of telephony features and the PSTN quality to which customers have become accustomed. This alternative is circuit extension over IP using TDMoIP.

Why Use IP Networks?

Before delving into the technical details of carrying voice over IP networks rather than traditional TDM networks, it is important to understand why it is worthwhile to do so. The existing telephony infrastructure has an extremely high reliability (99.999%), supports reasonable audio quality (Mean Opinion Score, or MOS, 4.0 on a scale of 1 to 5), has almost universal market penetration, and offers a rich feature set. Accordingly extremely potent incentives are required before one should consider supplanting it. There are two such incentives, one economic and one technological.

The obvious part of the economic advantage of IP networks is shared by all packet networks, namely that multiple packetized data streams can share a circuit, while a TDM timeslot occupies a dedicated circuit for the call's duration. Under “polite conversation” assumption of each party speaking only half of the time, and the “optimal engineering” assumption of minimal overhead, packet networks will on average double the bandwidth efficiency, thus halving operational costs. Taking overhead and peak statistics into account the savings will be somewhat less, but a 30% reduction is attainable. However, it is doubtful whether this savings alone would be a strong enough encouragement to make the switch from TDM to IP.

The added catalyst has to do with the raw rates for data traffic as compared to voice traffic. At present, data communications are metered separately from traditional voice communications and are offered at substantial savings. These savings are partly due to tariffs and access charges that increase the cost of traditional voice services, and partly due to the attractive pricing of IP traffic. Put another way, voice service pricing is still mostly determined by incumbent carriers with high overhead costs, while IP traffic costs are much more competitive as the provider incurs lower costs and is more focused on increasing market share.

The technological incentive has come to be called *convergence.* The reasoning is that technological simplification and synergy will result from consolidation of the various sources into an integrated environment. For example, a single residential information source provisioned for telephony, IP data and entertainment programming would in principle decrease end user prices, result in a single unified billing package, and eventually enable advanced services, such as video on demand.

The Limitations of VoIP

In principle, it would not seem difficult to carry voice over IP networks; a digitized voice signal is simply data and can be carried by a packet network just like any other data. The major technological achievement of the telephone network, that of least cost routing, has its counterpart in IP networks as well. There are, however, two fundamental problems which have to be solved before VoIP can be realistically considered to compete with TDM networks, namely QoS and signaling.

Quality of Service

The meaning of Quality of Service is completely different for data and voice. Although most data can withstand relatively significant delay, low delay and proper time ordering of the signal are critical for voice applications, even though loss of a few milliseconds of signal is usually not noticeable. These requirements are completely at odds with the basic principles of IP networks (although not necessarily with those of other packet networks). To overcome these constraints, mechanisms such as tunneling and jitter buffers need to be employed. Additional components of voice quality such as echo cancellation and voice compression are not inherent in data-based networks at all, and need to be added *ad hoc* for VoIP.

Almost all of the massive R&D effort in the field of VoIP is directed towards solving these QoS problems, leaving the signaling problem largely unsolved. By signaling, we mean the exchange of information needed for a telephone call other than the speech itself. Signaling consists of basic features such as the fact that the phone is off-hook or needs to ring; more advanced properties required for reaching the proper destination and billing; and still more sophisticated characteristics, such as caller identification, call forwarding, and conference calls; as well as more recent additions necessitated by intelligent networking. There are literally thousands of such telephony features, with dozens of national and local variations. Phone customers are mostly unaware of this complexity, at least until you try to deprive them of any of the features to which they have become accustomed.

Adding auxiliary information to digital voice on an IP network is in principle much simpler than signaling in telephone networks. One needn't “rob bits” or dedicate CAS channels. One needs only send the signaling data in some appropriate format along with the voice. Indeed, the advantage of VoIP is that it becomes possible to add features that could not exist in the classic telephony world, for example video and “whiteboards.” This is true as long as the two sides to the conversation are using special VoIP terminals or computers. The problems arise when one must interface between the IP network and the standard telephony network, a connection that is imperative in view of the universal availability of standard telephone sets.

VoIP enthusiasts often emphasize conversations between two PC users or a PC user conversing with a telephone user. Consider instead a conversation between two telephone users, each connected via a standard local loop to a central office, but with an IP based network replacing the TDM network between the central offices. To properly pass the requisite signaling, the IP network has to be enhanced to handle all the thousands of features and their variations (for example, 911 and *67 service). Although not an impossible task, it is one that VoIP developers have not yet accomplished.

TDMoIP Technology

Concept, Bandwidth and End-to-End Delay

In order to explain the principles of TDMoIP we need first to recall those of TDM. A T1 frame consists of 24 single byte timeslots and a single synchronization bit, for a total of 193 bits. An E1 frame consists of precisely 32 bytes (256 bits), one of which is used for synchronization and one traditionally reserved for signaling. In both cases, frames are transmitted 8,000 times per second.

The simplest implementation of TDMoIP encapsulates each T1 or E1 frame in an IP packet by tacking on the appropriate header. Since the packets provide the segmentation, the synchronization bit/byte need not be included. Accordingly, the payload length is 24 or 31 bytes for T1 or E1, respectively. For reliable connection-oriented service one might consider using TCP/IP, which requires a 20-byte TCP header and a 20-byte IP header, for a total of 40 overhead bytes per packet. The end-to-end reliability offered by TCP, however, is not useful for voice packets since re-transmitted voice packets will reach the receiving side out of order and be dropped anyway. A more reasonable alternative would be the real-time transport protocol RTP, with its header of at least 12 bytes, to which one must add an 8-byte UDP header and the IP header, resulting in the same overhead. A 40-byte overhead for a payload of 24 or 31 bytes is indeed a bit extravagant, but there are two solutions to this problem.

The first solution involves header compression schemes. RFCs exist that reduce the average header of both TCP and RTP to only three bytes, diminishing the overhead percentage to between eight and nine percent.

The second solution involves grouping together multiple frames into a super-frame before encapsulation. For example, grouping eight T1 (E1) frames results in a payload of 192 (248) bytes, so that the overhead percentage drops to a reasonable 17 (14) percent. Grouping does add a certain amount of buffering delay, but since each frame is only 125 μsec in duration, this latency is negligible as compared with that of VoIP systems. For example, a super-frame comprised of eight successive frames introduces a one millisecond one-way delay, about half that of the standard 16 kbps *low delay* encoder used in VoIP, and much lower than the 15 millisecond delay of the 8 kbps encoder.

Simple encapsulation of the raw frames is not the only way of implementing TDMoIP. Alternative approaches first encode the TDM data in some other protocol before IP encapsulation. Why would we want to impose another layer of protocol between the TDM and the IP? There may be many advantages. Intermediate encoding may be employed when the natural TDM-induced frame sizes are not appropriate; to provide error correction; to enable interoperability with other systems; and to allow us to compress the speech or to enhance QoS.

<u>Whatever the details, it is important to note that TDMoIP in unframed mode transparently transports the TDM frame without any attempt at interpreting the data. It is completely oblivious to such TDM internals as time slots, signaling channels, etc. Thus TDMoIP can be used to transport arbitrary T1/E1 services, even if some of the channels are actually used to carry data or if the entire frame is an unstructured bit-stream. Similarly, the basic TDMoIP concept is easily extended to fractional T1 or channelized E1 systems. To reduce traffic, only the information carrying bytes need be included in the IP packet</u>

Signaling

How does TDMoIP solve the signaling problem inherent in interfacing between IP networks and the telephony network? To answer this question we should differentiate between three types of signaling: in-band, CAS and CCS.

In-band signaling, as its name implies, is transferred in the same audio band as the speech. It can take the form of call progress tones such as dial tone and ring back, DTMF tones, FSK for caller identification, MFR1 in North America or MFCR2 in Europe, etc. Since these are all audible tones, they are encoded in the TDM timeslot and automatically forwarded by TDMoIP. Speech compression algorithms used by VoIP systems often do not pass these tones well. Therefore, VoIP systems need to implement tone relay protocols to ensure that in-band signaling functions properly.

The most common CAS, or channel associated signaling, is carried in the same T1 or E1 frame as the voice signals, but not in the speech band. T1 robs bits for this purpose while E1 devotes an entire time slot to carry four bits for each of the 30 remaining channels. Since CAS bits are carried in the same T1 or E1 stream, once again they are automatically forwarded by TDMoIP. VoIP systems would need to detect the CAS bits, interpret them according to the appropriate protocol, send them through the IP network using some messaging protocol and finally regenerate and recombine them at the far end.

SS7 is a CCS, or common channel signaling, method. SS7 links are 56 or 64 kbps data links and are thus often found occupying a TDM timeslot. In such cases, they are automatically forwarded by TDMoIP. If not, one can get the required information already in IP format from a SS7 signaling gateway, and readily forward it as additional traffic through the network without any further processing.

Synchronization

Up to now we have ignored another function available in customary TDM networks, that of time synchronization. In the public switched telephone network – and in SONET/SDH networks – one node, called the clock master, provides a time reference to the other, called the slave. Somewhere in the network there is at least one extremely accurate *primary reference clock*, with long term accuracy of one part in 10^{11}. This node, the accuracy of which is called *stratum 1*, provides the reference clock to secondary nodes with *stratum 2* accuracy, and these in turn provide a time reference to *stratum 3* nodes. This hierarchy of time synchronization is essential for the proper functioning of the network as a whole.

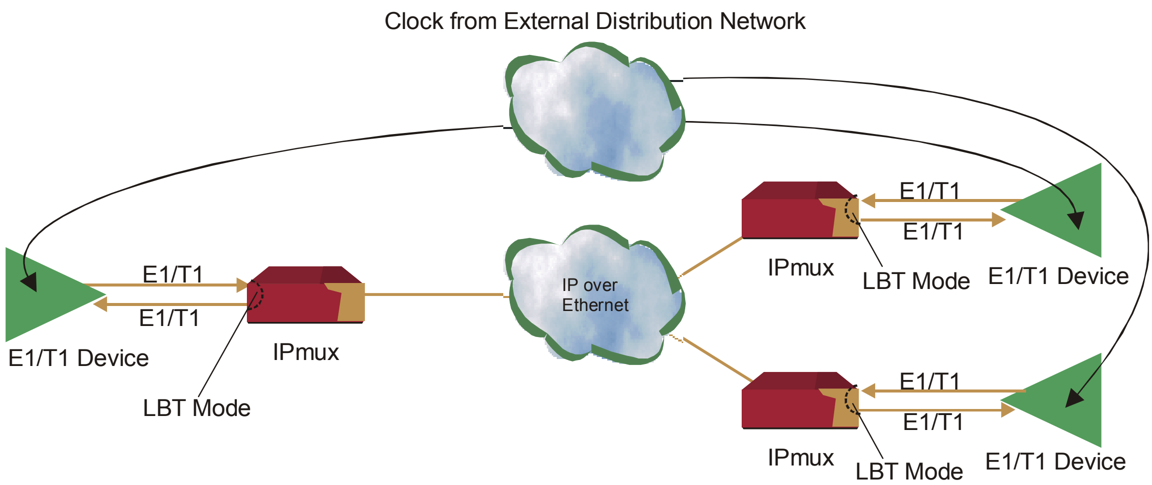
Packets in IP networks reach their destination with delay that has a random component, known as jitter. When emulating TDM on an IP network, it is possible to overcome this randomness by using a buffer to smooth out all incoming data, assuming the proper time reference is available. For the most part, however, the original time reference information is no longer available.

In principle there are two different levels of integration of TDMoIP into the telephone network. In the *toll bypass* scenario, a competitive carrier introduces an alternative link, based on TDMoIP, between two central offices. Due to the cost advantages discussed above, "toll bypass" services can be offered to customers at a lower rate than the incumbent's rate. In such applications, both TDMoIP devices should receive a time reference from the central offices to which they connect.

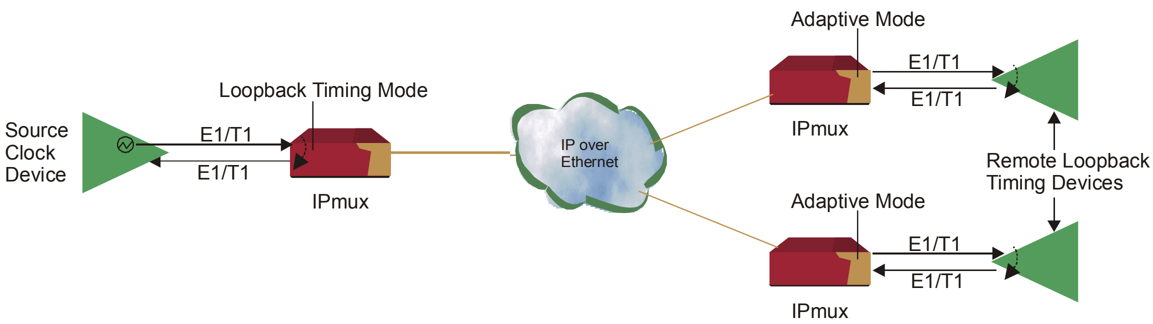
In the *whole network* scenario, major portions of the primary infrastructure are replaced with TDMoIP networks, and a method of time synchronization is required. IP networks also disseminate clock information through a protocol known as NTP; but unless the IP network is completely private and dedicated to the TDMoIP link, there will be no connection between the NTP clock and the desired TDM one. One solution is to provide time standards, such as atomic clocks or GPS receivers, to all TDMoIP devices, thus relieving the IP network of the need to send synchronization information. When the provision of accurate local time references is not possible or too costly, then recovery is possible when the destination regenerates and synchronizes the clock.

Timing Schemes

1) External master clock source provides out-of-band timing



2) In-band clock recovery and regeneration



Competing and Complementary Technologies

TDMoIP and VoIP

TDMoIP is simpler than VoIP because it is transparent to voice and data signaling and protocols, even when they are proprietary. VoIP, on the other hand, still has issues with new protocols and translation between signaling formats. VoIP does hold the promise of new applications, whereas TDMoIP automatically takes advantage of existing PBX and CTI features. As far as bandwidth optimization, VoIP gateways include DSPs which allow voice compression and silence suppression and therefore reduce bandwidth requirements. However, this is at the expense of lower quality and higher latency. The simplicity of TDMoIP translates into lower cost of ownership with initial expenditure savings and operational cost benefits. These savings are significant for the enterprise consumer who is typically not interested in replacing his legacy TDM equipment and dealing with “forklift upgrades” with the associated training and maintenance expenses. Another important difference is that due to transparency, TDMoIP can offer mixed voice-data services.

From a service provider point of view, TDMoIP and VoIP complement each another. Extending TDM trunks over the IP network transparently from the customer site to the carrier POP makes it simple for the carrier to deploy larger, scalable VoIP gateways and softswitches at the POP where resources are available, and provide the user with a simple TDMoIP Network Termination Unit (NTU) at the customer premises. These TDMoIP circuits could then be used to provide a number of services in addition to VoIP, such as regular PSTN access, Centrex, Frame Relay and ISDN.

TDMoIP and ATM

TDMoIP provides many of the benefits of ATM in that end-to-end delay starts at less than 2 ms and integrity of structured or unstructured T1s or E1s is maintained. TDMoIP is much simpler, cheaper and more efficient than ATM. Most importantly, it can be carried over IP and Ethernet networks. TDMoIP is more efficient than ATM because its payload size and therefore percentage of overhead can be set on a per-application basis. This is possible because the number of octets per frame is configurable. With ATM, payload is always 48 bytes, and therefore the percentage of overhead is much higher.

TDMoIP and Gigabit Ethernet

Gigabit Ethernet (and 10-Gigabit Ethernet) are being used in Metropolitan Area Networks (MANs) and Wide Area Networks (WANs). In particular, Gigabit Ethernet over dark fiber is becoming a popular alternative to SONET and ATM. However, Ethernet is basically a data network technology and cannot by itself handle voice traffic. TDMoIP empowers Gigabit Ethernet with voice and circuit extension capabilities and therefore is an excellent complementary technology. Together these technologies are expected to increase market share by offering a simpler, less expensive alternative to VoIP, ATM and SONET. History shows that the simpler, cheaper technologies such as Ethernet and Frame Relay tend to prevail over the more complex and expensive alternatives such as FDDI, ATM and Token Ring, even if the latter are more robust.

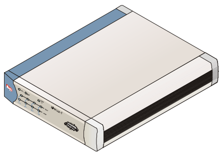
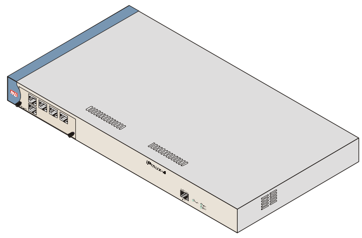
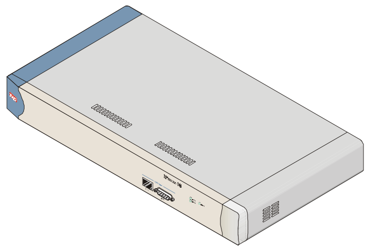
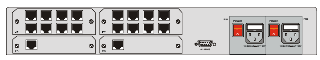
Initial concerns regarding simpler technologies like Gigabit Ethernet are usually rapidly put to rest by subsequent enhancements. For example, a SONET ring topology is considered very reliable because of its rapid recovery from a failure or fiber cut. Gigabit Ethernet does not inherently have this capability, but an alternate trunk can usually be switched within a few milliseconds. Even if there is only a single fiber between the switches, protocols such as OSPF enable routing tables to be updated within a few seconds, and the IP data stream to be quickly reconnected.

Another important example relates to QoS, where ATM has the most defined service level categories. However, today's Gigabit Ethernet switches and Terabit routers implement advanced mechanisms to prioritize packets and reserve bandwidth for specific applications. By marking TDMoIP packets (using 802.1p&q, ToS, and set UDP port numbers) they may be easily identified and prioritized.

Combining TDMoIP with Gigabit Ethernet switches and Terabit routers ensures a worthy alternative to SONET and ATM, providing a simpler, more cost-effective solution with more bandwidth and more efficient grooming of T1s or E1s to Gigabit Optical Networks.

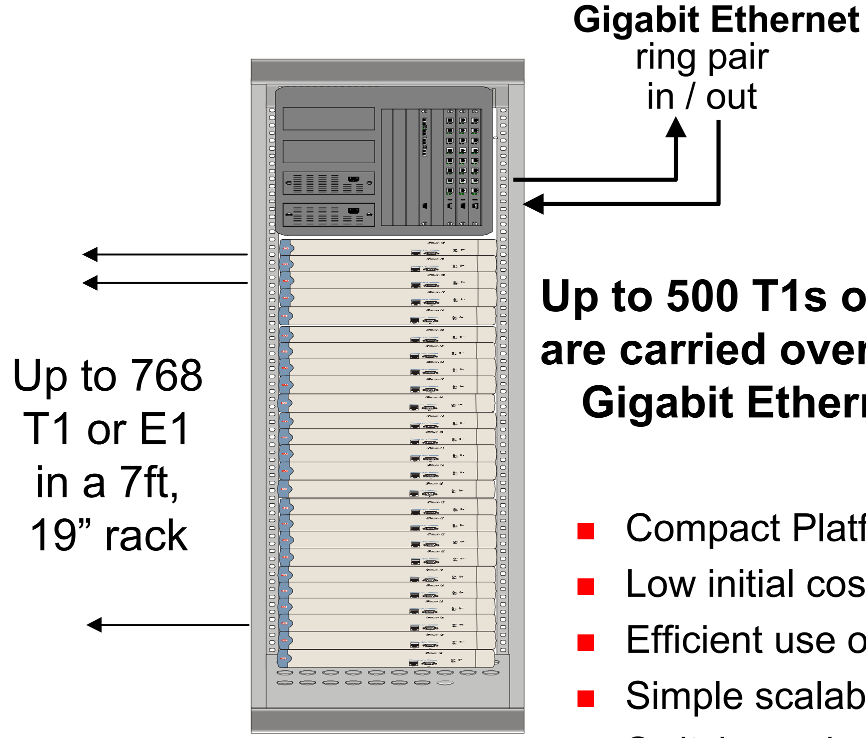
RAD's Implementation of TDMoIP

RAD implements TDMoIP technology in its IPmux product family. The family comes in three form factors: the small, low-cost IPmux-1 CPE device includes a T1/E1 TDM port and a 10/100BaseT data user port. The uplink can be either 10/100BaseT or 10/100BaseFx to eliminate the need for a local switch. There is also an external TB/4W option to extend the link to 1,250 feet over UTP. The IPmux-4 includes four E1/T1 ports and a 10/100BaseT uplink. The IPmux-16 provides a highly scalable solution supporting up to 16 T1s or E1s in a 1.5U high enclosure for up to 768 T1 or E1 interfaces in a standard 7 foot, 19" rack.

IPmux-1	IPmux-4	IPmux-16
Single T1/E1 or 4 BRI	1 or 4 T1/E1 ports	4, 8, 12 or 16 T1/E1 ports
User + Network 10/100BaseT ports	10/100BaseT network port	2 10/100 BaseT ports for network redundancy
Sub-T1/E1 support	Multi-bundle support	Multi-bundle support
Optional redundant power supply with AC/DC support	Optional redundant power supply with AC/DC support	Redundant power supply with hot-swap capability and AC/DC support
Optional 10/100BaseFx network port		
External clock port		External clock port
SNMP management	SNMP management	SNMP management

Carrier Class Scalability



Up to 500 T1s or 400 E1s are carried over a single Gigabit Ethernet link

- Compact Platform
- Low initial cost
- Efficient use of rack space
- Simple scalability
- Switch vendor independent

<u>IPmux-16 stacked in a Central Office</u>

Carrier Market

Local Exchange Carriers (LECs)

At the local level, one finds three types of service providers offering connectivity services for voice and data. These include incumbent local exchange carriers (ILECs) which formerly had a monopoly of local services; the state-certified competitive local exchange carriers (CLECs), which compete either by building their own facilities or by leasing “unbundled” facilities for resale from the ILEC; and the ISPs that provide Internet access as well. Sometimes the ISP is a subsidiary of the ILEC, but state regulation means that these ISPs must be customers of the ILEC, just like everyone else. This arrangement retains the split between basic transport and enhanced services. In addition to providing pure Internet access, ISPs are also offering web and application hosting for the business user. Enhanced services such as unified messaging and VoIP are being targeted at the residential and small business customer. VoIP is also being used to offer savings on international calls, and once QoS is assured we can expect to see more businesses adopt this technology. In fact, CLECs are becoming ISPs by offering data services and ISPs are becoming CLECs by offering IP telephony.

Network unbundling, the process of breaking the network into separate functional elements, opens the local access area to competition. CLECs select unbundled components they need to provide their own service. If the unbundled price is still too expensive, the service provider will provide its own private resources or turn to an alternative access vendor (AAV), which can offer private line service between an entity and facilities at another location. Innovative providers will take advantage of whatever technology is available to attack the competition, including wireless microwave systems, leased dark fiber, infrared and laser based wireless and CATV.

Next Generation carriers employ wireless for its easy deployment, and fiber for its bandwidth. Additional new breeds of CLEC have emerged,such as the data CLEC or DLEC that deliver high-speed Internet connectivity and other data offerings, or the building CLEC or BLEC that wire commercial buildings with fiber-optic cable to provide Shared Tenant Service (STS).

Despite the technological and structural advances, the new carriers must still tackle the problem of grooming T1 circuits onto optical and wireless networks.

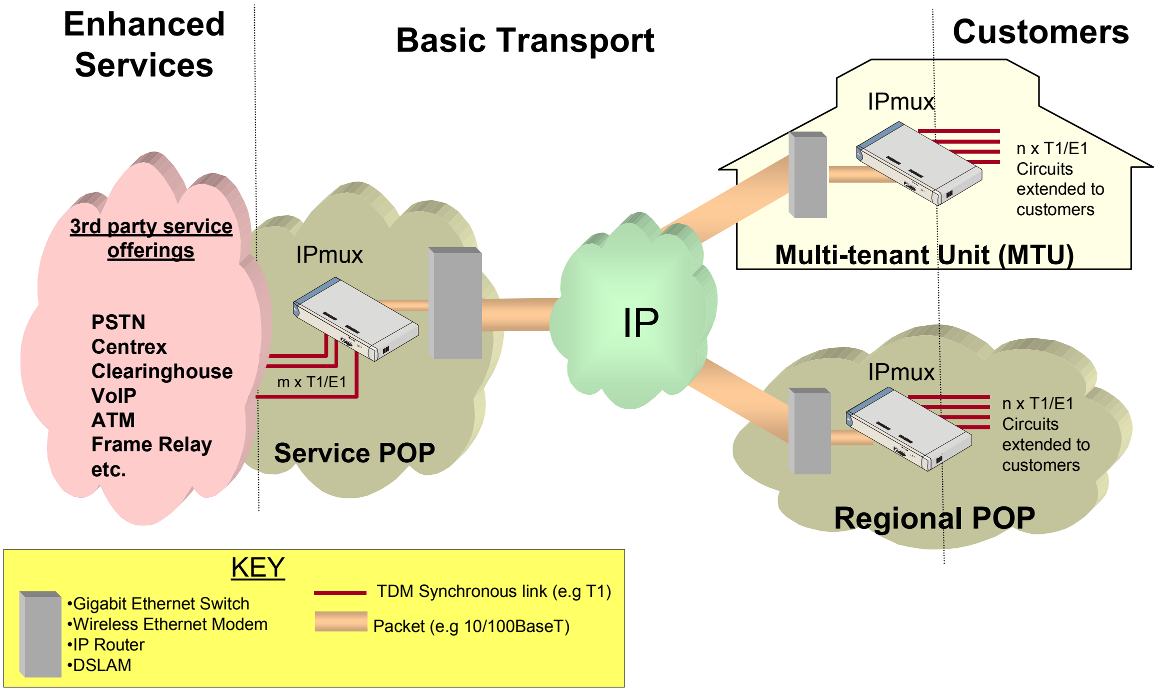
Interexchange Carriers (IXCs)

Interexchange carriers provide telecommunications service between local calling areas. The IXCs have to pay access charges to the ILECs and therefore have an interest in being offered competitive basic transport services from facilities based CLECs with alternative access solutions. IXCs with spare capacity can take advantage of this available bandwidth for circuit extension services across their own packet networks. Circuit Extension over IP benefits the IXC because it offers more local access alternatives and more options for providing voice trunking solutions to their customers across their own IP networks.

Applications

Carrier Voice Trunking and Circuit Extension over IP

Competitive local exchange carriers (CLECs) can take advantage of TDMoIP technology to expand their market share quickly, simply and inexpensively with more points of presence (POPs) and more services. Incumbent local exchange carriers (ILECs) providing basic transport of circuits wholesale to the CLEC can use TDMoIP to extend the T1 or E1 circuits cost-effectively over optical or wireless links back to the CLEC Service POP.



Building, Data or other Competitive Local Exchange Carriers or utility companies can increase their revenues by taking advantage of their wireless or optical data networks and use TDMoIP to extend T1 or E1 circuits between an entity and facilities at another location. In this way, the service provider can bundle private line services in addition to broadband data. Bundled services could include:

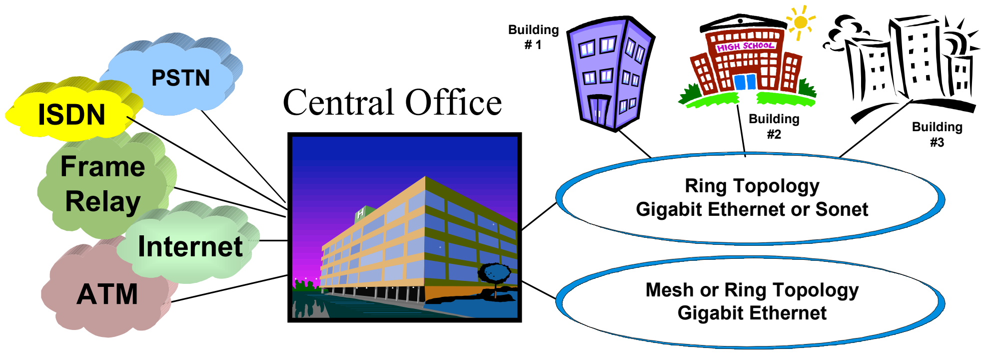
- High speed Internet access, Web hosting and VLAN
- Traditional PSTN access, or Centrex service, by extending circuits from the class 4/5 switch at the CO to the customer site
- New voice and unified messaging services using VoIP gateways or switches and extending circuits to the remote site
- Private line services
- Local Loop access to ATM, Frame Relay, ISDN and X.25 networks

The IPmux offers unprecedented flexibility to carriers looking for low cost extension of T1 or E1 services over an IP or Gigabit Ethernet network. Some emerging applications are shown on the following pages.

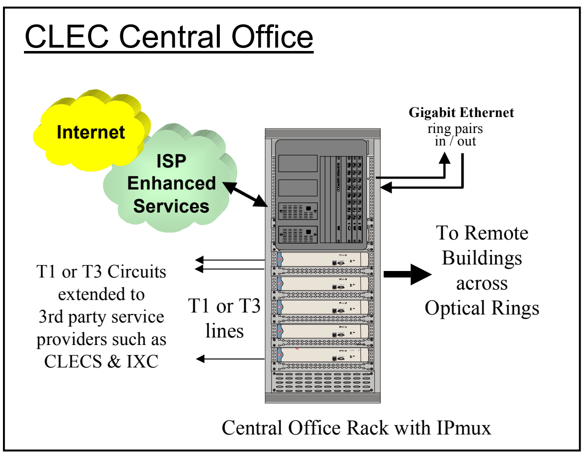
T1/E1 Circuit Extension over Packet Networks as a Local Loop Replacement

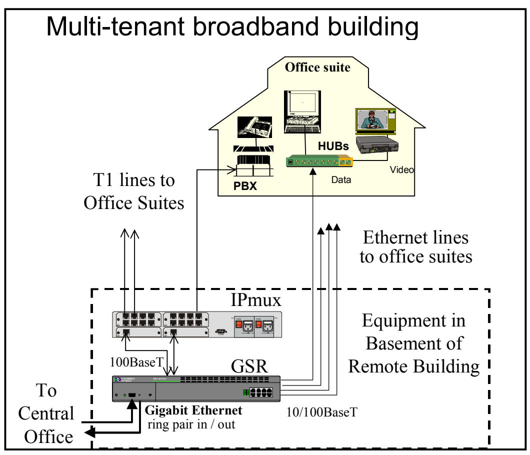
TDMoIP offers the CLEC an alternative method of extending services to their customers. Traditionally it was necessary to lease the Local Loop from the incumbent local exchange carrier (ILEC), but now any packet network will do. All voice and data services that traditionally run on T1 or E1 circuits are automatically supported over IP. This not only includes simple PSTN access, but also PRI, Centrex, and VoIP for voice services and ATM, Frame Relay, PPP, ISDN, SNA, X.25 for data services. The packet networks are also no longer limited to synchronous ATM and SONET, but can run equally well on 10/100BaseT wireless radios, Gigabit Ethernet, or any other network.

In this application, T1/E1 circuit extension is made possible by TDMoIP technology. The CLEC now has an alternative to leasing “unbundled” facilities from the ILEC.



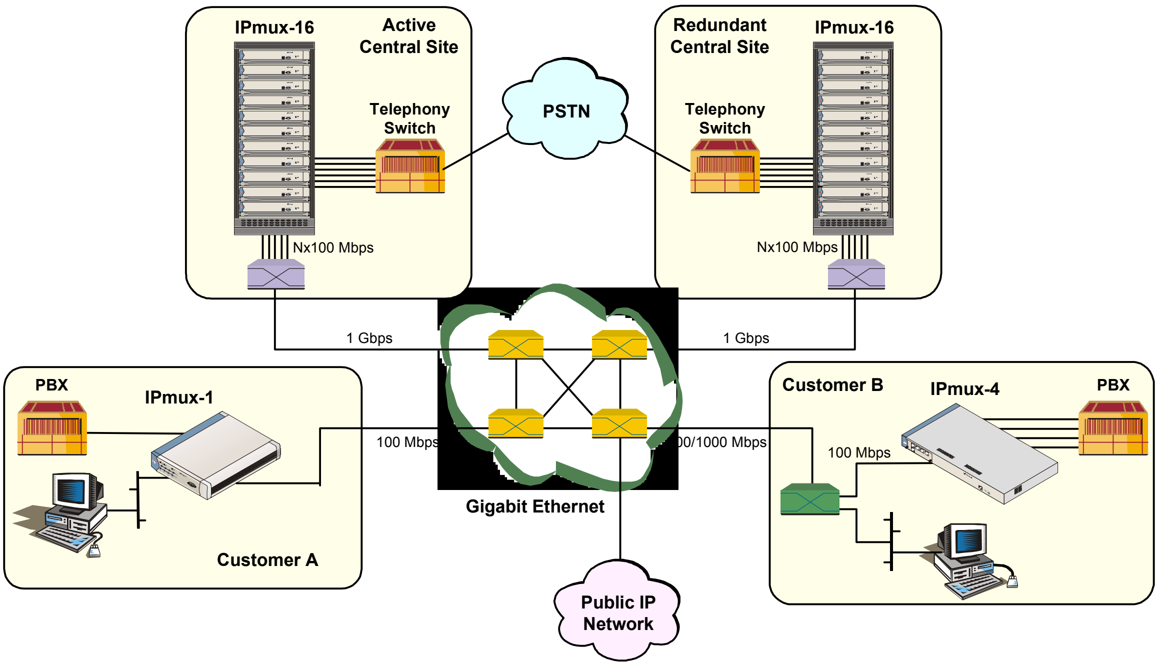
Multi-tenant broadband buildings are wired to offer high-speed data to each suite over fiber or copper lines. In each building, the CLEC has a small POP (typically in the basement). This POP is then connected to the CLEC Central Office using an optical network made up of Gigabit Ethernet, SONET, or even FDDI (this broadband connection could also be wireless or laser). Ethernet or IP is used to provide high-speed data. TDMoIP is then used to extend traditional T1 or E1 circuits from the central office to each building over this network. In this way, the tenants can be offered a variety of services either directly from this CLEC, or from third-party providers contracted by the CLEC. This architecture is also relevant for the Building or Data CLEC that simply wants to provide basic transport service to other CLECs or IXCs and lets them offer bundled services over these leased lines. T3 aggregation shown below (left-hand figure) requires the RAD DXC-30 option.





PSTN Access over Fiber IP-based Networks with Redundancy

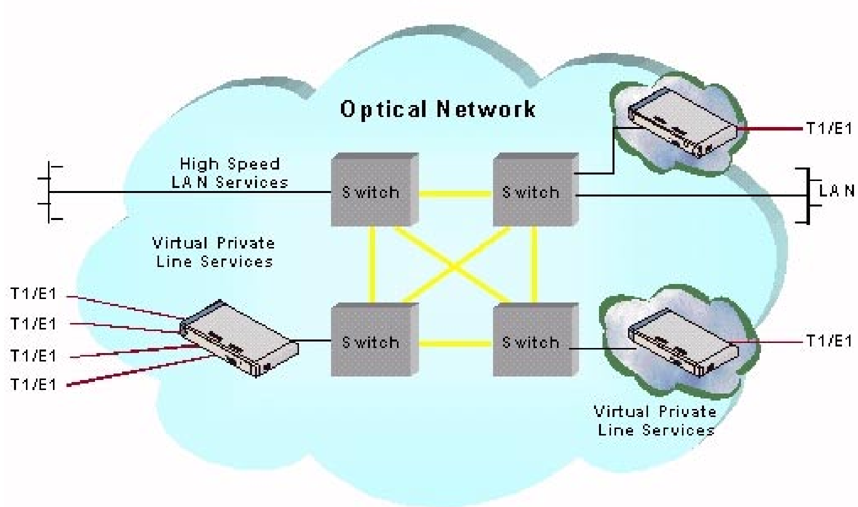
Telephony services such as PSTN access, Centrex and VoIP can simply and inexpensively be provided to customers over a metropolitan area network (MAN). Alternate routing and switching capabilities found in Gigabit Ethernet switches and Terabit routers can be used to reroute traffic quickly in the event of network failure. Private point-to-point leased line services between customers sites can also be offered over the same fiber IP-based network.



Traditional T1/E1 Leased Line Services over IP network

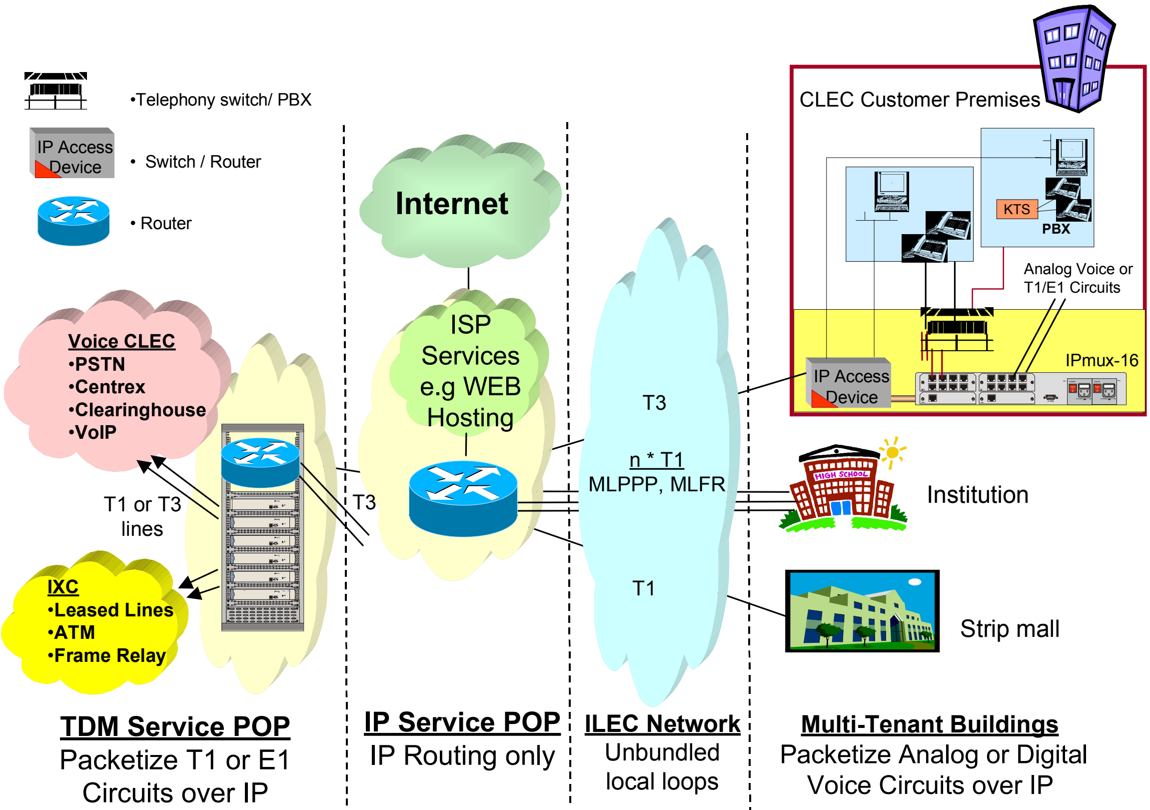
Carriers with a high speed IP infrastructure can take advantage of under-utilized bandwidth and offer virtual T1/E1 leased lines. If packet delay and loss through the network is low, then this service is just like any other T1/E1 leased line. In this way, carriers can also offer E1 services in the US and T1 circuits internationally.

Below, the customer is provided with point-to-point T1 or E1 leased lines.



Grooming of T1, E1 and Fractional Services over Routed IP Networks

In this application, a pure IP network architecture can be used not only for data and management, but also for voice and circuit extension over IP. The ability of the IPmux to groom DS0s and extend T1 or E1 circuits efficiently and transparently over IP networks provides service providers with a simple, low cost alternative to the traditional digital cross connect (DACS). In this way, the service provider can use the efficiency and flexibility of IP routing to extend its network into tier 4 towns and multi-tenant broadband buildings without compromising integrity and quality.



The network can be divided into four segments starting at the multi-tenant building where the customers can be offered a variety of services, which when bundled together provide the convenience of a single bill and point of contact. Bundled services do not need to originate from the same service POP. In the above diagram, an IP service POP providing WEB hosting and Internet access, and a TDM service POP acts as the gateway to 3rd party service providers such as voice CLECs and IXCs. Each network segment is explained below in more detail.

Multi-Tenant Buildings

In order to increase revenue and market share, many building CLECs are looking to bundle voice services over the same access network used today for VPN and Internet services. These services must be provided in a cost-effective, easily implemented manner and without compromising traditional circuit quality. VoIP gateways are an alternative, however they are not 100% transparent, and therefore compromise the level of service with limited support for V.90 modems, faxes, and standard voice features such as 911, *69 and caller ID. This alternative also adds considerable delay, complexity and cost that will make the business customer more reluctant to switch service.

A simpler alternative is to provide PSTN quality service with the same analog voice and digital lines, but now use TDMoIP technology with the IPmux to provide a scalable and efficient grooming platform. Voice and leased lines from the IPmux can be combined with packet data and routed across the network. A telephony switch / PBX could be used to provides local switching and statistically consolidate many voice lines to fewer ports on the IPmux gateway to reduce bandwidth requirements. The IPmux solution is scalable because it is now possible to use IP to grow the voice and leased line services from a single DS0 to many T1 or E1 circuits without needing a digital cross connect (DACS). In addition, increasing bandwidth to the building is simple and inexpensive because just the link interface of the IP access device (switch/router) needs to be matched to that of the line coming from the network. Everything behind the Access Device stays the same because it is all IP.

ILEC network

In most cases today, the building CLEC still depends on the unbundled local loops from the incumbent local exchange carriers (ILEC) to connect his multi-tenant buildings to his service POPs. T1 circuits can be added and MLPPP or MLFR can be used to bond the circuits together to provide a high-speed IP link. T3 and OC3 provide even higher speed options. If a wireless, laser or optical alternative becomes available then the CLEC can easily take advantage of the fact that IP works with any link layer protocol including Ethernet and SONET. Therefore building a pure IP access network makes this transition simpler.

IP Service POP

This service POP only needs to route IP packets. Packets containing voice and leased line services can simply be forwarded to the TDM Service POP. Similarly packets requiring ISP value-added services, such as WEB hosting can be forwarded to the relevant servers. In addition, simple Internet access can be provided from this POP.

Private leased line services can also be offered between buildings by tunneling DS0 bundles across this CLECs IP network. At each of the buildings, an IPmux is responsible for packetizing the TDM bit streams, which are then prioritized and routed between buildings.

TDM Service POP

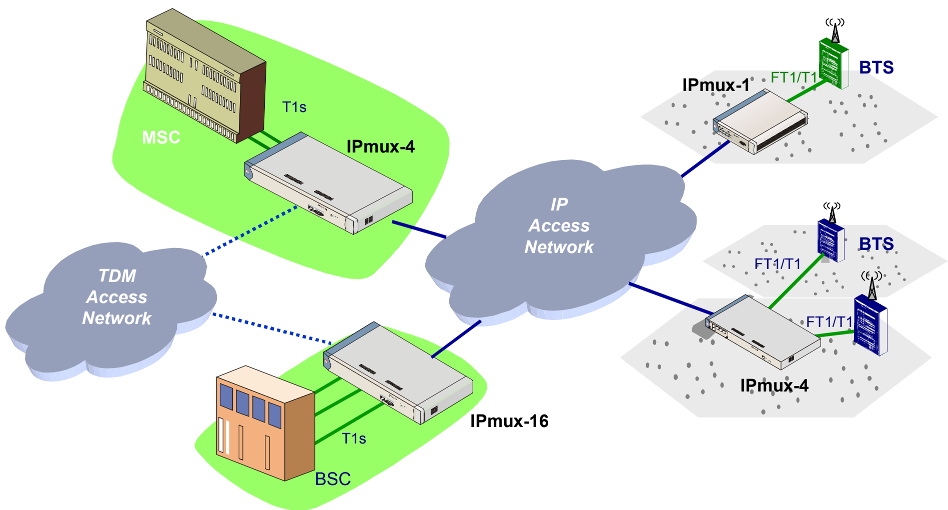
IP packet streams carrying voice, ATM, Frame Relay and other tunneled protocols that are not being used for private line services are routed to the TDM Service POP where the circuits are terminated. Using the IPmux-16, it is possible to terminate 768 T1 or E1 circuits in a single 19", 7ft rack. At this POP, the CLEC may decide to offer voice services by connecting the T1 or E1 circuits to his own class 4/5 switch or VoIP gateway, or deliver the circuits directly to 3rd party providers such as Voice CLECs and IXCs.

The Voice CLEC would receive voice channels in DS0 bundles and provides voice services ranging from regular PSTN access and Centrex to clearinghouses and VoIP for lowest cost international telephony services. The IXC (or Data CLEC) could provide traditional data services such as ATM and Frame Relay over these full or fractional T1 or E1 circuits. For example, assume a large corporation uses Sprint's ATM network for connecting head office to all branches, and now wants to open an office in one of the CLEC multi-tenant buildings. It would now be possible for the CLEC to extend a T1 circuit from the IXC at the TDM service POP to the office suite as a leased line, and the branch office will get ATM service just like any other branch.

Cellular Carrier's IP Backbone

Cellular carriers networks are based on TDM technologies. Connectivity between the Base Stations (BTSs), Base Station Controllers (BSCs), and the Mobile Switching Center (MSC) is achieved using TDM microwave links and T1/E1 leased lines. Until recently, ATM was the most logical alternative. Because of pressure from environmental groups, cellular carriers are looking for an alternative to microwave, and leased lines are costly. With the introduction of QoS in Gigabit Ethernet networks and the availability of TDMoIP, IP is very seriously considered as the preferred solution.

Third-generation cellular (UMTS) vendors are in advanced stages of implementing TDMoIP solutions where IP networks are the cellular backbone. In the future, current and third-generation cellular devices will be co-located to access on an IP-based backbone.



Conclusion

Circuit extension over IP using TDMoIP technology provides the benefits of IP networks without the dangers of a new technology. It supports legacy telephony switches or PBXs over data infrastructures with no loss of functionality. In addition, it empowers IP networks with leased line services for voice, ATM, Frame Relay, SNA, and so on. Finally, the fact that it's simple, inexpensive and transparent to all signaling and protocols makes it an ideal complement to Gigabit Ethernet in Next Generation Networks.