Ingate SIParator® 90

The Ingate SIParator® 90 is a powerful tool that offers large enterprises a controlled and secured migration to VoIP (Voice over IP) and other live communications, based on Session Initiation Protocol (SIP). With the SIParator, even the largest of businesses, with branch offices around the world and remote workers, can easily harness the productivity and cost-saving benefits of VoIP and other IP-based communications while maintaining current investments in security technology.
Available only from Ingate® Systems, the SIParator works seamlessly with your existing firewall to allow the flow of SIP traffic to reach the user in the enterprise, no matter where he or she is located, as long as there is an Internet connection.

While traditional firewalls are not SIP-capable and, therefore, block SIP traffic – including mission-critical applications like VoIP – the SIParator resolves this problem, and enables SIP-based communication to traverse the firewall, while working in tandem with your current security solutions.

Ingate SIParators solve the Network Address Translation (NAT) traversal issues inherent in SIP communications, and offer both far- and near-end NAT traversal to extend the SIP capabilities within the corporate network to remote workers. With Ingate products, enterprises can use VoIP and other live communications on the LAN and globally over the Internet or private IP networks.

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The Ingate SIParator 90 has eight interfaces, two of which are mini Gbic that can be used fiber optic interfaces for greater flexibility. The SIParator 90 is capable of handling 1200 concurrent VoIP calls (e.g. RTP-sessions), a critical capability for large enterprises.

All Ingate SIParators are fully featured, supporting stateful inspection and packet filtering with rules defined and maintained by the network security administrator utilizing the GUI.

Trusted Security for VoIP, IP Applications

The Ingate SIParator's SIP proxy architecture grants fully secure traversal of the SIP traffic. The ports for the media streams are only opened between the specific parties of a call and only for the duration of the call. The SIP proxy inspects the SIP packets before sending them on. TLS and SRTP encryption of traffic (both signaling and media) secure privacy when communicating, making call eavesdropping, call hijacking and call spoofing harder to do. Ingate also supports authentication of users and servers.

Choose the Right Features for Your Network

Ingate offers several add-on software modules that allow you to tailor the SIParators to meet the specific demands of your business. Ingate Quality of Service (QoS) sets priorities to different kinds of data and allocates bandwidth for varied purposes – for instance, giving priority to VoIP.

Ingate Remote SIP Connectivity extends the SIP capabilities of the enterprise to employees working remotely (home office workers, road warriors, etc.). Remote SIP Connectivity manages the traversal of the remote NAT from a central firewall and also includes a STUN server.

Ingate VoIP Survival adds a whole new dimension to hosted VoIP service by securing full redundancy in a SIP-based hosted IP-PBX environment all the way out to the customer premises. It serves as a backup to enhance the reliability and availability of a VoIP application platform.

Ingate Advanced SIP Routing provides the ability to fully control and route SIP traffic in an advanced and flexible manner. The module can handle least cost routing, enabling enterprises to make global calls for local fees.

Ingate SIP Trunking module allows enterprises to connect their IP-PBXs to a service providers SIP Trunk.

Support for SIP Trunking

More and more Internet Service Providers offer a SIP trunk - a combined Internet and voice connection. For enterprises with an IP-PBX this is an deal cost-saving solution as they no longer need local PSTN gateways or costly PRIs/BRIs. However, the SIP traffic, as all other data traffic, needs to traverse the enterprise firewall. Ingate SIParator 90 handles the firewall and NAT traversal using the built-in SIP proxy.

Add Global VoIP Connectivity to your IP-PBX

The SIParator 90 opens up a world of possibilities and cost savings when used with a SIP based IP-PBX. Businesses can route telephone calls via IP, not only between branch offices and home workers, but also to offices and other users using SIP-based Internet telephony. No longer limited to telephony voice, communication can also include video, instant messaging, presence and more.

In addition, the SIParator 90 makes it possible for home workers, road warriors and even branch offices to belong the same central IP-PBX -- with the highest level of security. The 90 also affords the possibility to set up a private VoIP network, if preferred. Advanced IP-PBX functions are supported, including such as call transfer, call hold, and voicemail.

Do More with Microsoft LCS 2005

Adding value to Microsoft Live Communications Server (LCS) 2005, Ingate SIParators allow voice and video applications to work outside the LAN. In addition, Ingate makes LCS available to mobile users through Remote SIP Connectivity. The SIParator also has upport for Microsoft encrypted media/Microsoft RTP (Realtime Transport Protocol).

Free Software Upgrades for the First Year

The Ingate SIParator 90 has no data user restrictions. Software upgrades are free for the first year. Thereafter, an annual licensing fee will apply. New software versions can be downloaded quickly and easily online from the Ingate website.

For more information, visit us at www. ingate.com or write to info@ingate.com.

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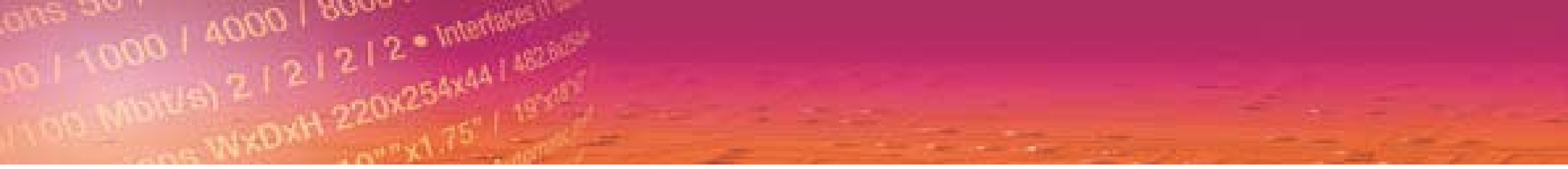
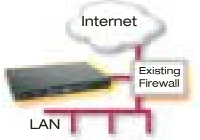
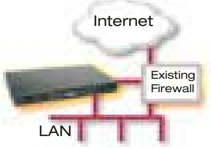


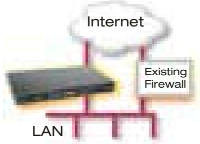
Configuration 1: DMZ



Configuration 2: DMZ/LAN



Configuration 3: Standalone



Configuration 1: DMZ

The Ingate SIParator connects to the existing firewall through the DMZ interface. All traffic will pass through the existing firewall. This configuration requires that a static range of UDP and TCP ports are opened between the Internet and the SIParator and between the SIParator and the LAN. SIP clients on the LAN need to have the SIParator defined as their outgoing proxy or be referred to it *via* DNS. The firewall continues to control security, but SIP traffic is routed to the LAN only through the SIParator.

Configuration 2: DMZ/LAN

The Ingate SIParator connects to the DMZ of the existing firewall and to the LAN. This means that SIP traffic and media streams only have to pass through the existing firewall once (or not at all for all calls inside the office). A static range of UDP and TCP ports needs to be opened in the firewall between the Internet and the SIParator. SIP clients on the LAN need to have the SIParator defined as their outgoing proxy or be referred to it *via* DNS.

Configuration 3: Standalone

The Ingate SIParator connects to both the LAN and the Internet, operating entirely in parallel with the existing firewall. The SIParator will only handle SIP signaling and media streams; everything else will pass through existing firewall. This setup has no requirements for the existing firewall and requires no configuration changes. SIP clients on the LAN need to have the SIParator defined as their outgoing proxy or be referred to it *via* DNS.

In configuration 2 and 3 the SIParator requires a public IP address.

Technical Specifications Ingate SIParators

Feature	Ingate SIParator 19	Ingate SIParator 50	Ingate SIParator 55	Ingate SIParator 65	Ingate SIParator 90
Interfaces (10/100 Mbit/s)	3	0	0	0	0
Interfaces (10/100/1000 Mbit/s)	0	4	4	4	6
Interfaces SPF (mini Gbic)	0	0	0	0	2
Redundant power supply	No	No	No	No	Yes
Flash disc for system operation	Yes	No	No	No	Yes
Dimension WxDxH (mm)	228x146x44	430x369x44	430x369x44	430x369x44	430x485x88
Certifications	CE, FCC, UL	CE, FCC, UL	CE, FCC, UL	CE, FCC, UL	CE, FCC, UL
Management					
Automatic check for new releases	Yes	Yes	Yes	Yes	Yes
Configuration options: Web GUI (HTTP, HTTPS) and CLI (SSH, serial cable)	Yes	Yes	Yes	Yes	Yes
SNMP	Yes	Yes	Yes	Yes	Yes
Max numbers of VLANs	16	32	64	128	256
Internal log to HD	No	Yes	Yes	Yes	Yes
Logging to PCAP file	Yes	Yes	Yes	Yes	Yes
Syslog	Yes	Yes	Yes	Yes	Yes
E-mail events	Yes	Yes	Yes	Yes	Yes
External RADIUS server authentication for GUI and SIP	Yes	Yes	Yes	Yes	Yes
Support for multiple ISPs	Yes	Yes	Yes	Yes	Yes
Free software upgrades	First year	First year	First year	First year	First year
SIP Functionality					
SIP proxy	Yes	Yes	Yes	Yes	Yes
SIP registrar	Yes	Yes	Yes	Yes	Yes
SIP traffic to private IP addresses (NAT/PAT)	Yes	Yes	Yes	Yes	Yes
SRTP (sdescriptions) and Microsoft encrypted RTP	Yes	Yes	Yes	Yes	Yes
TLS encryption	Yes	Yes	Yes	Yes	Yes
SIP Connection set up (SIP + RTP)	0.15 s	0.15 s	0.15 s	0.15 s	0.15 s
RTP data delay (10 Mbps/100 Mbps) network	0.19/0.08 ms	0.19/0.08 ms	0.19/0.08 ms	0.19/0.08 ms	0.19/0.08 ms
Number of concurrent voice RTP sessions (G.711)	40	150	300	650	1500
Concurrent encrypted voice RTP sessions (both SRTP and TLS)	20	75	150	330	750
Busy hour call attempt	36000	72000	79200	79200	234000
Billing and authentication of SIP users from an external RADIUS	Yes	Yes	Yes	Yes	Yes
SIPconnect compliance	Yes	Yes	Yes	Yes	Yes
Add-on modules					
SIP Trunking (connecting an IP-PBX to an ITSPs SIP-trunk)	Yes	Yes	Yes	Yes	Yes
Remote SIP Connectivity (Far-end NAT-passering incl STUN-server)	Yes	Yes	Yes	Yes	Yes
QoS (bandwidth limitation and prioritization	Yes	Yes	Yes	Yes	Yes
Advanced SIP Routing (flexible routing for SIP traffic)	Yes	Yes	Yes	Yes	Yes
VoIP Survival (VoIP redundance if Internet connection fails)	Yes	Yes	Yes	Yes	Yes

Ingate 20070501

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