Quality of Service (QoS) refers to the capability of a network to provide better service to selected network traffic. By prioritising audio and video, TCP (email, FTP, etc.) and other flow-controlled traffic will back off and only use the bandwidth not consumed by the real time traffic.

To support QoS a device needs to implement different mechanisms, algorithms, ensuring that for example VoIP traffic is not disturbed by common data traffic.

To achieve the goals of QoS end-to-end, the entire network including all its components need to support QoS.

The Internet Gate has three mechanisms ensuring Quality of Service for SIP media streams:

- SIP media stream prioritization
- Upstream traffic shaping
- DiffServ bits manipulation

SIP media stream prioritization
By having multiple packet queues with different priorities, SIP media stream packets can bypass ordinary data packets and be transmitted as soon as possible. SIP media stream packets will not become delayed by ordinary data traffic.

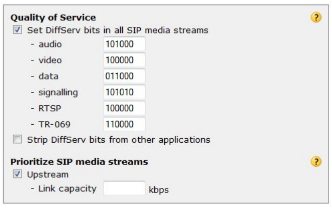
Upstream traffic shaping
By limiting outgoing traffic to the actual bandwidth available (that is often less than the physical bandwidth of the link), a smoother traffic flow is achieved.

DiffServ bits manipulation
By setting the DiffServ bits of SIP media stream packets they can get special attention in your ISP’s network if it supports QoS. This ensures continued priority treatment for those packets on their way through the network.

Performance
Obviously, as QoS prioritizes SIP media stream packets, other packets will be delayed, and get less bandwidth to use.

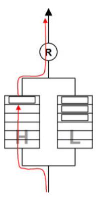
However, as QoS means extra processing for each packet going through the firewall, performance is actually slightly decreased even without any SIP media streams in progress. Therefore you are recommended to turn QoS off if you have no SIP traffic in your network.

The web interface
The QoS web interface is part of the Advanced SIP web page:



The web interface is very easy and intuitive. You can enable the different mechanisms one by one. To activate traffic shaping you need to enter the corresponding link capacity. The advanced QoS functions are then carried out automatically.

QoS block diagram



Upstream
Packets to be transmitted are placed in two different queues:

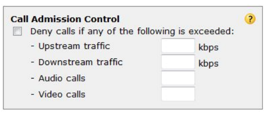
- one High priority queue for SIP media stream packets, and
- one Low priority queue for other, common, data packets.

High priority data packets have priority over low priority data packets – they are transmitted first. This way SIP media stream packets can bypass several common data packets and get transmitted without delays even during high data loads.

The Regulator limits data traffic to the bandwidth specified in the “Upstream Link capacity” field. By limiting data traffic transmit speed to the actual bandwidth available a smoother data traffic is achieved, without the speed fluctuations that would occur otherwise.

Downstream
Achieving QoS for downstream traffic, incoming packets from your ISP, should be a task for your ISP. It is always better to support QoS when sending the packets – it is too late to do anything after you have received the packets. The best way of ensuring downstream QoS is having an ISP supporting QoS.

Call Admission Control



Call Admission Control (CAC) limits the number of SIP media streams allowed through the unit. CAC can deny new SIP connections attempted if the unit is getting overloaded.

If the unit is being overloaded it is better to deny further SIP calls, and thus maintain quality of the ones currently in progress, rather than allowing too many calls that would overload the unit and reduce quality of all calls in progress.

If Call Admission Control is enabled, new SIP calls are allowed only if there is enough bandwidth and processing power left to be able to handle the resulting media streams effectively.

Several different measures can be controlled to detect if there is enough bandwidth and processing power left to be able to handle an attempted SIP call. By entering values into one or several of the fields one can control when further SIP calls should be denied:

Upstream / downstream traffic
If the unit already handles too much traffic further media streams would have too little bandwidth left to use.

Audio / video calls
If the unit already handles a vast amount of media streams it would not have any capacity left to handle further ones.

For example by entering 20 into audio calls the unit will allow up to 20 simultaneous SIP audio calls. If there are already 20 calls in progress, an attempted 21th call will be denied – BUSY is returned to the attemptee.

You can enter values into one, several, or all fields. If any of the values are exceeded further calls are denied. By leaving a field empty that measure is ignored. Once workload is lowered below the limits calls will be allowed again.