QOS CONFIGURATION FOR SRX SERIES FOR THE BRANCH WITH INTEGRATED CONVERGENCE SERVICES

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Introduction

The purpose of this application note is to walk the reader through the steps necessary to configure class of service (CoS) and quality of service (QoS) for voice traffic on Juniper Networks® SRX Series Services Gateways for the branch with Integrated Convergence Services (ICS).

Scope

This paper introduces the Juniper Networks Junos® operating system command-line interface (CLI) and helps the reader configure an SRX Series device with Integrated Convergence Services with QoS and provides a building block for more advanced configurations. It does not include advanced security configuration examples or network design guidelines. Additional Juniper Networks documentation is available for readers at **www.juniper.net/techpubs/software/junos/index.html#srx.**

Design Considerations

Hardware Requirements:

Junos OS Release 10.1 or later for all SRX Series Services Gateways with Integrated Convergence Services.

This includes the following SKU numbers:

- SRX210H-P-M
- SRX240H-P-M

Note: Certain features described in this document, including Integrated Convergence Services, are not available across the entire SRX Series platform. Readers should consult Juniper Networks product-specific literature for more details.

Software Requirements:

Junos OS Release 10.1 or later for all SRX Series with Integrated Convergence Services.

Description and Deployment Scenario

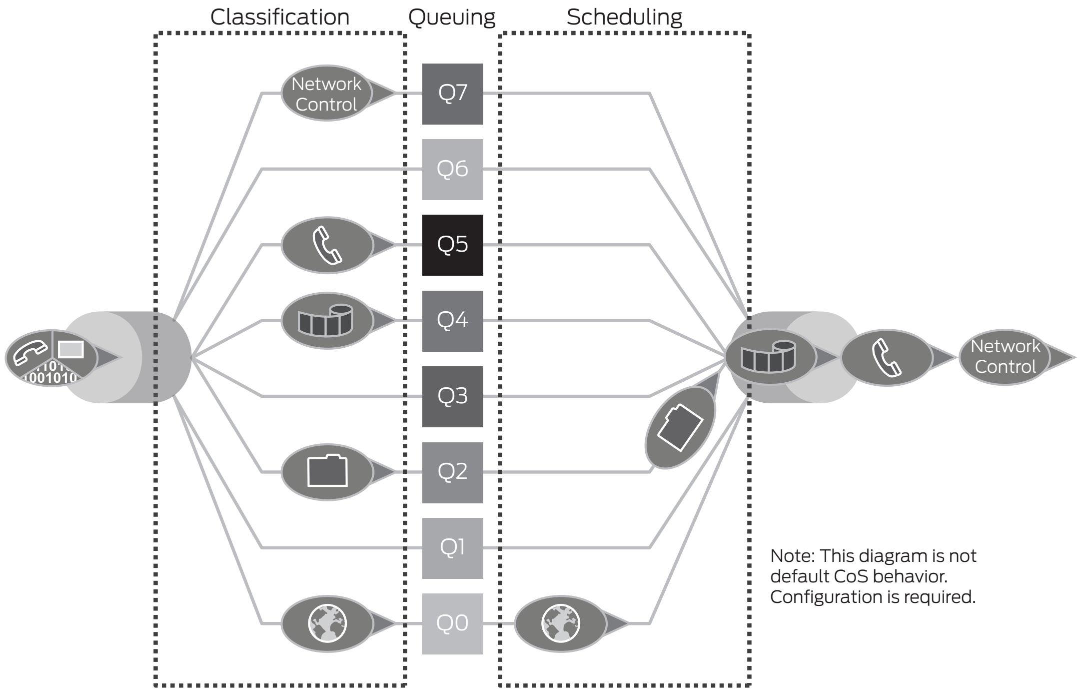
This document describes several deployment options for configuring CoS and QoS for VoIP Traffic when using SRX Series Service Gateways for the Branch. After reading this document, you should be able to configure an SRX Series with ICS device to provide QoS for locally connected IP phones using either a port-based or DiffServ code point (DSCP)-based configuration. You should also be able to configure traffic prioritization, DSCP Marking and queuing for self-generated VoIP traffic to an external SIP trunk or peer call server. QoS over MPLS, IPsec VPN, and Frame Relay are outside the scope of this document.

Defining your Forwarding Classes

Before you can classify VoIP traffic, you must determine which forwarding classes you are using for VoIP.   Junos OS has four default classes, but four additional “custom” classes can be configured—giving you eight total classes.

In these examples two classes are used, VoIP-5 for through-traffic (traffic for phones or other SIP endpoints) and VoIP-6 for self-traffic (traffic to or from the SRX Series itself—for trunks, FXS stations, or announcements). While you can use a single forwarding class for both types of traffic, using two allows you to give higher priority to outbound traffic to SIP trunks. Also, you can have separate QoS counters and statistics. VoIP-6 is given priority over VoIP-5, and network-control is the only class with higher priority then VoIP-6. This leaves six remaining classes for other traffic.

Figure 1: CoS model for classification, queuing, and scheduling



Forwarding class is assigned with packet loss priority (PLP) and DSCPs, which are used for queuing and BA in the core router. For the following UC applications, we recommend the following classifiers and PLP—also known as drop precedence (DP). PLP sets the packet drop precedence value (low or high) to help prevent queue congestion. Packets with a low PLP have higher buffer thresholds than packets with a high PLP. By default, the high threshold is 100 percent of the buffer. Table 1 lists the recommendations for using DiffServ and PLP for voice, video, and other traffic.

Table 1: DiffServ Table

APPLICATION	DIFFSERV	PLP	RECOMMENDED CODE POINT
Network control	CS6	Low	110000
Voice	EF	Low	101110
Video	CS4	Low	100000, 100010
	AF41	High	100100, 100110
	AF42		
	AF43		
Business application	AF21	Low	010010, 010100
	AF22	High	010110
	AF23		
Best effort	Remaining Code Points	Low	010001, etc.

1. Create a forwarding class called Voice-5 using queue #5 and Voice-6 using queue #6.

```
edit class-of-service forwarding-classes
```

```
set queue 5 Voice-5
set queue 6 Voice-6
```

2. Create a scheduler profile **VoiceSched** to map schedulers to a forwarding class.

```
top
```

```
edit class-of-service schedulers

set network-control-scheduler buffer-size percent 10
set network-control-scheduler priority strict-high

set voice6-scheduler priority high
set voice6-scheduler buffer-size percent 10

set voice5-scheduler priority medium-high
set voice5-scheduler buffer-size percent 10

set best-effort-scheduler priority low
set best-effort-scheduler transmit-rate remainder
set best-effort-scheduler buffer-size remainder
```

Note: If using a T1 or serial interface, you need to increase the size of the buffer by using the following command.

This increases the buffer to two seconds, allowing for more packets in the queue for mPIM slot 1.

```
top
```

```
set chassis fpc 1 pic 0 q-pic-large-buffer large-scale
```

3. Create your scheduler maps (a group of forwarding classes), which map your voice schedulers to Junos OS forwarding-class.

```
top
edit class-of-service

edit scheduler-maps VoiceSched

set forwarding-class network-control scheduler network-control-scheduler
set forwarding-class Voice-6 scheduler voice6-scheduler
set forwarding-class Voice-5 scheduler voice5-scheduler
set forwarding-class best-effort scheduler best-effort-scheduler
```

4. Apply your scheduler maps to the interface(s). In this case, fe-0/0/6 and fe-0/0/7 are used. This allows outbound scheduling of voice traffic over the WAN to SIP trunks or peer call servers using Voice-6, and all other intra-enterprise voice endpoints using Voice-5.

```
set interface fe-0/0/6 scheduler-map VoiceSched
set interface fe-0/0/7 scheduler-map VoiceSched
```

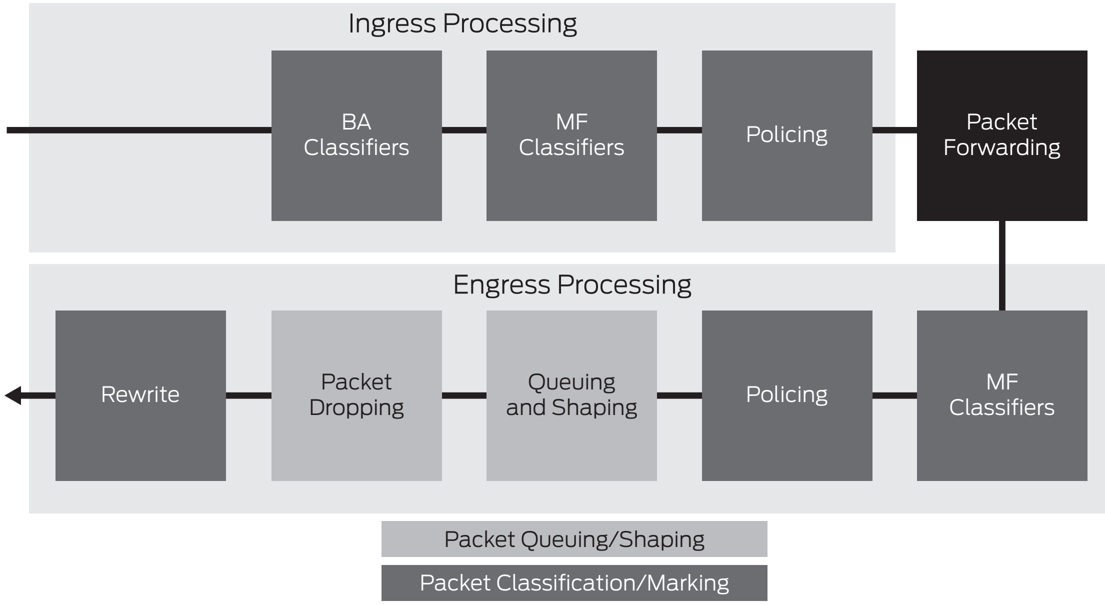


Figure 2: In Junos OS, VoIP and other traffic are classified at the ingress interface, by BA or MF classifiers, Traffic Queuing, Shaping and DSCP rewriting is performed on the egress interface

Port-Based QoS for IP Phones

The following example shows how to classify traffic on a per-port basis. If your IP phones do not support DSCP, or you want certain ports to be configured as "voice only" switch ports on the SRX Series, this places all traffic on these ports in the Voice-5 class—regardless of IP, protocol, or DSCP information received in the packet. This is known as static port-based QoS and is the most simple to configure. The drawback and security risk is that anyone physically connected to that port, even if an IP phone isn't being used, can abuse the forwarding class.

1. Apply the forwarding class Voice-5 to the physical interfaces. In this case, the four PoE ports on a Juniper Networks SRX210 Services Gateway are used.

```
set class-of-service interfaces ge-0/0/0 unit 0 forwarding-class Voice-5
set class-of-service interfaces ge-0/0/1 unit 0 forwarding-class Voice-5
set class-of-service interfaces fe-0/0/2 unit 0 forwarding-class Voice-5
set class-of-service interfaces fe-0/0/3 unit 0 forwarding-class Voice-5
```

2. You can also apply the forwarding class Voice-5 to a VLAN interface to classify all traffic on that VLAN.

```
set class-of-service interfaces vlan unit 101 forwarding-class Voice-5
```

3. Once an IP phone has been connected to these ports, place a call and use the following show commands to validate that traffic is being assigned to the correct forwarding class. This command also shows packets queued, transmitted, and dropped.

```
show interface queues ge-0/0/0 forwarding-class Voice-5
Queue: 5, Forwarding classes: Voice-5
  Queued:
    Packets              :                1603                49 pps
    Bytes                :              343042             85384 bps
  Transmitted:
    Packets              :                1603                49 pps
    Bytes                :              343042             85384 bps
    Tail-dropped packets :                   0                 0 pps
    RED-dropped packets  :                   0                 0 pps
```

Filter-Based Classification

In addition to port-based classification, (firewall) filter-based classification allows you to classify packets arriving at an interface based on Layer 3 or 4 header information—such as source IP, port number, and packet size. The following filter terms are used and applied to the ingress interface to classify traffic as either SIP or RTP. You can optionally put source or destination IP addresses in the filters to be more specific to your application.

1. Create a filter term for SIP by using destination-port 5060.

```
set firewall filter VoIP-Thru term 1 from protocol udp port 5060
set firewall filter VoIP-Thru term 1 then log count Voice5-SIP
set firewall filter VoIP-Thru term 1 then forwarding-class Voice5 accept
```

2. Create a filter term for RTP by using a packet size of 200. This is the exact packet size of RTP messages when you use the G711 CODEC.

```
set firewall filter VoIP-Thru term 2 from protocol udp packet-length 200
set firewall filter VoIP-Thru term 2 then log count Voice5-RTP
set firewall filter VoIP-Thru term 2 then forwarding-class Voice5 accept
```

3. Create a filter to match all other traffic with counting and enabled.

```
set firewall filter VoIP-Thru term 3 from address 0.0.0.0/0
set firewall filter VoIP-Thru term 4 then log count Any-Rule accept
```

4. Apply the filters as input filters to the ingress interfaces, connected to IP phones. In this case, VLAN101 is used along with fe-0/0/6 as examples.

```
set interface vlan.101 family inet filter input VoIP-Thru
set interface fe-0/0/6 family inet filter input VoIP-Thru
```

5. Test the filters by clearing them and sending SIP and RTP traffic through the SRX Series. Use the following show commands to see the firewall term counters incrementing.

```
clear firewall all           # First clear the counters


root> show firewall

Filter: VoIP5
Counters:
Name                                Bytes              Packets
Any-Rule                             120839                150
Voice5-RTP                           111400                557

root> show firewall counter Voice5-RTP filter VoIP5

Filter: VoIP5
Counters:
Name                                  Bytes              Packets
Voice5-RTP                           130800                    654
```

DSCP-Based QoS for IP Phones (Thru-Traffic)

By having branch IP phones (and softphones) mark SIP and RTP traffic using a DiffServ bit, the SRX Series can prioritize and queue the VoIP traffic as it passes through the gateway. This technique can also be used for SIP trunks and peer call servers, provided the SRX Series receives DSCP market packets from these hosts.

The following assumptions are made about the QoS configuration:

- RTP/SIP traffic is classified, if this traffic has a DiffServ code point of 46 (`binary 101110`)
- Traffic classified as **Voice-5** is queued into forwarding-class 5, the third highest on the router.
- Classification is applied and occurs at the ingress interface,
- Queuing is performed only on the WAN egress interface (fe-0/0/7).
- The DiffServ marking is preserved as the packet enters the WAN, MPLS, or IPsec tunnel.

Configuration

1. Create a new classifier profile. Import the default classifier to avoid defining all DSCP values, and only reclassify the DSCP value of the VoIP traffic that needs to be prioritized. The IP telephone is configured to use the DSCP decimal value of 46 (**`101110`** in binary) for signaling and media packets.

Note: This DSCP value for Avaya is configured in the ip-network-region form of Communication Manager.

```
top
edit class-of-service classifiers
set dscp 46 import default
set dscp 46 forwarding-class Voice-5 loss-priority medium-low code-points 101110

top
edit class-of-service interfaces
set ge-0/0/0 unit 0 classifiers dscp 46
set ge-0/0/1 unit 0 classifiers dscp 46
set fe-0/0/2 unit 0 classifiers dscp 46
set fe-0/0/3 unit 0 classifiers dscp 46
```

2. Create a rewrite rule for DSCP 46 by using forwarding class Voice-5.

```
top
edit class-of-service rewrite-rules
set dscp 46-VoIP forwarding-class Voice-5 loss-priority medium-high code-point
101110
```

3. Apply your rewrite rule to all the VoIP egress interfaces, including Untrust and locally connected IP phones.

```
top
edit class-of-service interfaces
set ge-0/0/0 unit 0 rewrite-rules dscp 46-VoIP
set ge-0/0/1 unit 0 rewrite-rules dscp 46-VoIP
set fe-0/0/2 unit 0 rewrite-rules dscp 46-VoIP
set fe-0/0/3 unit 0 rewrite-rules dscp 46-VoIP
```

4. Apply the scheduler profile VoiceSched and classifiers for DSCP 46 to the access (IP phones) and uplink (Untrust) ports.

```
edit class-of-service interfaces
set ge-0/0/0 scheduler-map VoiceSched
set ge-0/0/0 unit 0 classifiers dscp 46
set ge-0/0/1 scheduler-map VoiceSched
set ge-0/0/1 unit 0 classifiers dscp 46
set fe-0/0/2 scheduler-map VoiceSched
set fe-0/0/2 unit 0 classifiers dscp 46
set fe-0/0/3 scheduler-map VoiceSched
set fe-0/0/3 unit 0 classifiers dscp 46
```

5. Test the configuration by first setting up your IP phone to use DSCP code point of 46 for all SIP and RTP traffic. You might want to validate this using a sniffer to examine the IP headers being transmitted. Once confirmed, you can display statistics on the Junos OS device by using the following commands.

```
root> show interfaces queue fe-0/0/6 forwarding-class Voice-5
Physical interface: fe-0/0/6, Enabled, Physical link is Up
  Interface index: 137, SNMP ifIndex: 120
Forwarding classes: 8 supported, 5 in use
Egress queues: 8 supported, 5 in use
Queue: 5, Forwarding classes: Voice5
  Queued:
    Packets              :                 1603                    49 pps
    Bytes                :               343042                 85384 bps
  Transmitted:
    Packets              :                 1603                    49 pps
    Bytes                :               343042                 85384 bps
    Tail-dropped packets :                    0                     0 pps
    RED-dropped packets  :                    0                     0 pps
     Low                 :                    0                     0 pps
     Medium-low          :                    0                     0 pps
     Medium-high         :                    0                     0 pps
     High                :                    0                     0 pps
    RED-dropped bytes    :                    0                     0 bps
     Low                 :                    0                     0 bps
     Medium-low          :                    0                     0 bps
     Medium-high         :                    0                     0 bps
     High                :                    0                     0 bps
```

Implementing QoS on SRX Series Self-Traffic

In addition to VoIP traffic passing through the SRX Series, it is also possible to configure QoS on VoIP traffic being initiated from the SRX Series, or terminating to the SRX Series. The following section covers this configuration—which enables queuing and prioritization of RTP packets leaving the WAN Interface—fe-0/0/6 in this example.

To prioritize traffic flowing to/from the DSP, the following Junos OS firewall term is created and applied to the mpu (DSP) interface of the SRX Series.

1. Create a filter term for SIP by using destination-port 5060.

```
set firewall filter VoIP-Self term 1 from protocol udp port 5060
set firewall filter VoIP-Self term 1 then log count Voice6-SIP
set firewall filter VoIP-Self term 1 then forwarding-class Voice5 accept
```

2. Create a filter term for RTP by using a packet size of 200. This is the exact packet size of RTP messages when you use the G711 CODEC.

```
set firewall filter VoIP-Self term 2 from protocol udp packet-length 200
set firewall filter VoIP-Self u term 2 then log count Voice6-RTP
set firewall filter VoIP-Self term 2 then forwarding-class Voice6 accept
```

3. Create a filter to match all other traffic with counting and enabled.

```
set firewall filter VoIP-Self term 3 from address 0.0.0.0/0
set firewall filter VoIP-Self term 4 then log count Any-Rule accept
```

4. Apply the filters as input filters to the MPU interface as RTP packets from the DSP on SRX Series comes from the MPU interface. This will classify the traffic as it comes into the SRX Series (from the DSP).

```
set interface mpu-0/0/9 family inet filter input VoIP-Self
```

5. Apply the filters as output filters to the WAB interface, fe-0/0/7 in this example. This is so that SIP and RTP packets leaving the box get classified appropriately before leaving the WAN Interface.

```
set interface fe-0/0/7 family inet filter output VoIP-Self
```

6. Be sure you have the scheduler maps applied to any LAN interfaces that IP phones are connected.

```
edit class-of-service interfaces
set ge-0/0/0 scheduler-map VoiceSched
set ge-0/0/1 scheduler-map VoiceSched
set ge-0/0/2 scheduler-map VoiceSched
```

7. Test the filters by clearing them and sending SIP and RTP traffic through the SRX Series. Use the following show commands to see the firewall term counters incrementing.

```
clear firewall all           # First clear the counters


root> show firewall counter Voice6-RTP

Filter: VoIP6
Counters:
Name                                  Bytes             Packets
Voice6-RTP                           130800                 654
```

Implementing DSCP Marking on SRX Series Self-VoIP Traffic

In addition to scheduling, DSCP marking may be desired on self-generated SIP or RTP traffic from the SRX Series. Junos OS 10.1 introduces new media-policy commands under services | convergence-services which allow you to mark outbound RTP traffic based on the address of the SIP Peer, which may be either an IP Address or FQDN. The following examples outline this configuration.

1. Create a media policy to match traffic for your local subnet with an action to mark the traffic with DSCP 46.

```
edit services convergence-services
set media-policy 1 term 1 from peer-address ip-address 10.0.0.0/8
set media-policy 1 term 1 then dscp 46
```

2. Create a media policy to match RTP traffic to/from a specific SIP trunk with an action to mark the traffic with DSCP 46.

```
set media-policy 1 term 2 from peer-address fqdn sipgate.com
set media-policy 1 term 2 then dscp 46
commit
```

After committing this configuration, RTP traffic destined to these endpoints will be marked with DSCP value of 46. If desired, separate DSCP values can be used for each term rule, for example RTP traffic destined for sipgate.com could be marked with DSCP 42, while local-subnets (10.0.0.0) could be marked with 46.

3. To mark outbound SIP traffic with a DSCP bit, the following rewrite-rules are created and applied to the fe-0/0/7 interface. This will mark all traffic in forwarding-class VoiP5 with DSCP value 46 (101110).

```
edit class-of-service
set rewrite-rules dscp sip-dscp forwarding-class VoIP5 loss-priority low code-
point 101110
set interfaces fe-0/0/7 unit 0 rewrite-rules sip-dscp
```

Implementing Call Admission Control on SIP Trunks

In addition to Classification, Prioritization and Marking of SIP and RTP traffic, the SRX Series with ICS is also capable of performing call admission control on any SIP trunk configured on the box. Junos OS 10.1 supports static CAC and allows you to simply define the max number of concurrent calls that will be allowed on a specific SIP trunk. The following examples outline the configuration necessary to setup a SIP trunk and limit the max calls to 5.

1. Create a SIP trunk, in this example a SIP trunk service provider is used. If you already have a SIP trunk defined you can skip ahead to step 2.

```
edit services convergence-services
set trunk sipgate.com trunk-type sip peer-proxy-server address fqdn sipgate.com
edit trunk sipgate.com trunk-type sip peer-proxy-server
set auth-id MySipUsername
set auth-password MySipPassword
set codec G711-MU
set dtmf-method rfc-2833
```

2. Edit the SIP trunk and set the max-concurrent-calls to 5.

```
edit trunk sipgate.com trunk-type sip peer-proxy-server
set max-concurrent-calls 5
```

3. Create a trunk group that includes both SIP trunk and PSTN, with SIP trunk as priority #1 and FXO as priority #2. Note that the third concurrent call will be routed over the FXO trunk because max-concurrent-calls are limited to 2 on the SIP trunk.

```
Set trunk-group Sip+PSTN trunk sipgate.com trunk fxo1 trunk fxo2
commit
```

After committing this configuration, the SRX Series will allow a maximum of 5 concurrent calls to be routed through the SIP trunk, the 6th call may routed to other trunks included in the trunk group (FXO trunks) or simply denied "All circuit are busy now." For more information on setting up the dial-plan to use the SIP trunk, please see the *SRX Series with ICS Golden Configurations* document.

Summary

SRX Series for the branch provide all the features required to securely connect modern remote and branch offices in a one-box solution. Junos OS offers users unparalleled flexibility designed to meet the most demanding network requirements. After reading this document, you can configure an SRX Series for the branch device to securely pass traffic, support both analog and SIP phones, and provide basic calling features. With a little practice, you can create advanced configurations required for more complex deployments.

About Juniper Networks

Juniper Networks, Inc. is the leader in high-performance networking. Juniper offers a high-performance network infrastructure that creates a responsive and trusted environment for accelerating the deployment of services and applications over a single network. This fuels high-performance businesses. Additional information can be found at **www.juniper.net**.

Corporate and Sales Headquarters
Juniper Networks, Inc.
1194 North Mathilda Avenue
Sunnyvale, CA 94089 USA
Phone: 888.JUNIPER (888.586.4737)
or 408.745.2000
Fax: 408.745.2100
www.juniper.net

APAC Headquarters
Juniper Networks (Hong Kong)
26/F, Cityplaza One
1111 King's Road
Taikoo Shing, Hong Kong
Phone: 852.2332.3636
Fax: 852.2574.7803

EMEA Headquarters
Juniper Networks Ireland
Airside Business Park
Swords, County Dublin, Ireland
Phone: 35.31.8903.600
EMEA Sales: 00800.4586.4737
Fax: 35.31.8903.601

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3500182-002-EN Nov 2010

Printed on recycled paper