snom 4S SIP NAT Gateway
Datasheet Version 2.0

4S

Key Features

- **Application Layer Gateway**
- **RFC 3261 and 2543 Compliant**
- **Includes NAT Traversal**
- **Stateless Proxy**
- **Support of Default SIP Server**
- **Media Forwarding**
- **Optional Transcoding**
- **Available as Library and in Full Source Code**
- **Available for Microsoft Windows and Linux**

snom VoIP phones

Application Area

Firewalls are one of the most critical problems for SIP in the enterprise today.

Most companies are using firewalls to protect their Internet connectivity and to share a common IP address. With the increasing importance of SIP as real time communication protocol, the demand for SIP enabled firewalls is rising rapidly.

Unfortunalety, SIP telephony assumes that clients may contact servers directly with an IP address. RTP media packets are supposed to be sent directy from the source to the destination.

The SIP NAT Gateway solves this problem by filtering SIP packets and forwarding RTP media according to the SIP media negotiations.

By watching traffic on port 5060 even packets that are not addressed to the application gateway can be forwarded properly.

Approaches

There are several approaches to solve this problem without an application layer gateway. For home usage, STUN (simple traversal of UDP through NAT) can turn standard DSL routers into SIP-capable equipment. UPnP also can be used to control the NAT gateway.

However, most of the SIP equipement is not able to support NAT traversal techniques. The results are half-way solutions where the others party phone rings, but they are not able to exchange media.

Integrating a SIP application layer gateway enables all RFC 3261 and 2543 compliant devices to make calls over the Internet and through VPN. Offering the gateway in a firewall avoids the effort to install a seperate solution parallel to a firewall.

Features

The SIP NAT gateway is available as stand alone application as well as library intended for integration into firewall solutions. The standalone application can be run in parallel to a firewall.

The gateway is compatible with the RFC 3261 and supports UDP and TCP transport layer. It features both loose and strict routing and supports the SIP DNS extensions for locating SIP servers.

The optional transcoding module uses the processor on the firewall to reduce the bandwidth of the media lines. This is important in cases when the firewall is connected to a DSL line and the attached endpoints are not able to do media compression. Supported low rate codecs are GSM 6.10 and G.729A. Other media types are supported on a flow-through basis.

AUthentication is done on the SIP application layer. The gateway controls which ports are associated with which media connections and controls the traffic that flows through these ports. When the applications stop the media, the ports are closed.

NAT traversal is done by translating private and public addresses in both SIP headers and SDP bodies. This is done in accordance with the opening and closing of RTP ports.

When the destination cannot be resolved properly, the gateway supports a default destination. This can be a company proxy or media server which handles all traffic to unspecified destinations. This could be an auto attendant which will prompt the caller for the final destination.

Compatibility

The compatibility of the application layer gateway is ensured by strict following or the RFC 3261 guidelines. The functionality has been tested with equipment from Microsoft, Cisco and other major vendors in the SIP area.

The application layer gateway modified only SDP attachments and adds routing elements into the SIP header. This maximizes interoperability and makes sure that future applications will be able to flow througth the proxy without problems.

The proxy supports the Path header. This way, user agents can register with a route and not only a contact. If the proxy does not support the Path header method, escaped headers can be used to encode the final destination of a request.

About snom

snom technology specialized in VoIP telephony. Its voice over IP phone snom 100 is able to handle both H.323 and SIP traffic. The snom 4S solution framework includes a SIP proxy and a SIP media server as wells as tools like a STUN server and CDR generation. The solution fulfills the requirements of a small to medium scale office and is a solution to medium size internet providers. snom was founded in 1996 and is located in Berlin, Germany.

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