MIER Communications Inc.

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LAB TESTING SUMMARY REPORT

April 2000
Report 030500

Product Category:

TDMoIP Gateways

Vendor Tested:

RAD Data Communications

Product Tested:

IPmux-4 E1/T1 Multiplexer for IP Networks

INDEPENDENT LAB TESTED
NETWORKS AS ADVERTISED™
MIER COMMUNICATIONS INC.

Key findings and conclusions:

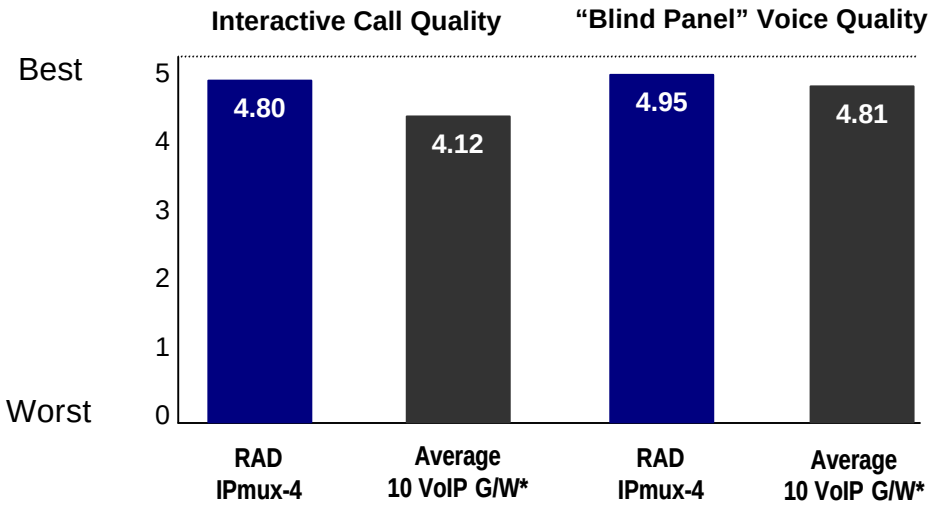
- **Outstanding voice quality in both interactive call quality and "blind" panel voice quality tests**
- **An extremely cost-effective alternative to VoIP**
- **Excellent call-completion rate of 99.996 percent**
- **Transparently extends T1/E1 circuits over IP for voice, video and data applications**

RAD Data Communications' IPmux-4 E1/T1 Multiplexer for IP Networks was lab-tested by MIER Communications using a methodology and test-bed that were developed especially for evaluating VoIP gateway products. The IPmux-4, a new classification of product best described as a TDM-over-IP system, is a VoIP-gateway alternative for voice/data convergence or leased-line services over IP. IPmux-4s—one at each end of a point-to-point link—translate T1 signals into IP packets that can be transmitted over an IP network and then decoded back into a T1 signal by an IPmux-4 at the opposite end of the IP link. Significantly, information transmitted over the link can be voice, data or video, allowing true convergence at a very competitive price (only about $70 per channel). The IPmux-4 delivered superb performance in all phases of testing, is easy to install and configure and highly reliable.

Performance

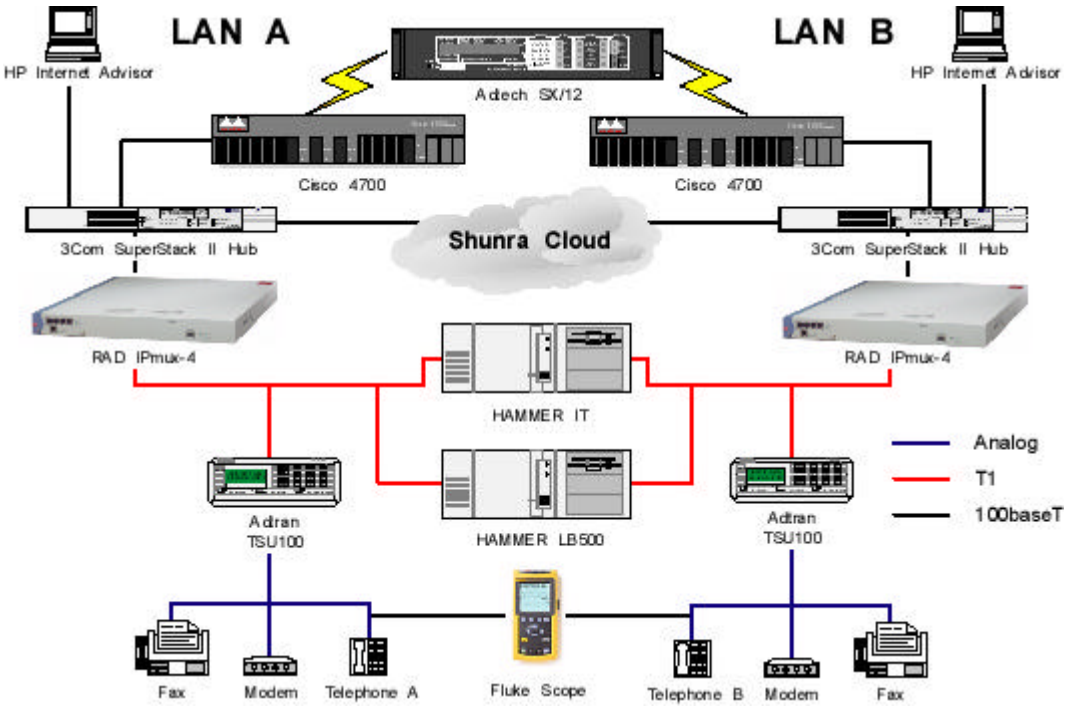
Because the IPmux-4 does not have to compress voice nor use silence suppression—both of which add overhead—it offers superior voice quality and latency over VoIP gateways. How superior? As we show in

Comparison of Voice Quality: Mean Opinion Score
MOS Ratings of RAD TDM/IP vs. VoIP Gateways



**Average rating of 10 VoIP gateways tested recently at MIERComms.*
All VoIP gateways were tested using G.711 encoding.

Test-bed set up



About the testing…Voices of male and female speakers were used as computer sound files (.wav/pcm). These were pre-recorded files covering much of the 3000-Hz telephone bandwidth. These files were played out through a computer fax/modem card (US Robotics 56 Kbps) and delivered, via simulated analog line, into one of the Adtran T1 multiplexers, which converts the signal to a 64-Kbps DS-0 channel that is then delivered to the first RAD IPmux-4 gateway. The RAD IPmux-4 encodes and packetizes the voice signal, and sends it through a hub to a Cisco 4700 router. A serial WAN link out of the router passes through the Adtech SX/12 to the other Cisco 4700 router and into the second RAD IPmux-4 gateway that de-packetizes the voice and delivers it over a DS-0 channel to another Adtran T1 multiplexer, which converts it to an analog signal that is sent out to another computer equipped with the same voice/fax modem card. At this computer, the played-back voice-sample files are re-recorded. The original male and female voice files, sent through back-to-back T1 multiplexers with no VoIP processing, were recorded and became our reference voice recordings. These were played to our panels of listeners, representing the "5" on the 1-to-5 voice-quality scale. Through the Hammer IT VoIP Test System, we generated VoIP traffic loads, which we delivered to the RAD IPmux-4s; measured voice quality as part of a loaded VoIP network; performed gateway stability testing; measured latency and call-connect times; and gathered, displayed, and logged statistics on VoIP network traffic. Interactive voice-quality tests were done over standard single-line telephone sets connected directly to the Adtran T1 multiplexers. We used Fluke Corp.'s Industrial ScopeMeter 123 to measure latency. To measure bandwidth consumption voice-packet sizes, we used Hewlett-Packard's Internet Advisor Ethernet network analyzers, which were attached to both LANs. We used Adtech's SX/12 Data Channel Simulator to simulate poor network conditions by setting a fixed profile of error "bursts."

Performance- continued

the chart (page 1), the IPmux-4 scored 4.80 out of a total of 5 points (MOS ratings) for interactive call quality, compared to 4.12—the average interactive call MOS rating obtained by 10 VoIP gateways we've tested recently. Furthermore, the IPmux-4 also beat out the VoIP gateways in "blind" panel voice-quality tests, achieving a MOS of 4.95, compared to 4.81, the average score of 10 VoIP gateways. "Blind" panel scores on all VoIP products MierComms has tested recently have been near the 5.0 "best quality" MOS rating. But it is significant that the IPmux-4, a new product, can achieve a 4.95 this early in its product cycle. The IPmux-4 also soars ahead of VoIP gateways in terms of latency, measured at less that 20 milliseconds (ms), compared to 90 to 110 ms for VoIP gateways tested.

In addition, IPmux-4 demonstrated an excellent 99.996-percent call completion rate. In a test run overnight—for about 12 hours—the IPmux-4 completed 28,539 calls out of 28,540 calls attempted, dropping only one call.

Comparison: RAD Data Communications TDM-over-IP Versus Traditional Voice-over-IP		
	RAD Data Communications TDM-over-IP	**Traditional Voice-over-IP**
Dominant application	Circuit extension over IP; supports traditional PBX and CTI applications, as well as time-division multiplexing over an IP network	Adaptation of classical telephony over an IP transport, including switching, call routing and call control
Vocoder(s)	64-Kbps standard PCM digitization per voice stream; does not use traditional "G.711" packet structure for 64-Kbps VoIP encoding	G.711 (64-Kbps encoding) and either G.729/G.729A (8 Kbps) or G.723.1 (5.3 or 6.3 Kbps) for low-bandwidth encoding now supported by virtually all VoIP gateways
Number voice streams carried per IP packet	1 to 24	1
Supports point-to-point mode through IP network (all channels of a T1 can be routed directly to distant gateway)	Yes	Yes
Requires a comparable system from same vendor at other end of the IP network	Yes; from RAD Data Communications	Yes; for the most part (due to the still unsettled VoIP standards situation)
IP bandwidth required per voice channel	74 to 82 kbps is typical, based on 2- and 1-ms voice samples, respectively, per channel per packet	70 to 100 Kbps for G.711; when using compression, 25 to 30 Kbps is typical, based on 20-ms voice samples per packet and no VAD (for G.723.1 or G.729 encoding);
Supports proprietary PBX signaling transparently	Yes	No
Supports all data protocols and signaling, in addition to voice	Yes; including H.320 video, ATM, Frame Relay and TDM multiplexing over IP	No
Typical latency (per Mier Labs standard interactive environment; simulates 2 routers and 1000-mile link)	Tested at under 20 milliseconds	90 ms to 110 ms is typical one-way latency
Cost per channel (per end, based on 4 T1 configuration)	$70 per channel	Typically $400 to $500 per channel

Installation/Configuration/Ease of Use

Configuration of the IPmux-4 is through a straightforward, menu-driven command-line interface (CLI) that stepped us through a configuration sequence in which we set jitter buffer and data-packing parameters. We completed the set up in less than 5 minutes. But some features were pre-configured for us, and we estimate that a "ground up" configuration would take from 15 to 30 minutes to enter IP addresses and configure the gateways and the hops. Compared to the typical set up time of at least an hour (and usually much more) for a VoIP gateway, the IPmux-4 set up was a breeze. Documentation is clear and thorough.

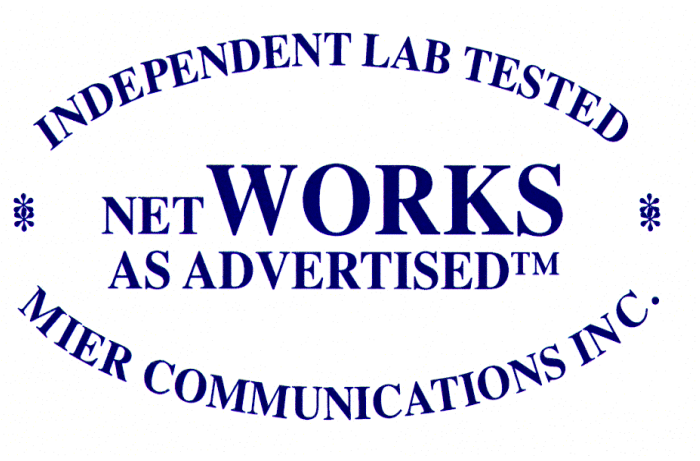
Features

The IPmux-4 supports a number of notable features, which include the following:

- Capacity up to 4 T1s/E1s; 16 planned
- SS7, ISDN-PRI, C7, R2/MFC, DTMF, and RBS/CAS signaling
- Modem and fax support (including V.90 and V.34)
- 10/100 Ethernet LAN support
- 802.1p/q (L2); IP TOS (L3) QoS facilities
- Redundant power supply

Meets Expectations

RAD Data Communications' IPmux-4, a new TDM-over-IP system, offered outstanding voice quality in a series of tests performed recently at the MIERComms labs. Combine top-notch voice quality with the IPmux-4's low latency (under 20 ms) and 99.996-percent call-completion rate, and this product proves a winner on all fronts. This includes a quick and straightforward set up and ease of use. At about $70 per channel, the IPmux-4 is a very cost-effective alternative to VoIP gateways.



In the unanimous opinion of the testers, RAD Data Communications' IPmux-4 E1/T1 multiplexer for IP networks fully meets the expectations and requirements of the target user community for which it was designed and is hereby presented the "Net**WORKS** As Advertised" award.

RAD Data Communications
IPmux-4

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About MIER Communications Product Testing Services…

With hundreds of its product-comparison analyses published over the years in such leading network trade periodicals as *Business Communications Review*, *Network World, and Internet Week*, MIER Communications' reputation as the leading, independent product test center is unquestioned. MIERComms is also a proud member of the Network World Test Alliance. Founded in 1988 by Edwin E. Mier, formerly managing editor of *Data Communications* magazine and a practicing network consultant for over 20 years, the company has pioneered the comparative assessment of networking hardware and software, having developed methodologies for testing products from ATM switches to network operating systems. MIER Communications' private test services include competitive product analyses, as well as individual product evaluations. Products submitted for review are typically evaluated under the "Net**WORKS** As Advertised" program, in which networking-related products must endure a comprehensive, independent assessment of the products' usability and performance. Products that meet the appropriate criteria and performance levels receive the "Net**WORKS** As Advertised" award and MIERComms Labs' testimonial endorsement.

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