TDMoIP vs. VoIP

Matching Technology to Requirements

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RAD Data Communications

TDMoIP

Tying the Circuit Switched World to the IP Network

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Abstract

Why migrate to IP? – Connecting traditional voice, video and data over IP/Ethernet/MPLS networks has become an attractive alternative to running parallel voice and data networks. It saves money on call and leased-line service charges, while consolidating management, cutting maintenance costs, and increasing user productivity. This is achieved by converging two important traffic types onto one infrastructure and takes advantage of the simplicity and efficiency of IP routing and Ethernet switching.

Obstacles – Since Voice over IP (VoIP) was first introduced in the mid-90s, many industry analysts, service providers and vendors expected it would be the future technology of choice for both carrier and enterprise telephony services. Nevertheless, multiple standards and technological immaturity, compounded with the recent economic downturn, have considerably slowed VoIP adoption rate.

Benefits – The merits of VoIP based services, compared with traditional TDM and Leased-Line based solutions lie in the compression and packetization of the voice traffic (enabling cost-effective transmission), as well as in the switched architecture that further optimizes switch resources and allows for effective service oversubscription. Voice switching is particularly important in environments where the existing PBX/switch facilities are inadequate. It is of lesser importance in typical trunking applications with denser traffic, where the added value of switching can be outweighed by the simplicity of a non-switched solution.

Price – The benefits of VoIP deployment come at a price. This is in the form of forklift upgrades of core and edge infrastructures offered by the newer VoIP insurgents or adaptation of existing equipment by incumbent legacy switch vendors who are promoting a less radical migration path to VoIP.

Future – While VoIP is constantly growing, carrier conservatism and a tremendous installed base of feature-rich legacy PBXs in the enterprise environment will cap VoIP growth rate in the short term until either economics change, technology issues are resolved and/or existing equipment is depreciated. This creates an opportunity for a more **evolutionary** solution that will benefit from the merits of both worlds: the quality of service and rich feature set of yesterday’s technology, with the cost effectiveness of tomorrow’s backbones.

TDMoIP (Time Division Multiplexing over the Internet Protocol) is a transport technology developed by RAD Data Communications that extends T1, E1, T3 or E3 voice or data circuits across packet-switched networks simply, transparently and economically.

TDMoIP vs. VoIP – In this paper we compare TDMoIP with VoIP. We see how TDMoIP multiplexing technology enables better bandwidth utilization (up to 60% less generated traffic compared with VoIP) and how its transparency to standard and proprietary signaling protocols eliminates the need for forklift upgrades and PBX fine-tuning. We describe how TDMoIP works and show 12 sample applications, such as trunking, cellular backhaul and enterprise PBX connectivity over IP.

RAD manufactures both VoIP and TDMoIP products.

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Contents

1. Introduction

Scope

This paper compares VoIP and TDMoIP, highlighting the operational benefits of each and comparing and contrasting their respective strengths and weaknesses. It also addresses some of the many factors that go into a procurement decision, such as the application, traffic types, bandwidth requirements, scalability requirements, investment protection, and budget.

Obstacles

Since VoIP was first introduced in the mid-90s, many industry analysts, service providers and vendors have made the logical connection between the need to take advantage of the efficiency, flexibility and ubiquity of IP, and the large market for voice services. This has led to a huge investment in VoIP and an alphabet soup of protocols such as H.323, SIP, MGCP, Megaco, and so on. Despite these efforts to promote the revolutionary VoIP concept, the reality has been that enterprise users have been reluctant to give up their legacy PBXs, many of which have only recently been upgraded to comply with Y2K. Enterprise users have not only been concerned with the capital expenditures for "forklift" upgrades and the operating expenses associated with maintaining a complex VoIP network, but also with the business issues such as voice quality, loss of existing PBX features, and retraining users.

Complementary Alternatives

TDMoIP offers a simple, evolutionary alternative to VoIP that addresses many enterprise concerns by supporting new and legacy PBX signaling protocols transparently over IP, Ethernet and MPLS.

TDMoIP and VoIP are **complementary** technologies.
VoIP may be appropriate in new applications that require features that are not provided by existing PBXs and telephony servers/switches. In such cases, the design, installation and maintenance must be carefully planned to overcome the obstacles of working with new, complex and evolving protocols.
TDMoIP, on the other hand, is the preferred solution for customers who are satisfied with their current voice, video and data services, but want to reduce expenses by running all traffic types over a single IP, Ethernet or MPLS network. Their goal is to take advantage of efficient packet-based infrastructures without incurring the costs and inconvenience of expensive upgrades and lost features. Therefore, TDMoIP retains the investment in existing equipment and provides full legacy functionality over IP, while VoIP is preferable when the goal is to reengineer the network and add fanciful new features such as phones with browsers.

RAD Data Communications

RAD Data Communications has more than 10 years experience in voice processing and voice compression. With a wide range of voice-over-packet solutions, including a VoIP Gateway/Gatekeeper, TDMoIP™ Gateways, and Voice Trunking Gateways with Ethernet or T1/E1 network interfaces, RAD has achieved broad recognition as a major provider of voice and data networking solutions.

2. TDMoIP

TDMoIP Concept

TDMoIP (Time Division Multiplexing over the Internet Protocol) is a transport technology developed by RAD Data Communications that extends T1, E1, T3 or E3 circuits simply across packet-switched networks. It is transparent to all protocols and signaling, and therefore supports legacy PBXs – including proprietary features. TDMoIP enables service providers to migrate to next-generation networks and continue to provide all their revenue-generating legacy voice and data services. It enables enterprises to run voice, data and video over their IP/Ethernet-based networks, minimizing network maintenance and operating costs. TDMoIP also benefits data carriers by enabling them to offer lucrative services – such as leased line, cellular backhaul and international toll bypass – over IP. This allows them to maximize revenues from their IP, Ethernet or MPLS infrastructure with revenue-generating voice and leased line services complementing their existing data services. There are two variants of TDMoIP:

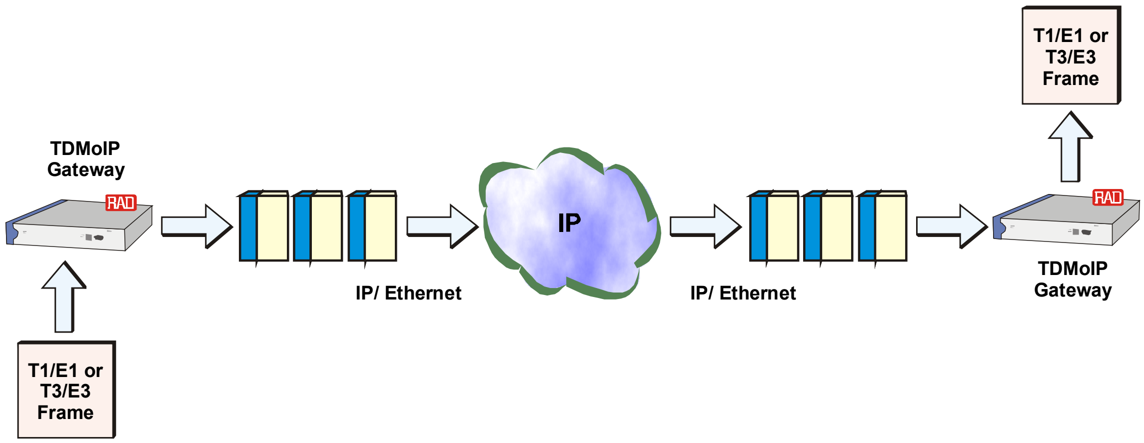
TDMoIP (CE)

TDMoIP (CE) provides T1/E1 and T3/E3 Circuit Emulation over IP/Ethernet/MPLS. This technology is ideal when requiring lowest latency, highest quality voice, video or data over IP. TDMoIP (CE) requires the network to transport a constant stream of high-priority packets with strict QoS (to ensure the TDM circuit is also error-free). TDMoIP (CE) is usually the best technology in environments where bandwidth is abundant (e.g. campus or MAN with Fast Ethernet/Gigabit Ethernet). With TDMoIP (CE), clock synchronization is maintained, making it possible to extend synchronous TDM circuits over asynchronous IP/Ethernet networks.

TDMoIP (CV)

TDMoIP (CV) is optimized for carrying Compressed Voice and Group 3 fax relay. This technology is ideal in environments where network bandwidth is limited, because it supports voice activity detection (VAD), silence suppression, comfort noise generation (CNG) and voice compression (down to 4 kbps/channel). TDMoIP (CV) also can be thought of as transparent VoIP because it has many of the same characteristics, but carries signaling transparently, uses less bandwidth and is more tolerant of packet loss. Some of these extra efficiencies result from the unique way that TDMoIP aggregates and multiplexes multiple voice and signaling channels into a single IP bundle, whereas VoIP handles each of these channels independently. Therefore, TDMoIP (CV) is ideal for applications where bandwidth is constrained (e.g. 802.11b wireless, cable modems, xDSL, Power Line Communications or even the public Internet).

TDMoIP Operation



TDMoIP (CE) works by chopping TDM data streams into IP packets and adding IP headers. Packets are then bridged or routed over the IP/Ethernet/MPLS network. At the destination site, the original bit stream is reconstructed by removing IP headers, concatenating packets, and regenerating the clock.

TDMoIP (CV) works by processing standard PCM voice and signaling channels into packets that are sent to the TDMoIP multiplexing unit, which aggregates a number of these packets into a frame. When the frame reaches its maximum size or the packetizing interval is reached, a TDMoIP header is added in order to complete the TDMoIP frame structure. Aggregation of packets from multiple channels into a single, configurable size frame makes TDMoIP more bandwidth-efficient than VoIP. TDMoIP (CV) is also more resilient to packet loss because each frame consists of distributed content from multiple channels, so the effect on individual channels is minimized. Packets are then bridged or routed over the IP/Ethernet/MPLS network. At the destination site, the original voice and signaling channels are reconstructed.

Transparent connectivity over the IP/Ethernet/MPLS network maintains all the features of the telephony network. TDMoIP thereby provides seamless migration of a variety of legacy services to packet-switched networks, with full support for legacy equipment such as Class 4 and 5 switches, PBXs, telephony switches and TDM multiplexers.

3. VoIP

VoIP Concept

Voice over IP (VoIP) is a voice switching and transport technology that works together with the Public Switched Telephone Network (PSTN) to deliver a phone call. Converting traditional telephony to IP has proven to be very difficult because of the many different types of signaling systems involved.

VoIP Disadvantages

VoIP differs from TDMoIP in that it must understand all the required signaling protocols and convert them to its own protocols (whereas TDMoIP simply carries the signaling transparently over IP). An additional limitation to the complex signaling conversion is that VoIP gateways typically support only a subset of PBX features.

VoIP Advantages

The advantage of VoIP is that dissimilar voice signaling protocols on either end of the IP network can interoperate when they are both converted to a common VoIP signaling protocol, such as H.323.

VoIP Phone Call over the PSTN – Example

The following example shows how the existing PSTN network control systems must communicate with the data network control systems:

Assume a call is going from a VoIP phone to an analog phone in a different city:

- The VoIP phone dials the destination phone number.
- The local data network only knows about its own local IP addresses and forwards it to the access network, which in this case is a city-wide VoIP network.
- Not knowing where that 10-digit phone number is located, the city IP network needs to locate the destination city and then find a data network (if it exists) to get to that particular city.

In the destination city:

- The data network needs to query the local PSTN analog system and converts the Connect IP messages to the proper signaling messages for that type of voice switch.
- The voice switch then checks to see if the phone is busy and, if it is, sends a message back to the IP network.
- Eventually, some piece of equipment needs to generate a busy signal waveform to send to the VoIP phone.

Converting between IP addresses and 10 digit phone numbers is not a trivial process and involves many steps.

In this example, the call may be a long distance call subject to certain rate charges, or this may be just a local call. Knowing the underlying regulatory structure is necessary to provide proper billing. The IP addresses for the devices may not even be public. In many cases, the temporary IP addresses assigned to devices are useable only within the company.

In the presence of these private addresses or company firewalls, how does an outsider reach a VoIP phone within a company? What happens when the temporary IP for the VoIP phone changes?

4. TDMoIP vs. VoIP

Highlights

	TDMoIP (CE)	TDMoIP (CV)	VoIP
Standards – Toll-quality compressed voice (G.711, G.729a, G.723.1)	G.711	All	All
Latency – minimum end-to-end	3 ms	45 ms	45 ms
Bandwidth – minimum per voice channel	74 kbps	3.8kbps*	10.4 kbps*
Transparency – to standard & proprietary signaling protocols	Yes	Yes	No
Group III fax relay support at 4.8 – 14.4 kbps	> 28.8 kbps	Yes	Yes
Investment protection in legacy PBX during migration to IP	Yes	Yes	No
Maintains features for all existing PBX (no staff retraining)	Yes	Yes	No
Circuit extension over IP/Ethernet/MPLS for T1/E1 & T3/E3	Yes	No	No

*Average bandwidth per voice channel with equivalent MAC, IP and UDP overhead (based on system carrying 60 channels of voice with G.723.1 coder, 5.3 kbps compression, 30 ms packetization interval, and 50% silence suppression).

Key Differences

- All three technologies support toll quality voice and fax, but only TDMoIP (CV) and VoIP include DSPs to compress voice and reduce bandwidth.
- The disadvantage of this additional processing is greater end-to-end latency and loss of synchronous T1/E1 and T3/E3 circuit extension capability.
- Both TDMoIP variants are transparent to standard and proprietary signaling protocols to protect investment in legacy PBXs when migrating to IP infrastructure (without compromising existing PBX features).

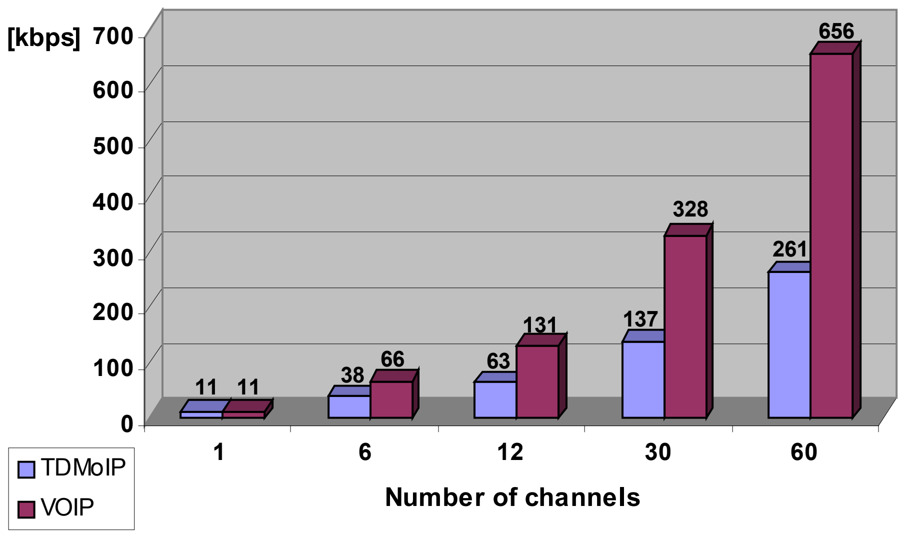
Framing

With equivalent MAC, IP and UDP overhead, TDMoIP and VoIP differ only in the inner structure of the frame.

The VoIP frame uses an 18-byte RTP header, compared with the 11-byte TDMoIP frame, which contains specific AAL2 and voice-type headers.

At equivalent G.723.1 6.4 kbps compression, VoIP and TDMoIP both feature a payload of 24 bytes.

The resulting difference is hardly significant when discussing single channels. But when channel bundling is considered, VoIP becomes increasingly inefficient compared with TDMoIP, as can be seen in the following graph for G.723.1 (compression 6.4 kbps, 50% silence suppression).



RTP multiplexing, which is becoming available on some solutions, would reduce the gap to approximately 15%, but such deployment offerings are not yet available.

TDMoIP Advantages

TDMoIP uses considerably less bandwidth than an equivalent VoIP based solution. TDMoIP multiplexing allows for up to 60% better bandwidth utilization compared with VoIP solutions.

Voice Considerations

Voice compression, VAD and CNG

As shown in the table below, all three technologies support standard G.711 PCM voice, but only TDMoIP (CV) and VoIP offer voice compression. TDMoIP (CV) and VoIP also provide voice activity detection (VAD), silence suppression and comfort noise generation (CNG) to decrease the bandwidth during periods of silence without disrupting the conversation. The CNG is intended to give the listener the feeling that the call is still connected (as opposed to producing absolute silence, which gives the impression the call has been dropped).

Latency

TDMoIP (CE) requires at least 74 kbps of bandwidth per voice channel, but its end-to-end latency, starting at just 3 ms, is significantly lower than TDMoIP (CV) and VoIP (starting at 45 ms). This lower latency usually makes echo cancellation unnecessary and is important in delay-sensitive applications (e.g. cellular backhaul, SCADA, polling protocols, etc.). FAX, DTMF/MFR/MFC signal detection, generation and relay are supported by all three technologies, transparently in the case of TDMoIP (CE).

Migration

Installation and maintenance are much simpler when deploying TDMoIP (CE and CV) because both implementations are transparent to all standard and proprietary CAS and CCS signaling protocols such as robbed-bit signaling, ISDN PRI, R2, E&M, Q.Sig, DPNSS, and SS7. With VoIP it is necessary to understand what the signaling protocols are, to confirm that the VoIP gateway supports them, and to then perform signaling conversion. TDMoIP (CE and CV) provide transparent support for all PBX features, but with VoIP these features may be compromised by VoIP gateway capabilities. New voice features not supported by existing PBXs are possible using VoIP, such as support for IP phones.

Super tandem, jitter and high quality voice

TDMoIP (CV) has a unique “super tandem” feature that maintains voice quality through multiple PBX/switch hops by synchronizing compressed voice at low frequencies. This feature ensures that voice is compressed and decompressed only once, at the terminating TDMoIP (CV) gateways, avoiding voice degradation or delay when calls are routed through several TDMoIP (CV) gateways. In addition, each voice channel has a dedicated jitter buffer that changes dynamically according to network delay variation. High quality voice is also maintained when there is up to a 5% packet loss. (See “Speech Quality Scores”, page 32, under various jitter, coder and packet loss conditions.)

Voice	TDMoIP (CE)	TDMoIP (CV)	VoIP
Compressed voice with toll quality (G.729a, G.723.1)	No	Yes	Yes
Uncompressed toll quality voice (G.711)	Yes	Yes	Yes
Low-latency, uncompressed, toll quality voice (G.711)	Yes	Limited	Limited
Minimum end-to-end latency (through two gateways without network delay)	3 ms	45 ms	45 ms
Minimum bandwidth required per voice channel	74 kbps	3.8 kbps*	10.4 kbps*
Bandwidth On Demand (BOD)	Limited	Yes	Yes
VAD (voice/silence activity detection) & CNG (comfort noise generation)	No	Yes	Yes
Fax & modem support (transparent - voice band data)	Yes	Limited	Limited
Group III fax relay support at rates of 4.8 to 14.4kbps	Transparent	Yes	Yes
Echo cancellation (G.168)	Limited	Yes	Yes
Transparent to standard & proprietary signaling protocols	Yes**	Yes**	No
DTMF/MFR2/MFC signal detection, generation & relay	Limited	Yes	Yes
Requires conversion to complex voice signaling protocols such as H.323, SIP, Megaco and MGCP with PBX features limited to VoIP Gateway capabilities	No	No	Yes
New voice features possible (e.g. supports IP phones)	No	No	Yes
Single compression end-to-end (Super Tandem)	N/A	Yes	No

*Average bandwidth per voice channel with equivalent MAC, IP and UDP overhead (based on system carrying 60 channels of voice with G.723.1 coder, 5.3 kbps compression, 30 ms packetization interval, and 50% silence suppression).

**Supports all CAS & HDLC-based CCS protocols, including R2, E&M, ISDN, Q.SIG, DPNSS, SS7 and even proprietary PBX protocols.

Data and Video Protocols

In the table below, you will notice that TDMoIP (CE) is most suitable for supporting a variety of data and video protocols over IP. The reason is that circuit emulation was designed to provide CSU/DSU-like transparency – including timing – over IP/Ethernet/MPLS.

Data and Video Protocols	TDMoIP (CE)	TDMoIP (CV)	VoIP
HDLC / Frame Relay / X.25 / CDMA / SS7	Yes	Limited	No
HDLC channel compression	No	Yes	No
Async/sync protocols including ATM, H.320 Video, GSM, etc.	Yes	No	No
Maintains synchronization over asynchronous IP/Ethernet	Yes	No	No
Maintains integrity of framed and unframed T1/E1 circuits	Yes	No	No
Transparent to payload (including framing, protocols, encryption, etc.)	Yes	No	No

Network Requirements

As shown in the table below, TDMoIP (CE) has the most demanding network requirements and TDMoIP (CV) has the least. This means that TDMoIP (CE) should be avoided if the network cannot provide at least the required TDM bandwidth + 15% for overhead in a reliable consistent manner (i.e. with QoS – see White Paper "*Determining the Suitability of a Network for TDMoIP*" for more details on bandwidth, PPS and QoS requirements). The table also shows that TDMoIP traffic is simpler to prioritize through the network, because in addition to supporting the standard VLAN tagging and ToS/Diffserv packet prioritization capabilities, it also uses a single, assigned, IANA-registered UDP destination port to simplify flow classification through routers and switches.

Firewalls

Another important consideration – especially for the enterprise customer – is how to implement VoIP or TDMoIP through firewalls without compromising security. In this situation, TDMoIP has a significant advantage over VoIP. With TDMoIP, each bundle of channels (TDM DS0s with CE variant, or voice and signaling with CV) uses only a single IP address and a single UDP port number. On the other hand, VoIP is more complex because it requires an IP address per IP phone. This makes network address translation (NAT) more complicated. In addition, with VoIP, firewalls must either leave open many UDP ports and expose internal networks to attack, or must support H.323/SIP intelligence to open ports dynamically – but that adds complexity and increases call setup time.

Network Requirements	TDMoIP (CE)	TDMoIP (CV)	VoIP
Bandwidth required	Abundant	Low	Moderate
QoS: sensitivity to latency	Usually High	Usually Low	Usually Low
QoS: sensitivity to packet loss	High	Low	Low
QoS: prioritization required	Yes	Yes	Yes
Minimum PPS required for 60 channels	364	33.3	1000
Single UDP destination port (IANA-registered), simplifying flow classification	Yes	Yes	No
Firewall must leave open many UDP ports (exposing internal networks to attack)	No	No	Yes
IP addresses	Single IP address for multiple T1s		IP address per IP phone

Target Network Environments

In the table below, you will notice that all three technologies are suitable in network environments that provide abundant, low latency bandwidth with QoS. TDMoIP (CE) is unique in that it delivers E1/T1 or E3/T3 circuits, which is very important in applications such as cellular backhaul. In order for these circuits to be reliable, the underlying infrastructure must be equally reliable. TDMoIP (CV) and VoIP are not required to support end-to-end clock synchronization. They are also much more tolerant of packet loss, jitter and latency; and therefore are also suitable in bandwidth constrained environments, such as those using DSL, Cable, Wireless, and satellite. Over the public Internet, which inherently offers no QoS guarantees, TDMoIP (CV) provides the best solution because it uses the least bandwidth and because of the way it multiplexes multiple channels and aggregates them into larger IP packets.

Target Network Environments	TDMoIP (CE)	TDMoIP (CV)	VoIP
Enterprise Campus (LAN)	Yes	Yes	Yes
Metro Ethernet (MAN)	Yes	Yes	Yes
Utility / UTelco Ethernet / Fiber networks	Yes	Yes	Yes
DSL	BW dependent	Yes	Yes
Cable HFC using DOCSIS 1.1 modems	Yes	Yes	Yes
Cable HFC (supporting Fast, or at least FDX, Ethernet)	Yes	Yes	Yes
Long distance / International (WAN)	BW dependent	Yes	Yes
Wireless (point-to-point / multipoint, e.g. 802.11b)	BW dependent	Yes	Yes
Satellite	BW dependent	Yes	Yes
Public Internet with minimum bandwidth and tolerance for lost packets	No	BW dependent	

BW= Bandwidth

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5. Return on Investment (RoI) Considerations

Regarding capital and operating expenses, both VoIP and TDMoIP help lower networking costs by taking advantage of efficient IP and Ethernet equipment. TDMoIP (CV) and VoIP also reduce leased line costs by saving bandwidth with voice compression and voice activity detection. In addition, TDMoIP lowers capital expenses because it protects the investment in legacy PBXs without compromising existing features and without the need to retrain staff on a new telephone system. TDMoIP also minimizes operating expenses because it is much simpler to deploy and maintains a pure transport solution rather than a more complex voice switching and transport solution (VoIP). TDMoIP just leaves all the voice switching to the PBX and continues to support all the features transparently without reconfiguration.

Return on Investment (RoI) Considerations	TDMoIP (CE)	TDMoIP (CV)	VoIP
Migration to IP – investment protection in legacy PBX	Yes	Yes	Limited
PBX features – maintained, no staff retraining	Yes*	Yes*	No
OPEX – reduced, simple installation and maintenance	Yes	Yes	No
CAPEX – low, avoids expensive “forklift” upgrades	Yes	Yes	Limited
Networking costs – lowered by taking advantage of efficient IP and Ethernet equipment	Yes	Yes	Yes
Voice compression, VAD – reduce leased line costs by saving bandwidth	No	Yes	Yes
TDM-based equipment – investment protected	Yes	No	No

*Supports all CAS & HDLC-based CCS protocols, including R2, E&M, ISDN, Q.SIG, SS7 and even proprietary PBX protocols.

RAD Products Supporting these Technologies

Since 1981, RAD Data Communications has developed core competencies in technologies such as Voice, ATM, Fiber, TDM, IP and Ethernet. RAD's IPmux and MP-2100 product lines implement TDMoIP (CE) for T1/E1 and T3/E3 circuit emulation over IP/Ethernet. Some important features of these product lines include:

- Ethernet switching with rate limiting capability
- Broad range of analog and digital interfaces, including FXS, FXO, E&M, ISDN BRI, Serial, etc.
- 100BaseFX uplinks.

The Vmux product line has many of the same features, but implements TDMoIP (CV) for voice compression and fax relay. The KM-2100, MP-2100 and MP-2200 product lines have the broadest range of interface options, to provide flexibility in more complex applications. The MP-2100H is a hybrid supporting VoIP and TDM interfaces.

RAD Products supporting the technologies	TDMoIP (CE)	TDMoIP (CV)	VoIP
IPmux-1 / IPmux-1E / IPmux-8 / IPmux-16	Yes	No	No
VMUX-2100 / VMUX-110	No	Yes	No
KM-2100 / KM-2104	Yes	Limited*	No
MP-2100 / MP-2104	Yes	Yes	No
MP-2100H / MP-2104H	No	No	Yes
MP-2200	Yes	No	No

*Kilomux supports some of the TDMoIP (CV) compressed voice features but is not interoperable with the technology.

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Correlating Technologies to Application Requirements

As shown in the table below, those applications requiring circuit emulation, such as cellular backhaul (BTS to MSC), T1/E1 leased line services and T3/E3 over FE/GbE/MPLS, all require TDMoIP (CE). On the other hand, voice / fax applications that need to make most efficient use of bandwidth use TDMoIP (CV) and VoIP. In the next few pages, we show solutions using TDMoIP (CE) and TDMoIP (CV).

Applications	TDMoIP (CE)	TDMoIP (CV)	VoIP
Connecting PBXs over IP/Ethernet/MPLS	Yes	Yes	Limited
Connecting legacy PBXs over the public Internet	No	Yes	Limited
Voice switching over IP (Internet telephony)	N/A – Performed by PBX		Yes
Encrypted / secure voice, video & data over IP/Ethernet/MPLS	Yes	No	No
Serial data (async /sync) and SCADA over IP/Ethernet	Yes	No	No
Rural telephony over WLL, Satellite, DSL and Cable	Limited	Yes	Yes
T1/E1 leased-line services over IP/Ethernet/MPLS	Yes	No	No
Private line services over IP/Ethernet/MPLS	Yes	No	No
Alternate routing (1+1 or 1:1) of T1/E1 circuits over IP	Yes	No	No
T3/E3 over FE/GbE/MPLS	Yes	No	No
International toll bypass	No	Yes	Yes
Remote call center (extending voice circuits)	Limited	Yes	Limited
Cellular backhaul (BTS to BSC/MSC)	Yes	No	No
Cellular (MSC to MSC)	Yes	Yes	No
SS7 signaling over IP	Yes	Yes	No
Disaster recovery –rerouting voice circuits over satellite backup facilities (transparency with compression)	No	Yes	No
Disaster recovery and path redundancy – rerouting T1/E1 circuits across alternate IP/Ethernet/MPLS paths (1+1 & 1:1)	Yes	No	No
Disaster recovery – rerouting voice circuits to backup PBXs and telephone switches over IP/Ethernet/MPLS	Yes	Yes	Limited
Monitoring of T1/E1 circuits over IP/Eth/MPLS networks	Yes	No	No
DS0 / DS1 cross-connect over IP/Ethernet/MPLS	Yes	No	No

6. TDMoIP (CE) Solutions

Leased Line Services over IP

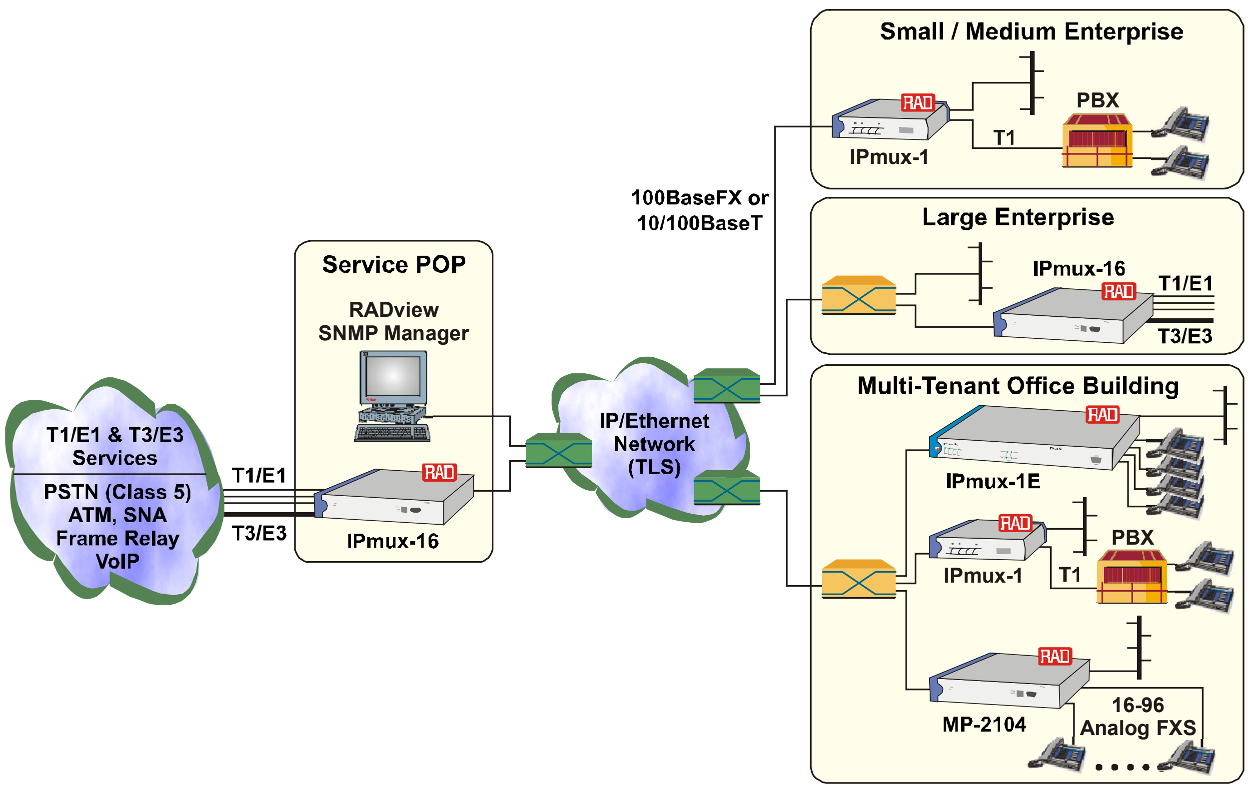
Your Needs

- Generate new revenue by delivering T1/E1 and T3/E3 services over scalable Ethernet/fiber infrastructures.
- Extend services to rural areas inadequately served by market-driven carriers that do not have infrastructure in rural areas (act as carrier's carrier).
- Simplify network and reduce costs by replacing complex ATM and TDM infrastructures with simple, scalable Ethernet and IP over Fiber or wireless links.

RAD Solution

RAD's **IPmux** extends T1/E1 and T3/E3 circuits transparently over Ethernet-based data networks, eliminating the need for leased lines, and supporting all voice, TDM and video applications transparently over IP. The IPmux-1 and IPmux-1E are ideal Integrated Access Devices for in-office integration of voice and data featuring T1/E1 or analog voice (and optional Ethernet) user interface(s) and an Ethernet uplink (10/100BaseT or 100BaseFX).

The IPmux-8 and IPmux-16 are ideal for in-building integration capable of dropping of 8 or 16 T1/E1 circuits. The IPmux-16 also can be used to deliver T3/E3 circuits over the Ethernet infrastructure.



Cellular Backhaul over IP/Ethernet – Fixed Wireless, Coax or Fiber Access

Your Needs

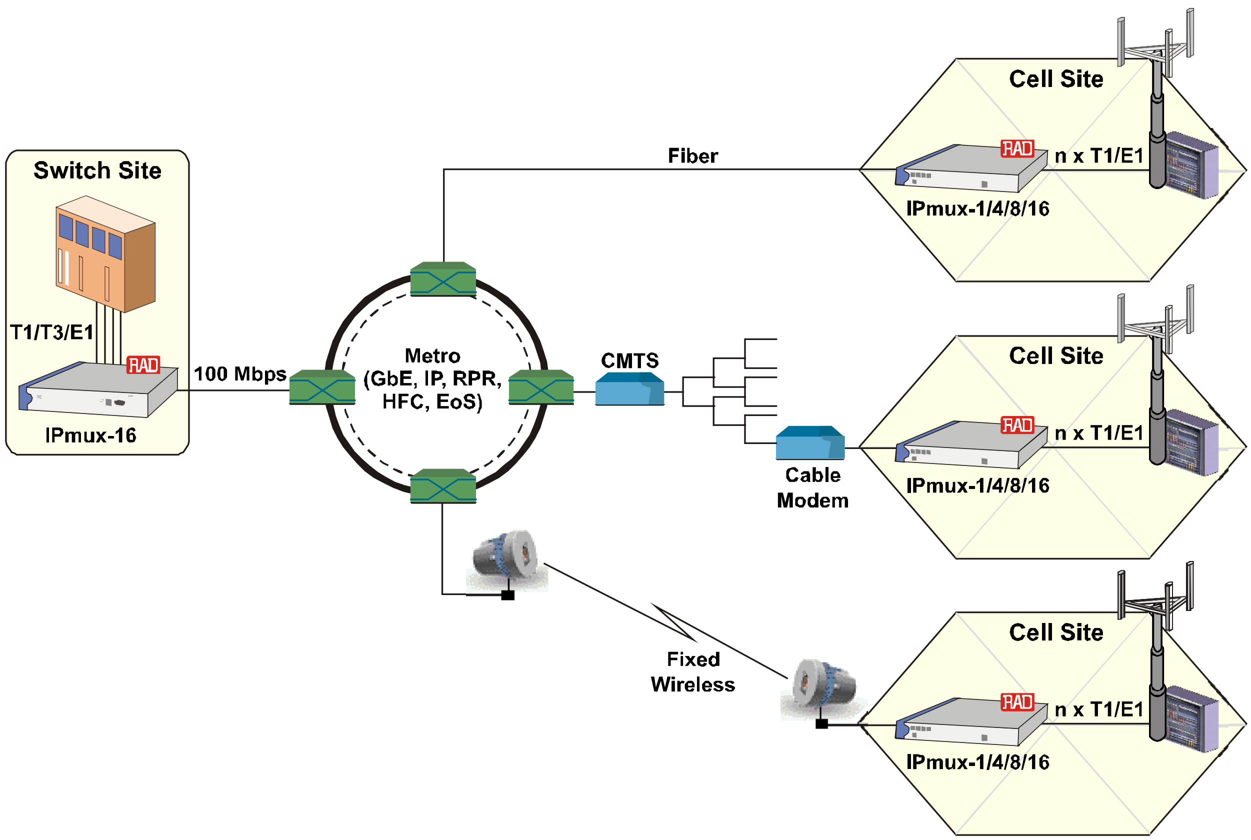
- Reduce 2G access costs, since T1/E1 leased lines accounts for 40%-60% of the operational expenses incurred by mobile operators.
- Help position the mobile operator for future 3G network expansion.

Background

Until now cellular operators have relied solely on traditional T1/E1 leased lines from the incumbents, which has caused frustrating delays and limited their footprint. TDMoIP makes it simple to use an Ethernet access network.

RAD Solution

RAD's **IPmux** extends T1/E1 circuits transparently over Ethernet-based data networks, eliminating the need for leased lines, and supporting cellular voice protocols, such as CDMA, transparently over IP. The Ethernet network can be built on fiber, coax or fixed wireless infrastructures.



SS7 Transport over IP

Environment

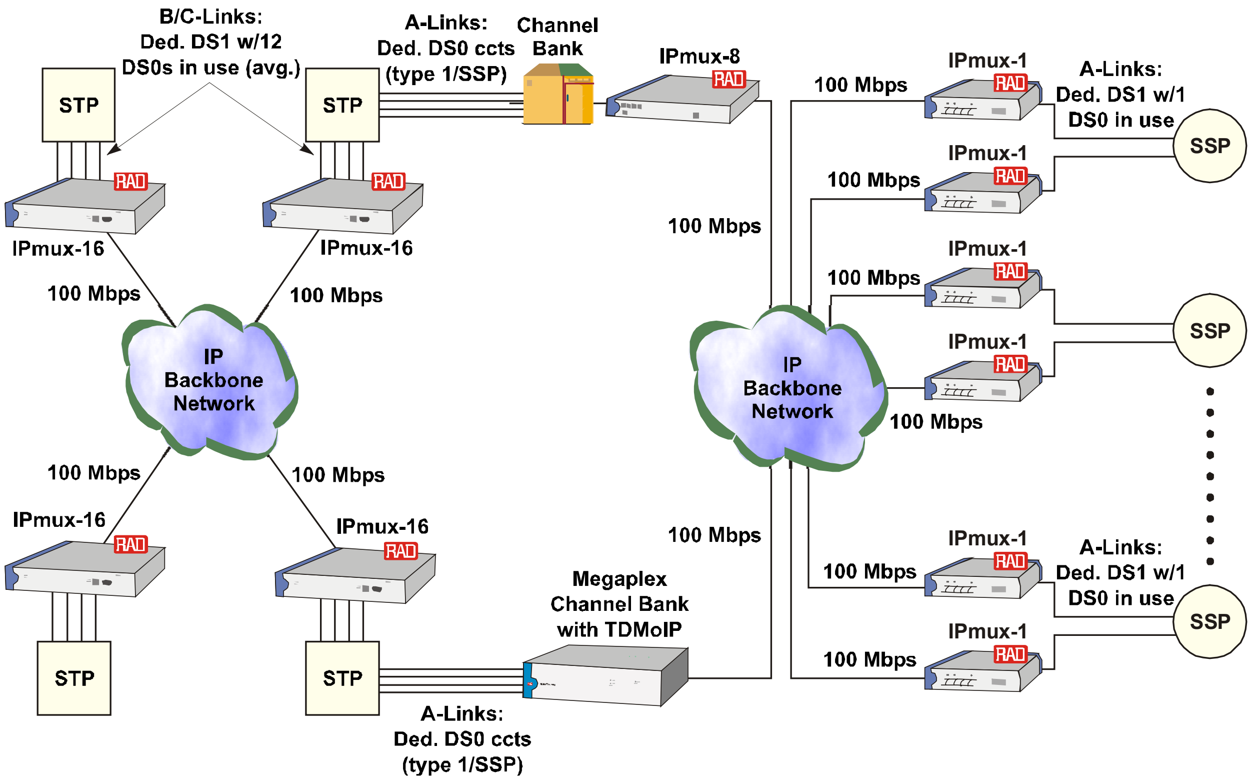
SSPs are located in different cities, and connected with DS0 circuits to STPs supporting that region. For redundancy, DS0s are required to be on separate DS1 circuits across the transport network.

Your Needs

Reduce SS7 transport costs without compromising redundancy.

RAD Solution

RAD's **IPmux** and **Megaplex** product lines extend full and fractional T1/E1 circuits transparently over IP networks, eliminating the need for leased lines and supporting signaling protocols such as SS7. The IPmux can therefore be used to connect between STPs or between STPs and SSPs in point-to-point and point-to-multipoint configurations with any-to-any DS0 level cross-connect capability. Note that the Megaplex also supports serial interfaces such as V.35, RS-530, RS-232 and DDS over IP, eliminating the need for a separate channel bank.



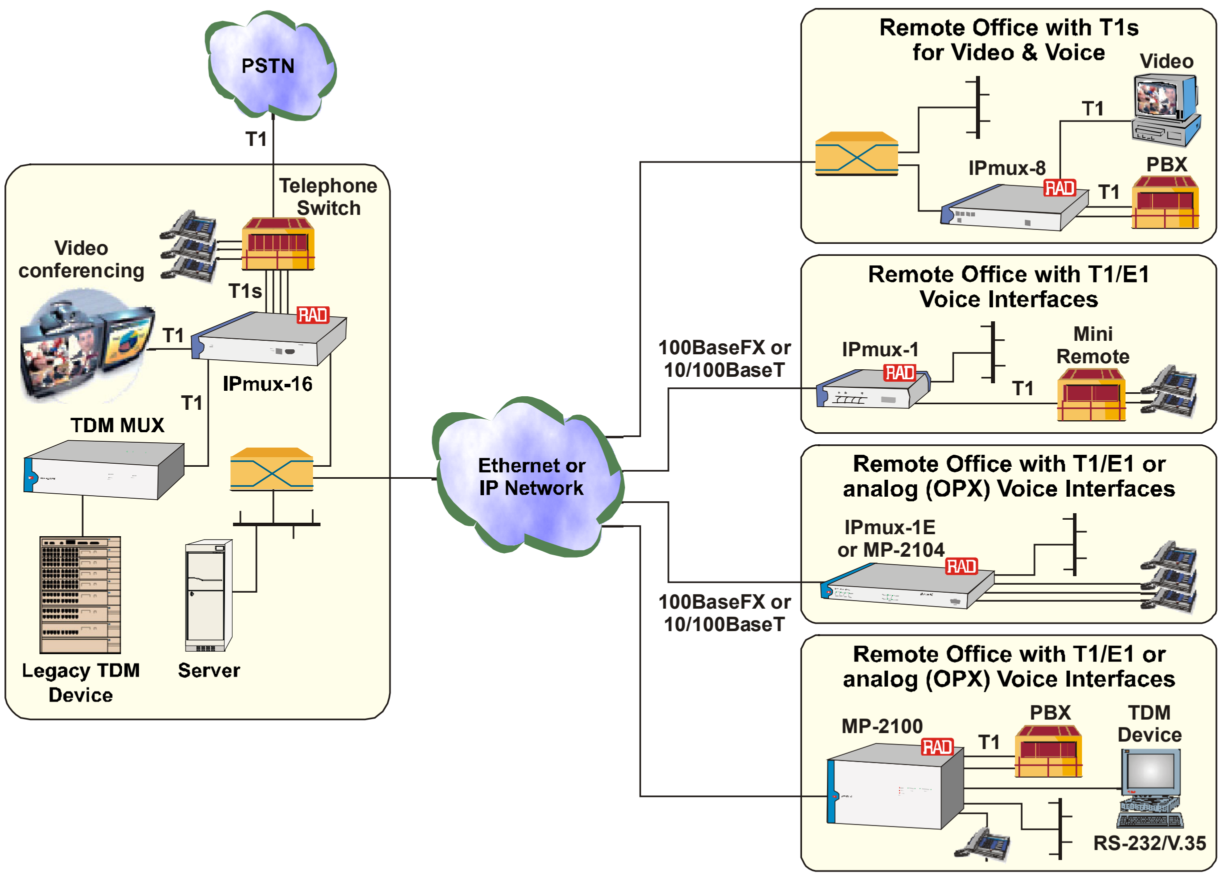
T1/E1, Voice, Video & TDM over Ethernet

Your Needs

- Better integration of remote offices with voice, video, LAN and filtered Internet services over campus or metro Ethernet networks.
- Reduce network expenses, especially for public schools, utilities, local governments and businesses, by eliminating leased lines.
- Achieve all these simply and inexpensively without requiring upgrades to existing telephony, video conferencing and security systems. The solution must be transparent to existing voice and video features such as voice mail, caller ID, direct station select (DSS) and E911.

RAD Solution

- Ethernet provides high bandwidth at low cost and is excellent for extending LAN and filtered Internet services to remote offices or schools.
- RAD's IPmux and Megaplex products extend T1/E1 circuits transparently over Ethernet-based data networks, thereby eliminating the need for leased lines and supporting all voice, TDM and video applications.
- Integrated Ethernet switches in the IPmux-1/1E and MP-2100 with ML-IP module simplify deployment at remote locations with a single box supporting all applications.



T1/E1 or T3/E3 over Fast/Gigabit Ethernet

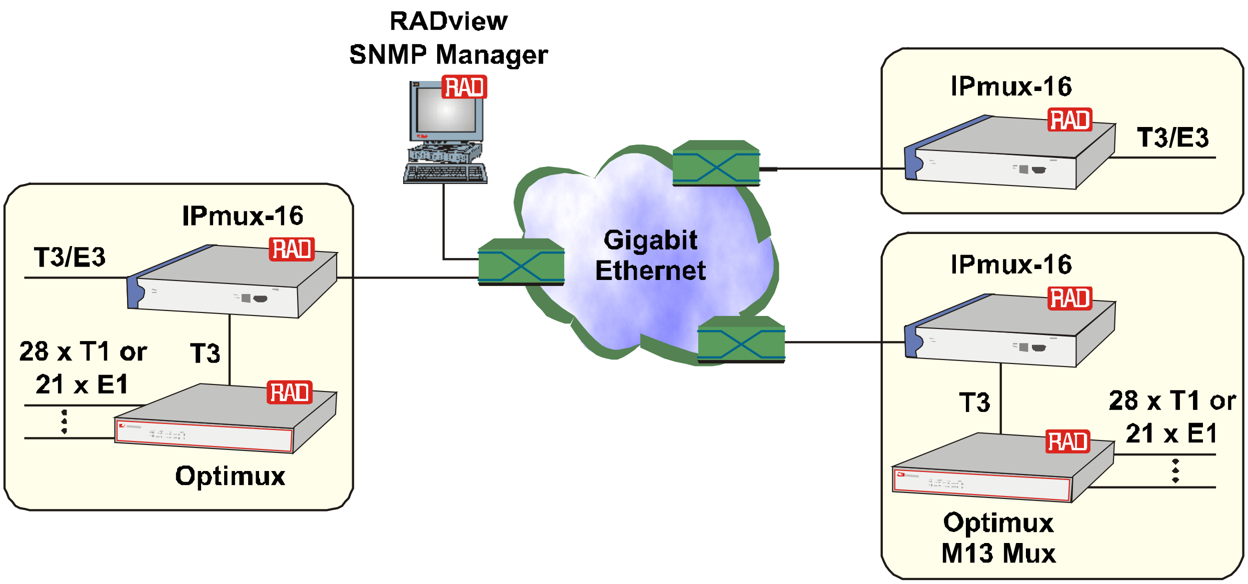
Your Needs

Inexpensively extend T3/ E3 circuits or large quantities of T1/E1 circuits over a high bandwidth Ethernet network (e.g. Gigabit Ethernet).

RAD Solution

The RAD **IPmux-16** can be used to extend single or dual T3/E3 circuits transparently over a high bandwidth IP or Ethernet network. RAD's M13 MUX, the Optimux-T3, complements this solution by allowing 28 T1s or 21 E1s (using the G.747 standard) to be extended over each of these T3s.

These circuits are totally transparent to signaling and protocols. As a result, the features and quality of traditional voice, video and data applications between locations are not compromised. For example, frame relay and ATM, even at T3/E3 rates, can now be carried over fast or Gigabit Ethernet.



Voice, SCADA and LAN in Daisy Chain or Ring Topology

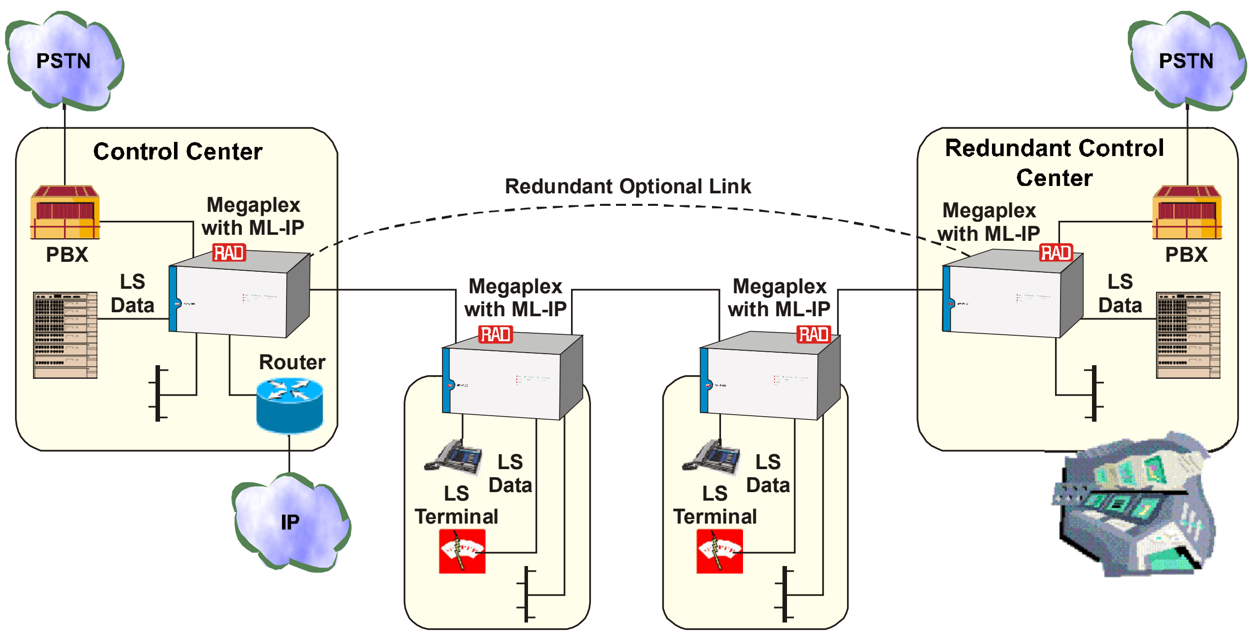
Your Needs

- Support voice, data (e.g., SCADA) and LAN applications in a daisy chain or ring topology over a fiber infrastructure.
- Single box solution must be capable of providing redundancy by connecting to both upstream and downstream system controllers and PBXs to ensure communications are not disrupted in the event of a fiber cut.

RAD Solution

RAD's **Megaplex-2100/2104** is an integrated Ethernet switch and TDM multiplexer capable of supporting traditional voice, video, data and LAN applications in a variety of topologies, including daisy chain and ring, with optional redundancy and resiliency (50 ms recovery).

The ML-IP module allows separate bundles of voice and data timeslots to be directed upstream and downstream to their respective host sites with the flexibility of a cross-connect and the simplicity and scalability of Ethernet.



7. TDMoIP (CV) Solutions

Connecting PBXs over IP – even over Public Internet!!!

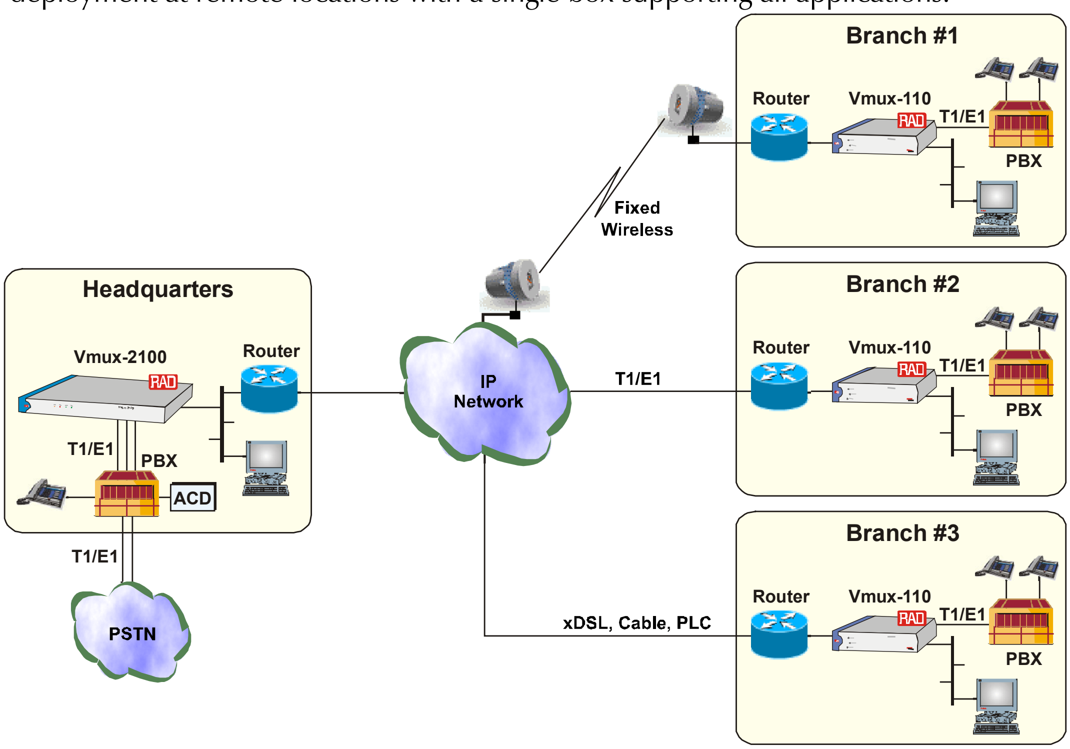
Your Needs

To reduce network expenses by eliminating T1/E1 voice circuits between enterprise offices, while retaining all central site PBX features that are not supported by VoIP, such as voice mail, caller ID, Direct Station Select (DSS) and Busy Lamp Field (BLF).

RAD Solution

RAD's **Vmux** product line utilizes TDMoIP (CV) technology to extend voice circuits (including signaling) transparently over IP networks and therefore eliminate the need for expensive T1/E1 voice circuits between enterprise offices. Transparent support for standard and proprietary signaling protocols, including ISDN PRI, Q.Sig, DPNSS and SS7, is important for the enterprise because it protects their investment in legacy PBXs and provides an evolutionary and simple means of supporting all phone features without expensive "forklift" upgrades. Business processes are not affected and there is no need to retrain users.

In this application TDMoIP (CV) performs voice compression and silence suppression, before aggregating and multiplexing multiple voice channels into a single IP bundle. This combination provides a solution that uses less overhead and bandwidth than VoIP and is more tolerant of packet loss, latency and jitter. Therefore, the Vmux is ideal for applications where bandwidth is constrained (e.g. 802.11b wireless, cable, xDSL, Power Line Communications (PLC) or even the public Internet). Integrated Ethernet switches and optional analog voice interfaces in the Vmux-110 and Vmux-2100 simplify deployment at remote locations with a single box supporting all applications.



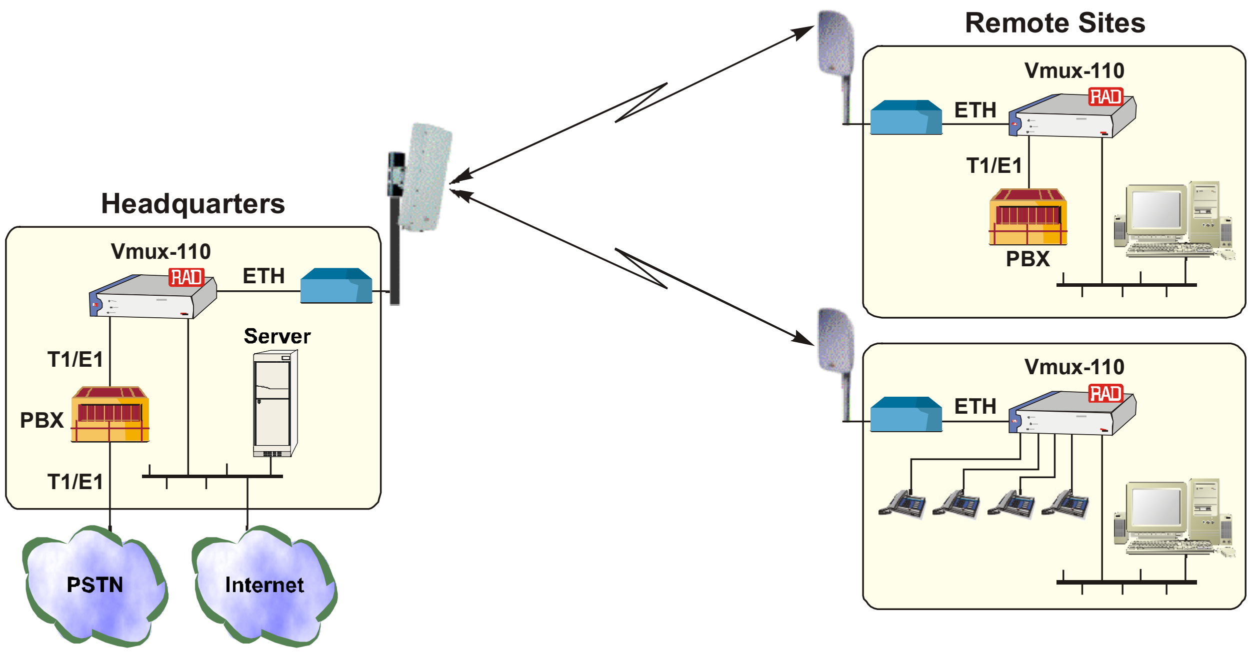
Voice, Data and Internet Service over Wireless (WLL/LMDS)

Your Needs

To extend LAN and voice traffic over an inexpensive wireless point-to-multipoint infrastructure.

RAD Solution

RAD's **Vmux-110** is a compact, 1U high product that is ideal for wireless links because it delivers LAN and high quality voice to remote locations over IP or Ethernet. The Vmux-110 uses voice compression, rate-configurable fax relay and dynamic bandwidth allocation to minimize bandwidth requirements. A single Vmux-110 with T1/E1 can provide 24/30 voice channels to as many as 12 remote locations using the TDMoIP bundling mechanism, making it ideal for not only point-to-point, but also point-to-multipoint applications. The remote Vmux-110s can be ordered with either T1/E1 voice interfaces for connection to remote PBXs or with FXS interfaces for connecting to analog telephones or key systems.



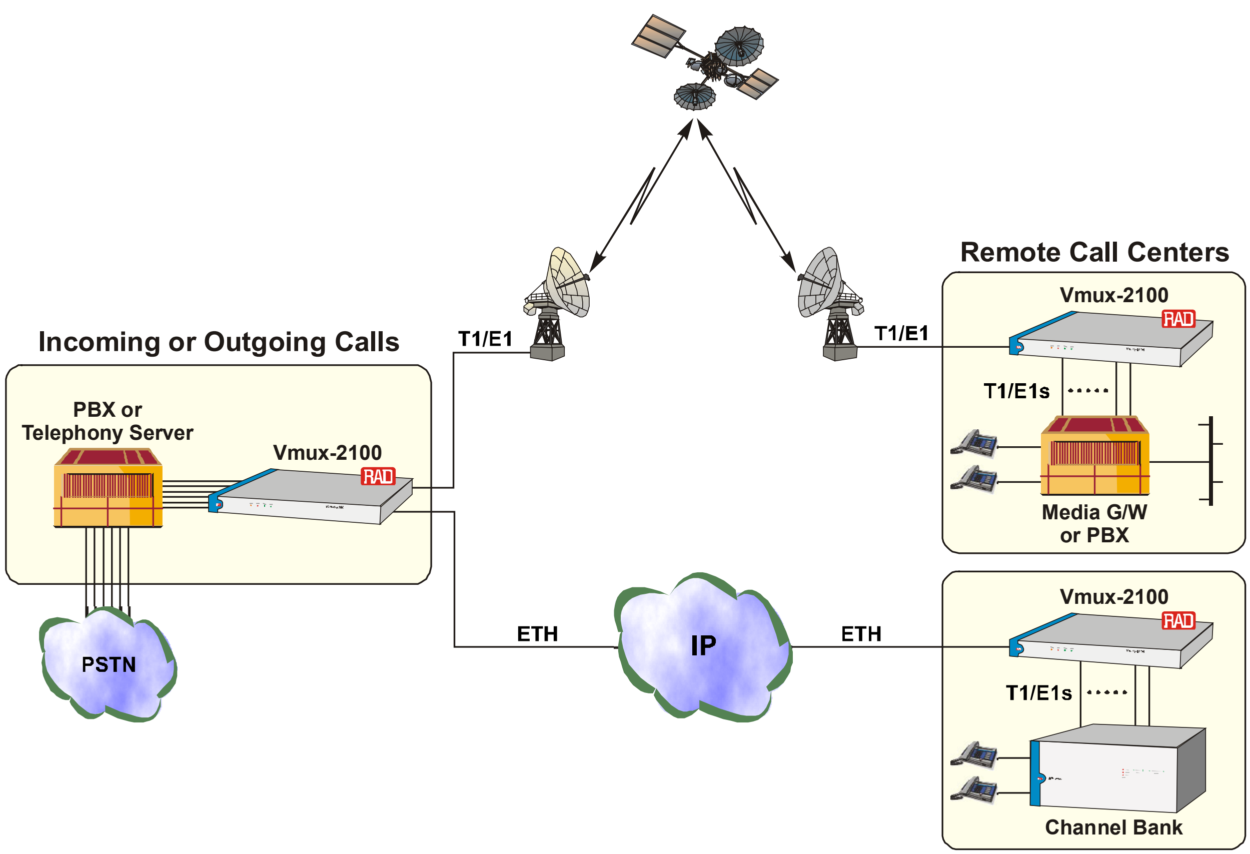
Remote Call Center – 16:1 Voice Compression for Lower Cost Transport

Your Needs

To deliver voice circuits to remote call centers for incoming customer service calls or outgoing sales/marketing calls. The solution must be simple to deploy at remote call centers that have limited technical support staff. It also needs to efficiently utilize expensive satellite, leased line and/or IP bandwidth to minimize networking costs. Since call center functionality cannot be compromised, all PBX or telephony server features, such as predictive dialing and automatic call distribution (ACD), must be supported transparently.

RAD Solution

RAD's **Vmux-2100** is a scalable, compact, 1U high product that is ideal for remote call centers because it is easy to deploy and efficiently delivers high quality voice with 16:1 voice compression. The Vmux-2100 also transparently supports all PBX or telephony features to ensure that call center functionality is not compromised.



RAD data communications

Voice Trunking (International Toll Bypass with 16:1 Compression)

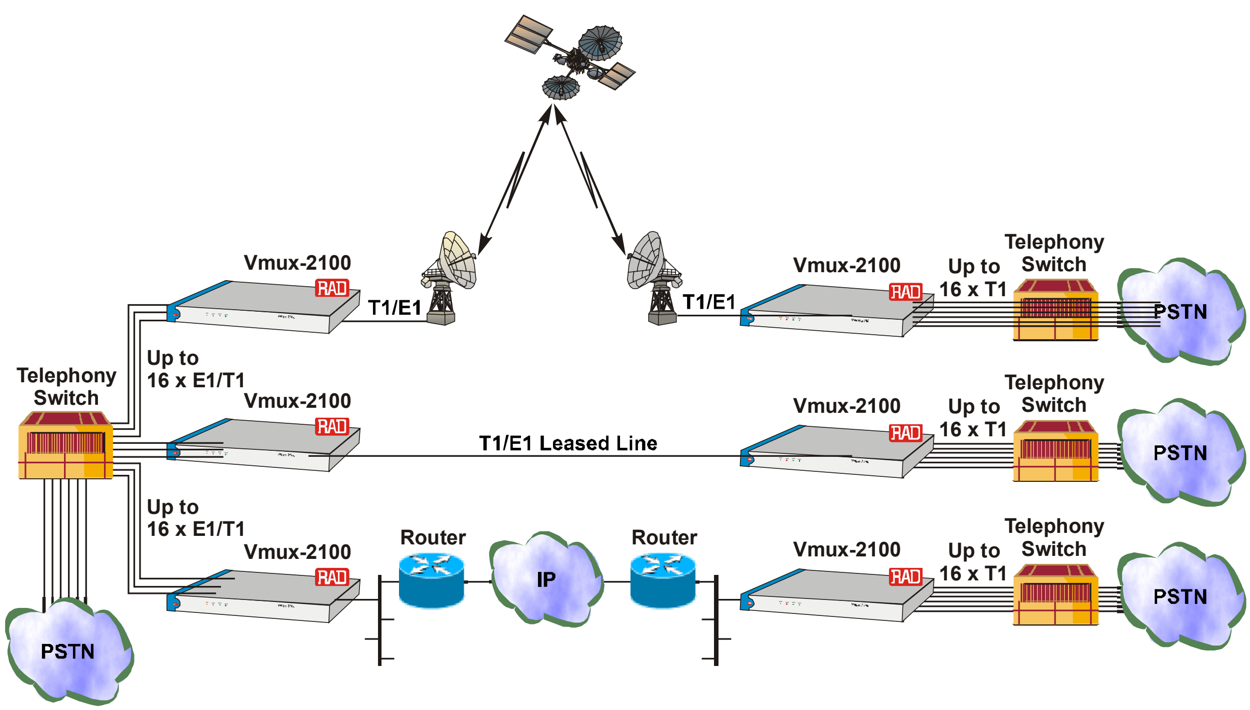
Your Needs

To reduce transmission costs significantly and improve utilization of expensive satellite, leased line and IP links. The solution must be optimized for voice, but also should support Group 3 fax relay at configurable rates and therefore support wholesale carrier voice services.

RAD Solution

RAD's **Vmux-2100** is a compact, 1U high product that is ideal for international and domestic toll-bypass. It can be configured to compress 16 T1 or E1 voice circuits into a single T1/E1 or IP link. The Vmux-2100 utilizes TDMoIP (CV) technology to deliver high quality voice with 16:1 voice compression, rate configurable fax relay and transparent support for common signaling protocols such as SS7, ISDN PRI and R2/MFC.

TDMoIP (CV) uniquely aggregates and multiplexes multiple voice channels into a single IP bundle to ensure minimum overhead, bandwidth and latency. The technology provides a high tolerance to packet loss and jitter. With integrated voice compression and silence suppression, it consumes about 1/3 the bandwidth of equivalent VoIP systems (~4kbps/voice channel).



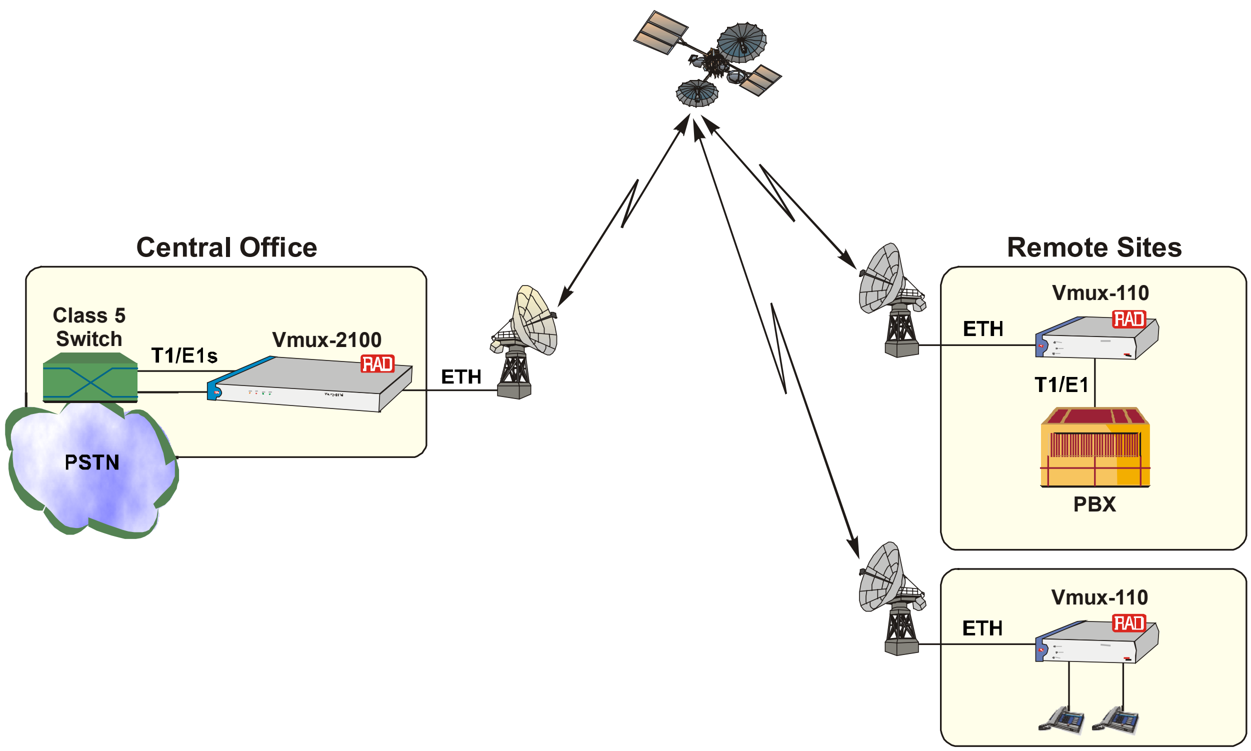
Rural Telephony over Satellite

Your Needs

To cost-effectively provision telephony services to rural towns/villages that do not have an established telephone infrastructure (for example, mountainous areas in developing countries). The solution must be optimized for voice, but also should support Group 3 fax relay.

RAD Solution

RAD's **Vmux-2100** and **Vmux-110** are compact, 1U high products that are ideal for delivering high-quality voice with 16:1 voice compression and rate configurable fax relay. FXS, T1 and E1 telephone interfaces are supported, and Ethernet traffic can be bridged over the same satellite links.



Disaster Recovery – Rerouting Voice Circuits over Satellite Backup Facilities

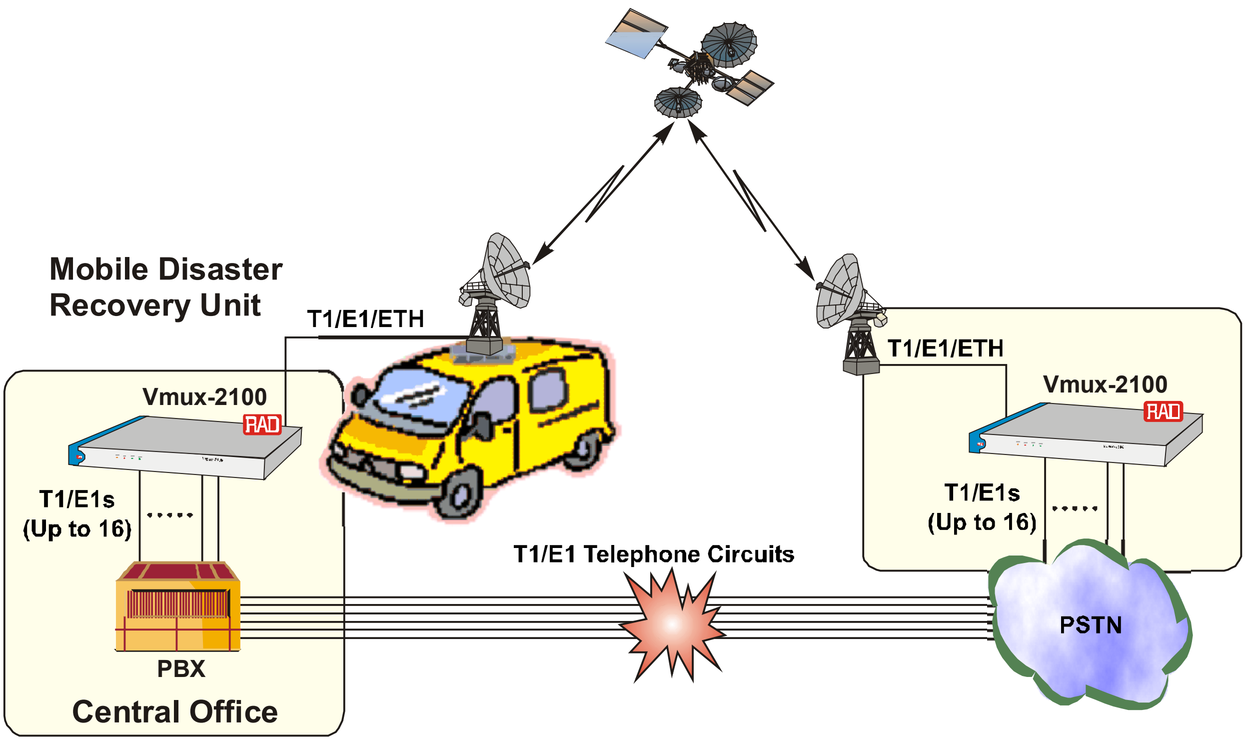
Your Needs

To efficiently reroute many T1/E1 voice circuits around a fiber cut that severs communications between central offices. The solution must be compact, mobile, and very easy to deploy in emergencies. The solution must take into account that satellite bandwidth is expensive and limited.

RAD Solution

RAD's **Vmux-2100** is a compact, 1U high product that is simple and quick to deploy. It is excellent for expensive satellite applications because it utilizes TDMoIP (CV) technology to deliver high quality voice with 16:1 voice compression. The Vmux-2100 also provides rate configurable fax relay and transparent support for common signaling protocols such as SS7, ISDN PRI and R2/MFC.

TDMoIP (CV) uniquely aggregates and multiplexes multiple voice channels into a single IP bundle to ensure minimum overhead, bandwidth and latency. The technology provides a high tolerance to packet loss and jitter. With integrated voice compression and silence suppression it consumes about 1/3 the bandwidth of equivalent VoIP systems (~4 kbps/voice channel).



8. Technology/Application Comparison

Applications	TDMoIP (CE)	TDMoIP (CE)	TDMoIP (CE)
Connecting PBXs transparently over IP/Ethernet/MPLS	Yes	Yes	Limited
Connecting legacy PBXs over the public Internet	No	Yes	Limited
Voice switching over IP (Internet telephony)	N/A – performed by PBX		Yes
Encrypted/secure voice, video & data over IP/Ethernet/MPLS	Yes	No	No
Serial data (async/sync) and SCADA over IP/Ethernet	Yes	No	No
Rural telephony over WLL, Satellite, DSL and Cable	Limited	Yes	Yes
T1/E1 and T3/E3 circuit extension over IP/Ethernet/MPLS	Yes	No	No
International toll bypass & remote call center	Limited	Yes	Limited
Cellular backhaul (BTS to BSC/MSC)	Yes	No	No
Cellular (MSC to MSC)	Yes	Yes	No
SS7 signaling over IP	Yes	Yes	No
Disaster recovery – rerouting voice circuits over Satellite backup facilities (transparency with compression)	No	Yes	No
Disaster recovery and path redundancy – rerouting T1/E1 circuits across alternate IP/Ethernet/MPLS paths (1+1 & 1:1)	Yes	No	No
Voice	**TDMoIP (CE)**	**TDMoIP (CV)**	**VoIP**
Compressed voice with toll quality (G.729a, G.723.1)	No	Yes	Yes
Uncompressed toll quality voice (G.711)	Yes	Yes	Yes
Low-latency, uncompressed, toll quality voice (G.711)	Yes	Limited	Limited
Minimum end-to-end latency (through two gateways without network delay)	3 ms	45 ms	45 ms
Minimum bandwidth required per voice channel	74 kbps	3.8 kbps*	10.4 kbps*
Fax & modem support (transparent – voice band data)	Yes	Limited	Limited
Group III fax relay support at rates of 4.8 to 14.4 kbps	Transparent	Yes	Yes
Transparent to standard & proprietary signaling protocols	Yes	Yes**	No
Requires conversion to complex voice signaling protocols	No	No	Yes

Data and Video Protocols	TDMoIP (CE)	TDMoIP (CV)	VoIP
HDLC / Frame Relay / X.25	Yes	Yes	No
Async/sync protocols including ATM, H.320 Video, etc.	Yes	No	No
Target Network Environments	**TDMoIP (CE)**	**TDMoIP (CV)**	**VoIP**
Enterprise Campus (LAN)	Yes	Yes	Yes
Metro Ethernet (MAN)	Yes	Yes	Yes
Utility / UTelco Ethernet / Fiber networks	Yes	Yes	Yes
DSL, Satellite and Cable (using DOCSIS modems)	BW dependent	Yes	Yes
Cable HFC – supporting Fast (or at least FDX) Ethernet	Yes	Yes	Yes
Long distance / International (WAN)	BW dependent	Yes	Yes
Wireless (point-to-point / multipoint, e.g. 802.11b)	BW dependent	Yes	Yes
Public Internet with minimal bandwidth and tolerance for lost packets	No	BW dependent	BW dependent
Return on Investment (RoI) Considerations	**TDMoIP (CE)**	**TDMoIP (CV)**	**VoIP**
Migration to IP with investment protection in legacy PBX	Yes	Yes	Limited
Maintains all existing PBX features – no staff retraining	Yes	Yes**	No
Reduced OPEX with simple installation and maintenance	Yes	Yes	No
Low CAPEX by avoiding expensive "forklift" upgrades	Yes	Yes	Limited
Lowers networking costs by taking advantage of efficient IP and Ethernet equipment	Yes	Yes	Yes
Reduces leased line costs by saving bandwidth with voice compression, voice activity detection and silence suppression	No	Yes	Yes

BW= Bandwidth

*Average bandwidth per voice channel with equivalent MAC, IP and UDP overhead (based on system carrying 60 channels of voice with G.723.1 coder, 5.3 kbps compression, 30 ms packetization interval, and 50% silence suppression).

**Supports all CAS & HDLC-based CCS protocols incl. R2, E&M, ISDN, Q.SIG, DPNSS, SS7 and even proprietary PBX protocols

Appendix A. TDMoIP (CV) Speech Quality Scores

PAMS Speech Quality Scores						Listening**	
Protocol	**Coder**	**Packet Size**	**Jitter* [ms]**	**Packet Loss**	**Timeout**	**Effort**	**Quality**
ISDN	G.723	max	250/50	5%	Default	3.65	3.42
	G.711	max	50 ms fixed latency	No	Default	4.63	4.54
CAS	G.729	max	150/30	1%	Default	4.08	3.94
CAS	G.723	max	150/30	1%	Default	4.06	3.95
ISDN	G.711	1400	50 ms fixed latency	No	30 ms	4.62	4.53
ISDN	G.723	1000	250/50	5%	60 ms	3.38	3.09
CAS	G.723	max	250/50	5%	Default	3.68	3.44
CAS	G.729	500	250/50	5%	10 ms	3.62	3.26
CAS	G.723	1400	150/30	1%	90 ms	3.93	3.86
CAS	G.729	500	150/30	1%	30 ms	4.05	3.91
ISDN	G.729, E1 data	1400	No	No	30 ms	4.30	4.18
CAS	G.723, E1 data	max	No	No	Default	4.19	4.11
CAS	G.729, E1 data	500	No	No	10 ms	4.26	4.12
CAS	G.723, E1 data	1400			90 ms	4.21	4.12
CAS	G.729, E1 data	500			30 ms	4.26	4.12

*Jitter=Network Delay Variation

**Voice quality scale: 1 – 5

☞*High voice quality is maintained even at high packet loss rates!*

RAD data communications

Appendix B. Latency and Interactive Voice Quality

Interactive Voice Quality* and Latency**

Vendor	Vocoder	Encoding Rate (kbps)	Latency (ms)	Interactive Rating (5=Excellent, 1=Poor)
RAD Data Communications	PCM – small payload	64	5	5.00
RAD Data Communications	PCM – large payload	64	10	5.00
Nuera	G.711	64	45	4.83
Nuera	GSM-EFR	12.2	75	4.83
Nuera	G.729a	8	77	4.83
Net.com	G.711	64	112	4.67
Arelnet	G.711	64	75	4.50
Arelnet	G.729a	8	152	4.50
Arelnet	G.723.1	6.3	205	4.33
Net.com	SX 9600	9.6	116	4.17
Net.com	G.723.1	6.3	203	3.50

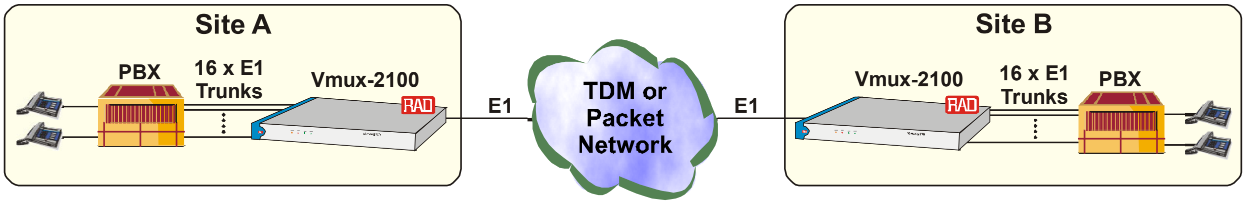
* Interactive ratings are the average of at least six individual ratings made by each of at least three experienced Miercom VoIP lab testers. The testers, in rotation, conduct real-time, two-way conversations and special tests, over separate connections, between identical analog phones connected via the VoIP gateway under test. The interactive (or conversational) quality of each connection is then rated, based on: the effect of latency; clarity, including background noises; and bidirectionality.

**One-way latency, including network delays, in milliseconds.

Source: Business Communications Review/Sept 2001

Appendix C. Vmux Bandwidth with 16:1 Compression

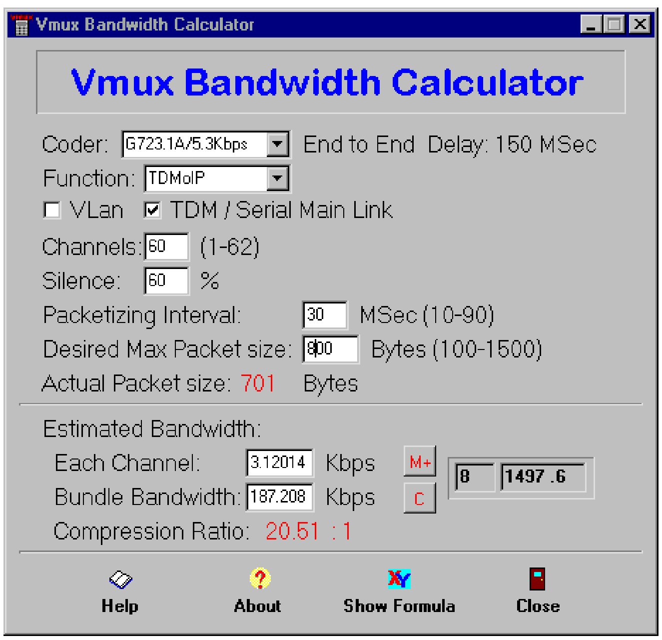
Following is an example showing 16:1 compression (16 E1s over a single E1 link) in a Vmux.



In this example the Vmux compresses 16 E1 trunks over a single E1 link. Since one Vmux voice bundle can carry up to 2 E1 trunks we will need 8 bundles between the two sites (8 x 2=16 E1s).

Bandwidth Required for Voice

We will use the calculator to estimate the bandwidth for one bundle with the following parameters:



Compression code	G.723 (5.3 kbps)
Bundle function	TDMoIP
Main Link	TDM/Serial
Voice channels	60 (2 E1 PRI Trunks)
Silence rate (estimated)	60% (40% voice activity)*
Packetizing interval	30 ms
Max packet size	800 Bytes

* Notes:

1. Studies show that an average person engaged in a telephone conversation speaks only 40% of the time, thinks 10% between spoken sentences and listens to the other party 50% of the time. This means that in regular conversation there is potentially 60% silence. In noisy environments (for example, a mobile phone conversation in the street) silence rate will be lower than 60%, since there is voice activity from both sides.

2. When channel is not active (ON Hook) this means there is 100% silence on the channel, so it is important to know how many channels are active in the system on peak hours.

3. If we review comparable VoIP estimates for silence suppression, we see that it is common for vendors to estimate 50%-65% silence suppression, so this is consistent with the studies. In this example, we used 60%.

The estimated bandwidth that will be required for the voice bundle is 187.208 kbps, about 3.12 kbps instead of 64 kbps for each channel, which is a compression ratio of 20.51 to 1. Now, since there are 8 bundles, we have to multiply the calculated bandwidth by 8; i.e. 8 x 187.208 = 1497.6 kbps.

Therefore, 1497.6 kbps is the total bandwidth that will be required for 16 E1s of voice traffic (i.e. 8 voice bundles × 60 voice channels/bundle).

Bandwidth Required for Signaling

Signaling

Everything in the telecommunications network is based on signaling – call setup, connection, teardown, and billing. The two forms of signaling used by networks are:

- Channel Associated Signaling (CAS)
- Common Channel Signaling (CCS)

Signaling System Number Seven (SS7) is a form of common channel signaling that provides intelligence to the network and allows quicker call setup and teardown, thereby saving time and money. Signaling information is processed in Vmux according to signaling mode:

- CAS – E1
- Robbed Bit MF – T1
- CCS – E1 or T1.

CAS/Robbed Bit MF

Call setup information (off-hook, dialtone, address digits, ringback, busy) is transmitted in the same band of frequencies as that used by the voice signal. Voice (talk) path is cut over only when the call setup is complete, using the same path that the call setup signals used.

The signaling data is processed in the Vmux by a separate DSP by extracting the ABCD bits and reporting any change in their status to the host. The reporting format is similar in E1, T1 ESF and T1 SF. No additional bandwidth is required to support this signaling with DTMF and R2-MFC tones being carried in-band.

CCS

CCS employs a separate, dedicated path for signaling. Voice trunks are used only when a connection is established, not before. Call setup time is quicker because resources are more efficiently used.

With the Vmux, the signaling information is transferred transparently to the host, which encapsulates the HDLC frame with the proper IP header and sends it to the main link. HDLC data can be extracted from any set of timeslots and sent to a single destination.

CCS is the technology that makes ISDN and SS7 possible.

ISDN-PRI

Primary Rate Interface (ISDN-PRI) divides digital transport services into bearer channels (B-channels) for voice and data transmission and data channels (D-channels) for signaling data. Typical CCS signaling protocols such as QSIG, DPNSS and CORNET are carried on timeslot 16/24 of E1/T1 PRI circuits (signaling messages in HDLC format). The Vmux uses HDLC flag suppression and then transfers the PRI signaling information over the network as HDLC over IP. The HDLC flag suppression provides a compression ratio of more than 16:1 of the PRI signaling channel.

SS7

While similar to ISDN-PRI, Signaling System Number Seven (SS7) uses different messaging for call setup and teardown. SS7 lets any SS7-enabled node talk to any other, regardless of whether they have direct trunk connections between them.

SS7 signaling is a special kind of CCS that uses more bandwidth because it sends keep-alive packets. When operating the Vmux with SS7 signaling, it is possible to control the amount of keep-alive bits transferred over the signaling links using SS7 suppression. SS7 signaling can be carried inband or out-of-band. Following are bandwidth requirements:

SS7 inband signaling with no suppression

Add 64 kbps for every signaling timeslot

SS7 inband signaling with suppression

Reduce the bandwidth required proportionally, i.e.:

- 10% suppression: 64 kbps – 10% = 64 kbps – 6.4 kbps = 57.6 kbps
- 20% suppression: 64 kbps – 20% = 64 kbps – 12.8 kbps = 51.2 kbps
  …and so on.

SS7 out-of-band signaling –

No additional bandwidth is required because signaling is not carried across the Vmux link.

SS7 out-of-band signaling through integrated 56 kbps channel

64 kbps of bandwidth required.

Bandwidth Required for Fax

Following is a description of the bandwidth required for fax transmission. All calls are only half-duplex, since the transmitter side sends almost all of the data.

Each fax call (14.4) requires ~18.3 kbps

Each fax call (9.6) requires ~13.5 kbps

Each fax call (4.8) requires ~8.7 kbps

Therefore, if we configure the Vmux for 4.8 kbps and assume that fax traffic is being initiated from both ends of the network, then we average 8.7/2 = 4.35 kbps in each direction. If all fax is initiated from one end, then the proportion of fax calls is reduced because more bandwidth is required in one direction.

Conclusion

Considering the above-mentioned assumptions, in situations where we use CAS, CCS (PRI,) or out of band SS7 and where fax ratio is under 50% of total calls, the result is bandwidth savings of at least 16:1.

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