

### SmartNode™ Quality of Service for VoIP on the Internet Access Link

#### Applies to the following products

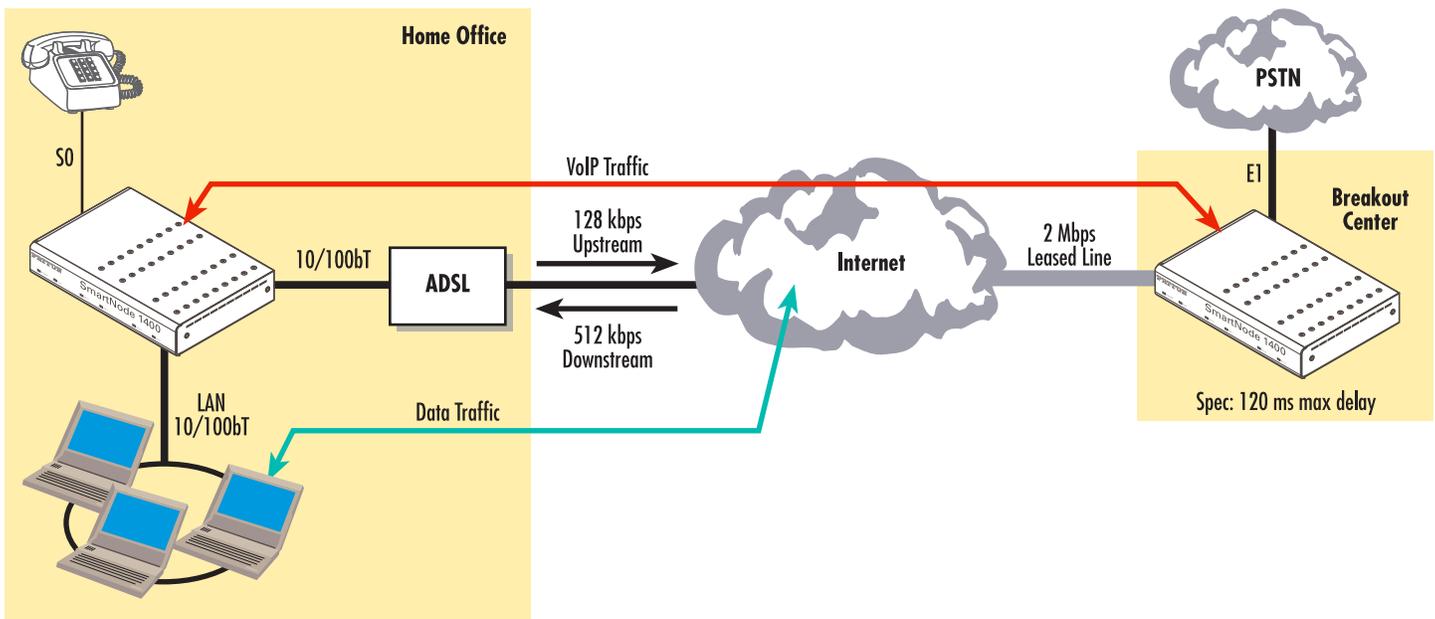
- SmartNode 1000 Series
- SmartNode 2000 Series
- SmartNode 4520 Series

#### Overview

Initially designed for data applications such as email and file transfers, today's Internet does far more. Voice-over-IP (VoIP)—also known as packet voice or packet telephony—has recently emerged as the latest and greatest Internet application. When routing voice through the Internet, ensuring VoIP users perceive the best possible voice quality becomes a most-critical issue.

For a successful VoIP deployment, voice quality must compare favorably with the voice quality that users have come to expect from public switched telephone networks. To achieve comparable voice quality over the Internet (or any IP WAN), packet loss must be prevented and packet delay minimized for all packets carrying digitized voice. To ensure perceived voice quality meets levels users deem acceptable, end-to-end packet delay should not exceed 120ms.

In VoIP systems that traverse the Internet, the bottleneck typically occurs at the access link, the low-bandwidth connection between the high-speed Internet (WAN) and the user network (LAN). Both networks typically run at 100 Mbps or above. A typical access link may easily run about 200 times slower than the LAN residing in the home or office (say, 512 kbps for example). Congestion, queuing delay, and queue overflows (resulting in dropped packets) are most likely to occur on this link.



Overall bandwidth of the access link, maximum packet size, and traffic burst size (the number of packets arriving all at once) are the primary factors that contribute to packet delay and loss. Consider the case of a 256 kbps link from the Internet's edge router to the user's LAN. Suppose the Internet's transmit queue contains five

1500-byte data packets, followed by one time-sensitive voice packet. It will take roughly 270 ms to send those data packets over the 256 kbps link. When the voice packet follows, it arrives with a delay longer than 120 ms (our target), resulting in degraded voice quality.

In systems where a single access link serves both voice and data traffic, special measures must be employed to ensure good voice quality. This technote describes how to configure SmartWare QoS features to ensure the highest possible voice quality.

**Upstream and downstream** traffic present different problems, and are best addressed by different Quality of Service (QoS) mechanisms. (*Upstream* traffic flows from the user to the Internet, while *downstream* traffic flows from the Internet to the user.)

## SmartWare's QoS mechanisms

SmartWare offers the following QoS mechanisms:

- **Upstream**—Shape the flow of upstream voice and data traffic to minimize the queuing delay that voice packets encountered.
- **Downstream QoS™**—Throttle downstream data traffic to minimize the risk of dropped or delayed voice packets resulting from data bursts.
- **Packet Size**—Limit the maximum packet size for all data packets, thus increasing the effectiveness of the mechanisms described above.

Additional considerations affecting voice quality:

- The **CODEC and compression** algorithm you select (in addition to the QoS mechanisms that SmartNode provides) will also substantially affect perceived voice quality.
- The **De-jitter Buffer** configuration must match the maximum packet size for all data packets.
- **ADSL** access links require that you configure additional settings.

## QoS procedures

### 1. Shape the Uplink Traffic to reduce delays in the modem's transmit queue.

To avoid overloading the modem in the Upstream direction, you should route all data traffic through the SmartNode. The following example defines a scheduler profile for use on the outgoing WAN interface. Using these settings, SmartNode will regulate outgoing voice and data traffic, preventing voice packets from having to wait in the modem's transmit queue.

```
profile service-policy Up-Link
  rate-limit 128

  source traffic-class local-voice
  priority

interface wan
  use profile service-policy Up-Link out
```

The parameters in the example function as described below:

- `rate-limit 128`: The **rate-limit** command option specifies the upstream capacity of the Modem in kilobits per second (kbps). In this example the modem's maximum upstream line rate is 128 kbps. SmartWare uses the value of **rate-limit** to determine how much traffic SmartNode can send without filling up the modem's transmit queue. The value of **rate-limit** must equal the actual upstream rate of the Internet access link.

- **local-voice**: Voice packets that originate or terminate at the SmartNode fall within the **local-voice** traffic class.
- **local-default**: All non-voice packets that originate or terminate at the SmartNode fall within the **local-default** traffic class. This class includes VoIP signaling, management traffic (Telnet, TFTP, SNMP etc.), and VPN traffic (IPsec). You should assign priority to local-voice traffic to ensure that SmartNodes delivers signaling traffic reliably and without delay.
- **VPN**: If VPN connections originate or terminate on this SmartNode You must **not** give priority to local-default traffic.

**Note** Packets not classified as **local-voice** nor **local-default** will fall within the **default** traffic class, and will be routed through the SmartNode to other destinations. Consult the SmartWare Configuration Guide for more information on this topic.

**Note** If VPN connections originate or terminate on this SmartNode You must **not** give priority to local-default traffic.

- **priority**: The **priority** command option tells the link scheduling mechanism to transmit packets in the specified traffic classes (**local-voice** and **local-default** in this example) before transmitting packets in other traffic classes (i.e. data packets). SmartNode will schedule packets in other classes for transmission within the remaining available bandwidth on the access link (up to the maximum transmit bandwidth defined by rate-limit).

## 2. Configure DownStreamQoS™ to reduce delayed or lost voice packets on the downstream link.

During phone calls the SmartNode throttles back downstream data traffic on the access link, minimizing the risk of lost or delayed voice packets coming from the Internet or WAN.

The example below defines a scheduler profile, which SmartNode applies to incoming traffic on the WAN interface.

```
profile service-policy Down-Link
  rate-limit 768 voice-margin 200

  source traffic-class local-voice
    priority

  source traffic-class default
    queue-limit 4

interface WAN
  use profile service-policy Down-Link in
```

The command parameters in the above example function as described below:

- **voice-margin 200**: When priority traffic (**local-voice** packets in this example) is present, SmartWare reduces the maximum bandwidth for incoming traffic by the specified bit rate in kbps (200 kbps in this example). The throttling effectively reduces the average rate of incoming data packets, which frees bandwidth for priority traffic thus improving voice quality. For best results, use a generous value for voice-margin to accommodate fluctuations in traffic flow. 30% of the total bandwidth is recommended, up to max 200 kbps. Note that SmartWare only reserves this margin when priority traffic (voice packets) is present (i.e. when the phone is in use). If no calls are active, the full downstream bandwidth is available for data traffic.

- `queue-limit 4`: This setting tells the SmartWare how many packets to enqueue before dropping excess traffic (data packets). A dropped packet wastes bandwidth but it also tells the sender to slow down (reduce its transmission rate) because the link is overloaded. The default queue size in SmartWare is 16. By reducing the queue limit, SmartWare throttles data more effectively, further improving voice quality.

### 3. Choose the preferred CODEC and compression algorithm

As well as selecting a compression algorithm, you can define how much speech time (in milliseconds) the CODEC will record in each packet. Defining longer speech durations increases packet size and reduces overhead, thus reducing the bandwidth required for transmission (good). However, it also increases queuing delay (bad). The example below shows how to define the preferred CODEC and packet length.

```
profile voip myProfile
  codec 1 g729 tx-length 30 rx-length 30
  fax transmission 1 relay t38-udp
  fax redundancy ls 1 hs 1

interface sip MySipIf
  use profile voip myProfile
```

The command parameters in the above example function as described below:

- `codec 1 g729`: These parameter values defines G.729 as the preferred CODEC and compression rate. In actual operation, SmartWare may or may not employ the preferred CODEC for every call. During call-setup, the VoIP devices at either endpoint will negotiate and select the CODEC employed for that particular call.
- `tx-length 30`: These parameter values instruct SmartWare to use the indicated CODEC to record digitized speech at the rate of one packet every 30 ms.
- `fax transmission 1 relay t38-udp`: These parameter values enable fax relay using the protocol T.38.
- `fax redundancy ls 1 hs 1`: These parameter values enable SmartNode to insert one redundant fax packet into each IP packet.

**Note** It is important that you choose a CODEC with a compression rate that can accommodate the required number of concurrent calls within the available bandwidth. At the same time, you must also consider that greater compression will reduce voice quality, as will longer packet duration, with longer packets having the greater negative impact. The following tables list the bandwidth required for one voice call using the indicated CODEC at the specified packet durations.

Bandwidth required per VoIP or FAXoIP call using differing CODECs at various packet-durations or fax-transmission-rates

<b>Cable Networks (Ethernet over DOCSIS)</b>								
<b>CODEC</b>	<b>Net Rate</b>	<b>10ms</b>	<b>20ms</b>	<b>30ms</b>	<b>40ms</b>	<b>50ms</b>	<b>60ms</b>	<b>70ms</b>
G.711	64 kbps	110 kbps	87 kbps	80 kbps	76 kbps	73 kbps	72 kbps	71 kbps
G.723	6.4 kbps			22 kbps			14 kbps	
G.726	32 kbps	78 kbps	55 kbps	48 kbps	44 kbps	41 kbps	40 kbps	39 kbps
G.729	8 kbps	54 kbps	31 kbps	24 kbps	20 kbps	17 kbps	16 kbps	15 kbps

<b>CODEC</b>	<b>Redundancy</b>	<b>2.4 kbps</b>	<b>4.8 kbps</b>	<b>7.2 kbps</b>	<b>9.6 kbps</b>	<b>12 kbps</b>	<b>14.4 kbps</b>
T.38	0	14 kbps	16 kbps	18 kbps	21 kbps	23 kbps	26 kbps
T.38	1	17 kbps	22 kbps	27 kbps	32 kbps	36 kbps	41 kbps
T.38	2	21 kbps	28 kbps	35 kbps	42 kbps	50 kbps	57 kbps

<b>ADSL Networks (PPPoE / PPP over ATM AAL-5)</b>								
<b>CODEC</b>	<b>Net Rate</b>	<b>10ms</b>	<b>20ms</b>	<b>30ms</b>	<b>40ms</b>	<b>50ms</b>	<b>60ms</b>	<b>70ms</b>
G.711	64	170 kbps	106 kbps	99 kbps	95 kbps	85 kbps	85 kbps	85 kbps
G.723	6.4			42 kbps			21 kbps	
G.726	32	127 kbps	85 kbps	71 kbps	53 kbps	51 kbps	49 kbps	48 kbps
G.729	8	85 kbps	64 kbps	42 kbps	32 kbps	25 kbps	21 kbps	24 kbps

<b>CODEC</b>	<b>Redundancy</b>	<b>2.4 kbps</b>	<b>4.8 kbps</b>	<b>7.2 kbps</b>	<b>9.6 kbps</b>	<b>12 kbps</b>	<b>14.4 kbps</b>
T.38	0	21 kbps	32 kbps	32 kbps	32 kbps	32 kbps	42 kbps
T.38	1	32 kbps	32 kbps	42 kbps	42 kbps	53 kbps	53 kbps
T.38	2	32 kbps	42 kbps	53 kbps	53 kbps	64 kbps	74 kbps

#### 4. Reduce jitter (variations in delay)

SmartWare reduces jitter (variations in delay) by receiving voice packets into a de-jitter buffer. The de-jitter buffer regulates the variations in packet delay so SmartNode can forward them on a consistent schedule to the user. By making voice packets available for decoding and playback at regular intervals (e.g. every 10 ms), the de-jitter buffer improves perceived voice quality. The following commands show how to configure the settings that control the de-jitter buffer.

```
zprofile voip default
  dejitter-max-delay 120
  dejitter-mode static
```

The command parameters in the example function as described below:

- `dejitter-max-delay 120`: This option allows the size of the de-jitter buffer to reach 120 ms, exceeding the default limit of 60 ms.
- `dejitter-mode static`: This option puts the de-jitter buffer into static mode. In static mode the de-jitter buffer seeks to maintain its average contents near the halfway-full mark. In many deployment situations excessively long delays may be rare, but they do occur every once in a while. Yet you may still want the de-jitter buffer to regulate such delays when they do occur. In such cases it may be advisable for you to use the settings shown here (i.e. configuring the de-jitter buffer for static mode with a max delay of 120 ms). This configuration effectively maintains an additional 60 ms of capacity within the de-jitter buffer, sometimes known as a “skip-buffer,” to handle the rare occurrence of an excessively long delay.

**Adaptive De-jitter Buffer.** The de-jitter buffer operates adaptively. If the actual jitter level causes SmartNode to utilize less than the configured capacity of the de-jitter buffer for an extended period of time, the buffer will adapt to the lower jitter level by reducing in size. If jitter increases suddenly, the de-jitter buffer will adapt again by increasing in size. During this brief adjustment period (while the buffer is enlarging), the user will hear a little “blip” (a short drop-out in the incoming voice). Because of the nature of downlink scheduling, long delays, though rare, are quite likely to occur from time to time. So for most situations configuring the de-jitter buffer in static mode is a good idea.

Finally, the degree of jitter will vary based on how you configure *all the settings discussed in this tech note*. The table below presents an estimate of the minimum and maximum delays for a sample configuration.

## 5. Reduce the maximum data packet size for access links with a small bandwidth

On a slow link, one large data packet will occupy the link for a long time, delaying the voice packets that follow. You can significantly reduce jitter (variations in packet delay) by limiting the packet size. Reducing the packet size will also improve the effectiveness of the downlink scheduling mechanism. The drawback of reduced packet size is that it requires transmitted data to be divided into a larger number of packets, which in turn increases overhead. The following commands show how to set up the packet-size limitation

The network and the down-link at the peer also contribute to the overall jitter. The dejitter buffer as configured in the last step can compensate 60 ms of overall jitter. We will apply the following jitter budget:

- Up-link jitter: 40 ms
- Network jitter: 0 ms
- Down-link jitter: 20 ms

We can control the up-link and the down-link jitter by adjusting the size of the IP packets. The following table indicates the MSS (size of the TCP payload) to be set for the up-link depending on the link speed:

Link Type	64 kbps	128 kbps	256 kbps	> 340 kbps
Cable Network	262 Bytes	582 Bytes	1222 Bytes	1440 Bytes
ADSL	210 Bytes	498 Bytes	1074 Bytes	1440 Bytes
Frame Relay	270 Bytes	590 Bytes	1230 Bytes	1440 Bytes

The following table indicates the MSS to be set for the down-link depending on the link speed:

Link Type	256 kbps	512 kbps	> 680 kbps
Cable Network	582 Bytes	1222 Bytes	1440 Bytes
ADSL	498 Bytes	1074 Bytes	1440 Bytes
Frame Relay	590 Bytes	1230 Bytes	1440 Bytes

The MSS shall not be bigger than 1440 bytes to guarantee seamless TCP connections to and through the SmartNode over different types of links. If a VPN connection is present, you must subtract 60 bytes from each of the values shown above.

For an ADSL-link with 768 kbps down-link and 128 kbps up-link speed, enter the following commands:

```
interface WAN
  tcp adjust-mss rx 498
  tcp adjust-mss tx 1440
```

The command parameters in the example function as described below:

- `tcp adjust-mss rx 498`: This set of parameters defines the size of the TCP payload (in bytes) in the up-stream direction (**rx** refers to the direction in which the MSS tag in the TCP SYN is modified)
- `tcp adjust-mss tx 1440`: Configures the size of the TCP payload (in bytes) in the downstream direction (**tx** refers to the direction in which the MSS tag in the TCP SYN is modified)

## 6. Tailor your configuration for ADSL service

By default, SmartWare's rate-limit command assumes the access link will employ Ethernet encapsulation. For other types of encapsulation you must issue command options that inform SmartWare which encapsulation method your modem uses. ADSL services use PPPoE over ATM encapsulation, which increases overhead on the access link. To accommodate this increased overhead, you must include the following settings in the service-policy profile.

```
profile service-policy Up-Link
  rate-limit 128 header-length 30 atm-modem
```

The command parameters in the example function as described below:

- `header-length 30`: This option instructs the rate-limiter to accommodate the additional constant overhead (in bytes) that ADSL service adds to each IP packet. For PPPoE with ADSL the additional overhead is 30 bytes (8 PPPoE, 14 Ethernet, 8 LLC/SNAP).
- `atm-modem`: This option instructs the rate-limiter to accommodate the additional overhead associated with ATM and AAL5.

## **Contacting Patton**

If you have any questions please feel free to contact Patton's Technical Support:

- E-mail support—e-mail sent to **support@patton.com** will be answered within 1 business day
- Telephone support—standard telephone support is available five days a week—from **8:00 am to 5:00 pm EST (1300 to 2200 UTC)**—by calling **+1 (301) 975-1007**

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