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TMC LABS

MVP810-8-Port VoIP Gateway

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INTERNET TELEPHONY® EDITORS' CHOICE AWARD 2004

RATINGS (0–5)

Installation: 5
Documentation: 4.75
Features: 4.75
GUI: 5
Overall: A+

There is always the need to drive down telephony-related costs for call centers and service providers, who have higher call volumes (and phone bills) than your average business. VoIP has primarily changed the communications landscape… indeed not just for service providers and call centers but for us all. You could belong to any category a supplier, end user, networking equipment manufacturer or provider of next-generation equipment or even a service provider who markets new services to end users in businesses or residential homes and we wholeheartedly recommend VoIP as a tool to do business cost effectively.

VoIP has come a long way from the proprietary phone books to standard implementations with a number of enhancements that has helped VoIP technology make inroads into the corporate mainstream. Worth mentioning are: Improved voice quality (true toll quality gateways), products that are more reliable and scalable (Alternate routing PSTN fail over), Quality of Service (QoS), etc.

VoIP has marched forward because it can provide you with many benefits ranging from more cost-effective traditional voice and fax service to new world VoIP-based services and applications like Web-enabled call centers, collaborative whiteboard, remote telecommuter, and personal productivity applications ("follow-me" services, unified message handling).

These innovations are helping us to harness the dormant power of converged voice, fax, and data communications. VoIP gateways have trudged along by virtue of low cost, ease of use, and good performance coupled with reliability. The only prerequisite being a decent Internet or intranet connection and with the Internet becoming better in terms of quality and reliability, this bodes well for VoIP. The VoIP gateway may soon become a commodity — a true plug and play appliance. The key enabler being cost as customers get to keep their PBX, routers and all other equipment, simply add the gateways into the network and the cost savings meter starts ticking. The advent of the H.323 and SIP standards has laid a good foundation that provides interoperability among many vendors and VoIP solutions.

As we already know, H.323 and especially SIP will be the cornerstones for the transmission of real-time audio, video, and data communications over packet networks. In most cases as your voice infrastructure evolves and transitions from a traditional PBX, to a hybrid IP/TDM, IP PBX, IP Centrex or a fully converged network or even the connection to an IP service provider, the media gateway can keep you company by playing different roles.

The VoIP media gateways first performed trunking between traditional PBXs, and that was the norm, but not any longer. Nowadays, gateways play a multifaceted role within your organization and infrastructure taking on many roles and guises. For instance, real-time faxing between sites over an IP network, or your road warriors using a notebook equipped with a VoIP client to make IP calls to your headquarters, and to make off-network calls through the headquarters PBX.

As an added bonus, media gateways will also support H.450 supplementary services (to provide for call transfer, call forwarding, call hold, call waiting and name identification), T.38 real-time fax relay, voice prioritization using the industry-standard protocols etc.

In some ways it is a miniature PBX, of course not boasting a big feature list of 100+ features but in the same breath does admirably well helping users to place and receive calls and get access to voicemail as the minimum. As IP contact centers come in vogue the gateways will have a definite role to play. All agents attach to an IP network and all incoming calls from the PSTN can go through the gateway, can be combined with any incoming IP voice traffic, and can be routed to a central call center or even to remote call center agents. In short, one connection is enough for every type of contact, be it a regular telephone call or an IP call. It also means that you can better integrate the phone with the computer for screen pop-ups.

We investigated the MultiVoIP family, which is available in various flavors from analog to digital models ranging from one to 60 ports. These media gateways connect directly to phones, fax machines, PSTN lines, or a PBX to provide real-time, toll-quality voice connections to any office on a VoIP network. They are designed to help maximize investments already made in the existing data and voice network infrastructure.

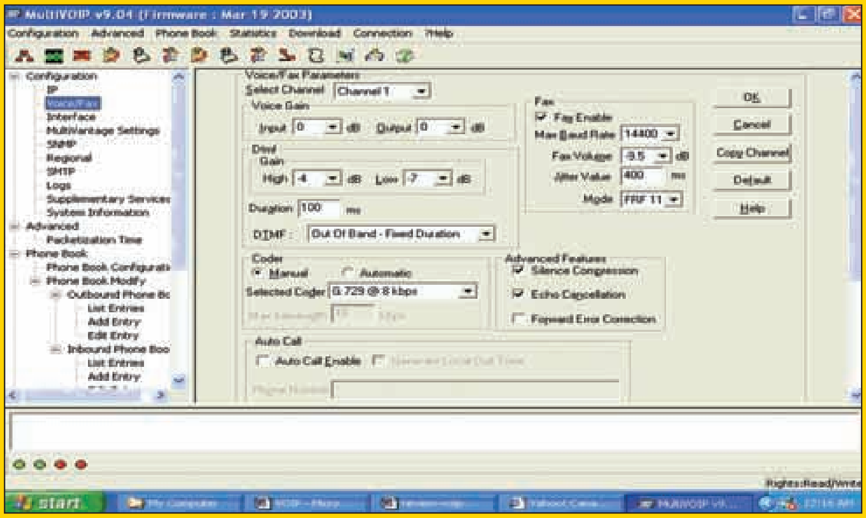

Figure 1: VoIP voice/fax sample setup screen

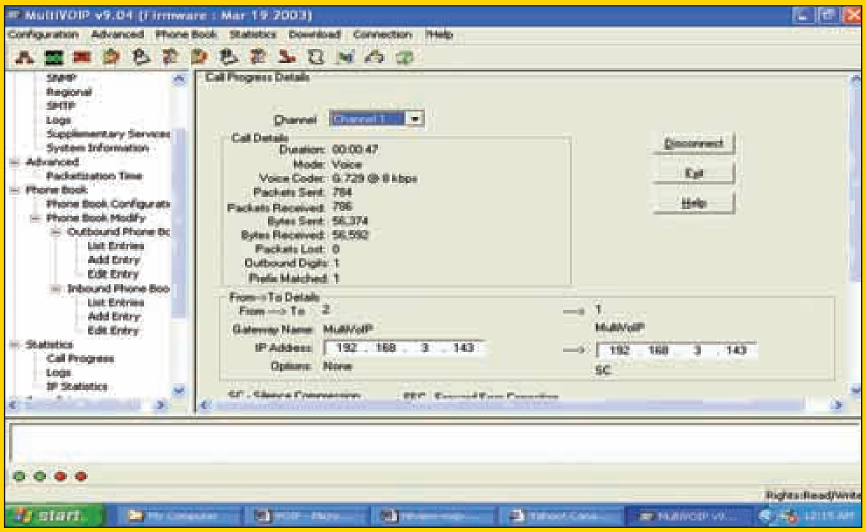

Figure 2: VoIP Gateway sample Call progress screen

Operational Testing

We did a test drive of the MVP800 unit to see how it performs on a real network. Our objective was to extend voice connections from the existing PBX to remote sites on our enterprise data network. There is no better way to check it out than take it out on a spin on our existing data highway — our enterprise network — and if you may, go burn some rubber. The routers, Ethernet switches, the LAN, WAN etc., were all in place, so we just dropped the unit into our live network. For our testing we used the MVP 800 at the main site and for the remote we used the MVP 400.We primarily extended a NEC PBX line to the remote site and along with some phones on both gateways using the FXS ports. One good thing about the MVP800 gateways is that FXO/FXS/E&M ports are standard equipment unlike many other vendor boxes.

The GUI application, which comes with the VoIP gateways, is functionally useful. We programmed the required site based IP addresses for the units, voice/fax parameters including coder selection, silence compression, echo cancellation, forward error correction and more (Figure 1).

Moving on, we set up the interface depending on whether we needed FXO or FXS based on whether you are going to hook up a phone or you are going to connect a line from the PSTN or you're your office PBX. Then swiftly we moved on to the phone book here we set up the entries for the inbound and outbound phonebooks.

Primarily that was enough for us to get started. Under 'statistics' we used the call progress screen and found it to be very useful to get a glimpse of what is happening to the calls being setup (Figure 2). We found the log screens very handy giving time, duration, status, mode, from, to and the like. We made test calls and the voice quality was very good. We made calls between the phones at both sites on both the gateways, we made calls to other NEC PBX users and finally we made calls out to the PSTN switch.

One important feature of notes is the "Alternate Routing" function, which facilitates PSTN Failover protection, allowing you to re-route VoIP calls automatically over the PSTN if the VoIP system fails. The MultiVoIP can be programmed to respond to excessive delays in the transmission of voice packets, which the MultiVoIP interprets as a failure of the IP network.

Room For Improvement

We laud the unit's PSTN failover capability, however under such circumstances it would have been good if the unit sent out an alert to the system administrator that the IP network is congested or down.

Conclusion

This media gateway MVP810 supports FXO, FXS, and E&M all in one compact box with support for H.323v.4, SIP, or SPP (proprietary Multi-Tech protocol), silence suppression, VAD, voice compression, Voice Quality via DiffServ, forward error correction dynamic jitter buffers in totality with management via Web browser or MultiVoIP Manager drove us to conclude that the unit is a very good buy for those looking to save money on long-distance charges. No doubt the return on investment will pay for itself very quickly the higher your call volume.