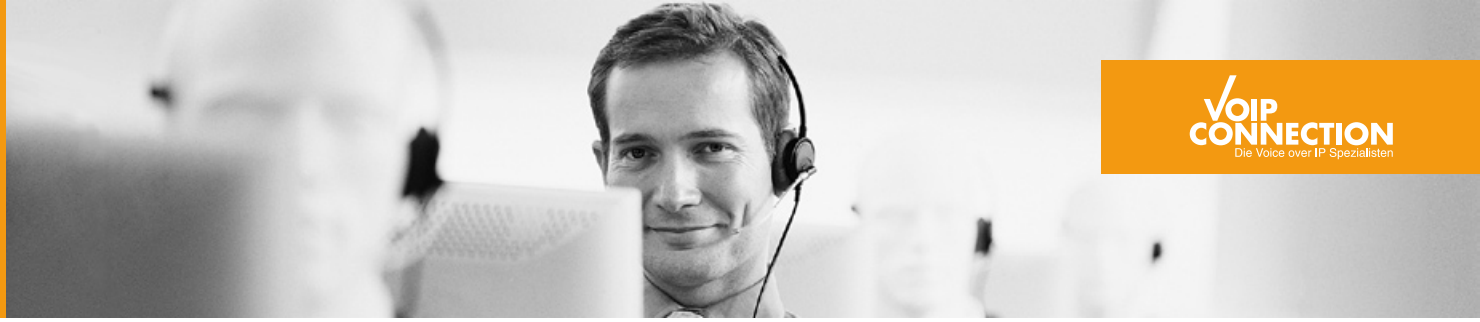

Ingate SIParator® 50/55/65

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 offi ces to use the corporate IP-PBX, and many more. **Ingate's SIP** proxy-based solution delivers maximum control over SIP signaling, traffi c, 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. Additional SIP traversal 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 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 specifi c 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 spoofi ng 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 other add-on software modules that allow you to tailor the SIParator 19 to meet the specifi c 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 offi ce workers, road warriors, etc.). Remote SIP Connectivity manages the traversal of the remote NAT from a central fi rewall 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.

Ingate Enhanced Security Module provides Intrusion Detection and Intrusion Prevention for SIP as well as encryption of the communication. The SIP Registrar Module allows for making the Ingate Registrar the primary registration server.

GLOBAL VOIP CONNECTIVITY FOR YOUR IP-PBX

The SIParators 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 offi ces and home workers, but also to offi ces and other users using SIP-based Internet telephony. No longer limited to telephony voice, communication can also include video, instant messaging, presence and more.

FREE SOFTWARE UPGRADES FOR THE FIRST YEAR

The Ingate Firewalls have 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.

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.

Configuration 1: DMZ

Configuration 2: DMZ/LAN

Configuration 3: Standalone

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

TECHNICAL SPECIFICATIONS INGATE SIPARATORS	90	50	55	65	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 disk for system operation	yes	no	no	no	yes
Dimension W x D x H (mm)	228 x 146 x 44	430 x 369 x 44	430 x 369 x 44	430 x 369 x 44	430 x 485 x 88
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	Firstyear	Firstyear	Firstyear	Firstyear	Firstyear
SIP functionality					
SIP proxy	yes	yes	yes	yes	yes
SIP registrar	yes	yes	yes	yes	yes
SIP traffic to private IP adresses (NAT/PAT)	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 consurrent encrypted 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
Enhanced Security (IDS/IPS for SIP, SRTP and TLS)	yes	yes	yes	yes	yes
SIP Registrar (Ingate is used as the primary SIP registrar)					
VoIP survival (VoIP redundance internet conncetion fails)	yes	yes	yes	yes	yes

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