

Ingate SIParator[®] 50/55/65



The Ingate SIParators 50, 55 and 65 are powerful tools for businesses wanting to step up to the next level of using VoIP and other IP-based realtime communications, and to do so not only within the company, but outside the enterprise as well. The Ingate SIParator[®] from Ingate[®] Systems works seamlessly with an existing firewall to allow the traversal of SIP traffic through the enterprise edge. While traditional firewalls block SIP traffic – including mission-critical applications like VoIP – the SIParator resolves this problem while working in tandem with your existing security solutions. With the SIParator, businesses can harness the productivity and cost-saving benefits of VoIP and other IP-based communications while maintaining current investments in security technology.

Ingate's SIParators are installed all over the world. Common scenarios: enterprises connecting to Internet telephony service providers (ITSP) and SIP trunks, connecting remote users and branch offices to use the corporate IP-PBX, and many more.

Ingate's SIP proxy-based solution delivers maximum control over SIP signaling, traffic, and network security. With Ingate products, enterprises can use VoIP and other live communications on the LAN as well as globally over the Internet or private IP networks.

Ingate SIParators 50/55/65

The Ingate SIParators 50/55/65 have four ports each. The smallest – the SIParator 50 – can handle up to 150 concurrent RTP sessions. The Ingate SIParator 55 can handle up to 300 concurrent RTP sessions and the Ingate SIParator 65 has a capacity of 650 concurrent RTP sessions. The software upgrade module available for the Ingate SIParator 50 and 55 offers enterprises the flexibility to buy only as much capacity as needed. Capacity can be added at any time. Hardware and the functionality are otherwise identical for all three models.

Included in the Ingate SIParator 50/55 and 65 are five SIP traversal licenses, allowing up to five calls to traverse at the same time. Ten user licenses that can be used for the registration of SIP users on the built-in SIP registrar are included. Additional SIP traversal licenses and user licenses can be purchased at any time.

Ingate SIParators feature an encrypted Virtual Private Network (VPN) termination module. The SIParators can be configured as a part of the DMZ or in a standalone mode. In both cases, the benefits of SIP-based communications can be added to the network quickly and easily.

Trusted Network Security for VoIP

Ingate'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 ensures privacy, making call eavesdropping, call hijacking and call spoofing harder to do. Ingate also supports authentication of users and servers.

Support for SIP Trunking

More and more Internet telephony service providers offer a SIP trunk – a combined Internet and voice connection. For enterprises using an IP-PBX, SIP trunks are an ideal cost-saving solution as they no longer need local PSTN gateways or costly PRIs/BRIs. The service provider provides the PSTN connection. However, in order for SIP trunks to work, SIP traffic (as well as all other data traffic) must be able to traverse the enterprise firewall. Ingate's SIP Trunking software module, available for all Ingate SIParators, enables firewall and NAT traversal using the built-in SIP proxy, allowing the enterprise to connect to the SIP trunk.

In addition, Ingate SIParators and the Ingate SIP proxy deliver advanced security for all SIP communications, including those *via* a SIP trunk. Ingate products also help ease compatibility issues between the IP-PBX and Internet telephony service provider.

Choose the Right Features for Your Network

Ingate offers several add-on software modules that allow you to tailor the SIParator 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 the central Ingate SIParator 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 calls globally for local fees.

Global VoIP Connectivity for your IP-PBX

Ingate SIParators open up a world of possibilities and cost savings when used with a SIP-based IP-PBX. Businesses can not only connect to a SIP trunk, but also route telephone calls via IP, between branch offices, home workers, offices and others using SIP-based VoIP. With an Ingate SIParator, the enterprise is no longer limited to voice; communication can also include video, instant messaging, presence and more.

Free Software Upgrades for the First Year

Software upgrades for the Ingate SIParators 50/55/65 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.

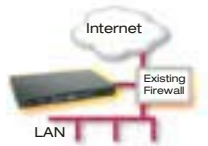
For more information, visit us at www.ingate.com or write to info@ingate.com.

inGate[®]

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To learn more about our SIP Firewall and VoIP range freecall 1800 817 807 or visit www.voip.alloy.com.au

Configuration 1: DMZ



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.

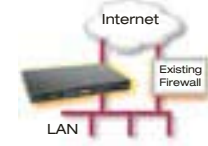
Configuration 2: DMZ/LAN



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



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 SFP (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-passing 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 redundancy if Internet connection fails) | Yes | Yes | Yes | Yes | Yes |

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