



Avaya Solution & Interoperability Test Lab

Application Notes for the Packeteer PacketShaper with Avaya SIP IP Telephony – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring the Packeteer PacketShaper to guarantee WAN link bandwidth to SIP Voice over IP (VoIP) RTP traffic generated by Avaya SIP Telephones and Softphones, and give high priority to SIP VoIP signaling traffic generated by Avaya SIP Enablement Services (SES) servers, and Avaya SIP Telephones and Softphones. During compliance testing, SIP phone calls traversing the WAN link were successfully established and maintained while non-VoIP traffic such as HTTP and FTP traffic was sharing the WAN link. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the *DeveloperConnection* Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe a compliance-tested configuration comprised of Avaya SIP Enablement Services (SES), Avaya Communication Manager, Avaya SIP Telephones, Avaya SIP Softphones, and the Packeteer PacketShaper. PacketShaper is a traffic management appliance that monitors and controls IP network traffic going over WAN links. Typically placed between a site's WAN router and LAN, PacketShaper identifies and analyzes inbound and outbound WAN traffic up to and including the OSI Application Layer (Layer 7). In addition, PacketShaper manages WAN link utilization and throughput based on the bandwidth reservations and policies applied to the identified traffic classes. PacketShaper can thus ensure that SIP Voice over IP (VoIP) packets receive their guaranteed share of the WAN link capacity and do not encroach upon the guarantees provided to other traffic classes and applications sharing the WAN link.

Figure 1 illustrates a sample configuration consisting of an Avaya S8710 Media Server, an Avaya G650 Media Gateway, an Avaya SIP Enablement Services (SES) Server, Avaya SIP and H.323 IP Telephones, an Avaya SIP Softphone, and a Packeteer PacketShaper 6500 appliance. Avaya Communication Manager runs on the S8710 Media Server, though the solution described herein is also extensible to other Avaya Media Servers and Media Gateways. The Avaya C364T-PWR Converged Stackable Switch and Avaya P333T-PWR Stackable Switch perform Layer 2 switching within the enterprise site VLANs, and the Avaya C364T-PWR also routes packets between the enterprise site VLANs. The Cisco 2900 and 1841 routers route packets between the enterprise site and the simulated WAN.

In **Figure 1**, the SES server is configured as a combined home/edge SIP proxy and registrar server. The Avaya 4620SW SIP Telephone within the enterprise, as well as the 4610SW SIP Telephone and Avaya SIP Softphone in the WAN, register with the SES server. A T1 PPP link connects the enterprise site to a simulated WAN. The WAN SIP phones (4610SW SIP Telephone and Avaya SIP Softphone) exchange SIP signaling messages with the SES server over the T1 link. RTP packets between the WAN SIP phones and the enterprise also traverse the T1 link.

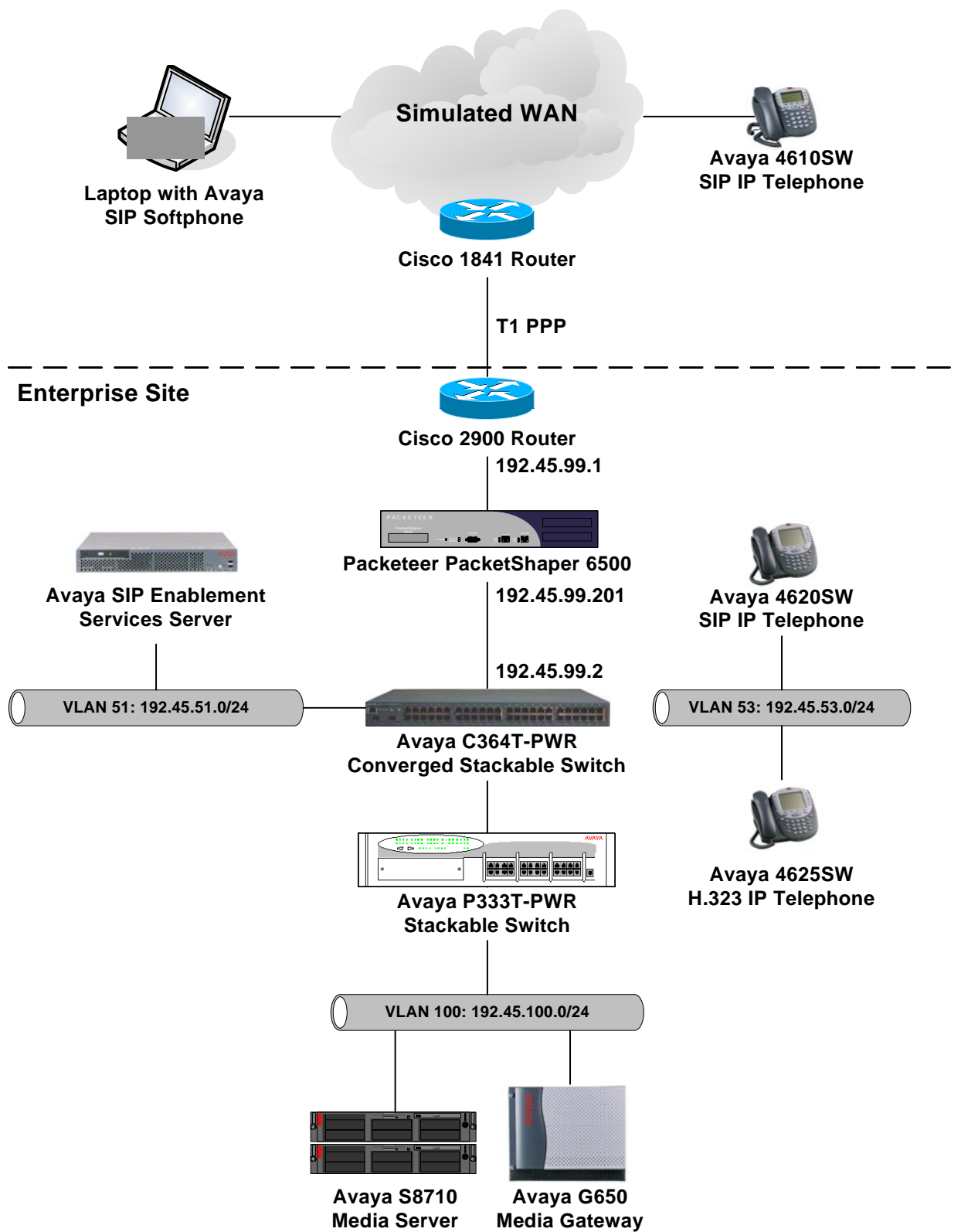


Figure 1: Sample configuration.

2. Equipment and Software Validated

The following equipment and software/firmware were used for the sample configuration provided:

Equipment	Software/Firmware
Avaya S8710 Media Server	Avaya Communication Manager 3.0 (340.3)
Avaya G650 Media Gateway	-
TN2312BP IP Server Interface	21
TN799DP C-LAN Interface	15
TN2302AP IP Media Processor	104
Avaya SIP Enablement Services Server	SIP Enablement Services 3.0.0.0-031.0
Avaya 4600 Series IP Telephones	2.2 (4610SW SIP) 2.2 (4620SW SIP) 2.5 (4625SW H.323)
Avaya SIP Softphone	2.0 Build 43
Avaya C364T-PWR Converged Stackable Switch	4.5.14
Avaya P333T-PWR Stackable Switch	4.0.17
Packeteer PacketShaper 6500	7.2.1g1
Cisco 2900 Router	IOS 12.3
Cisco 1841 Router	IOS 12.3
Laptop	Windows XP Professional SP2

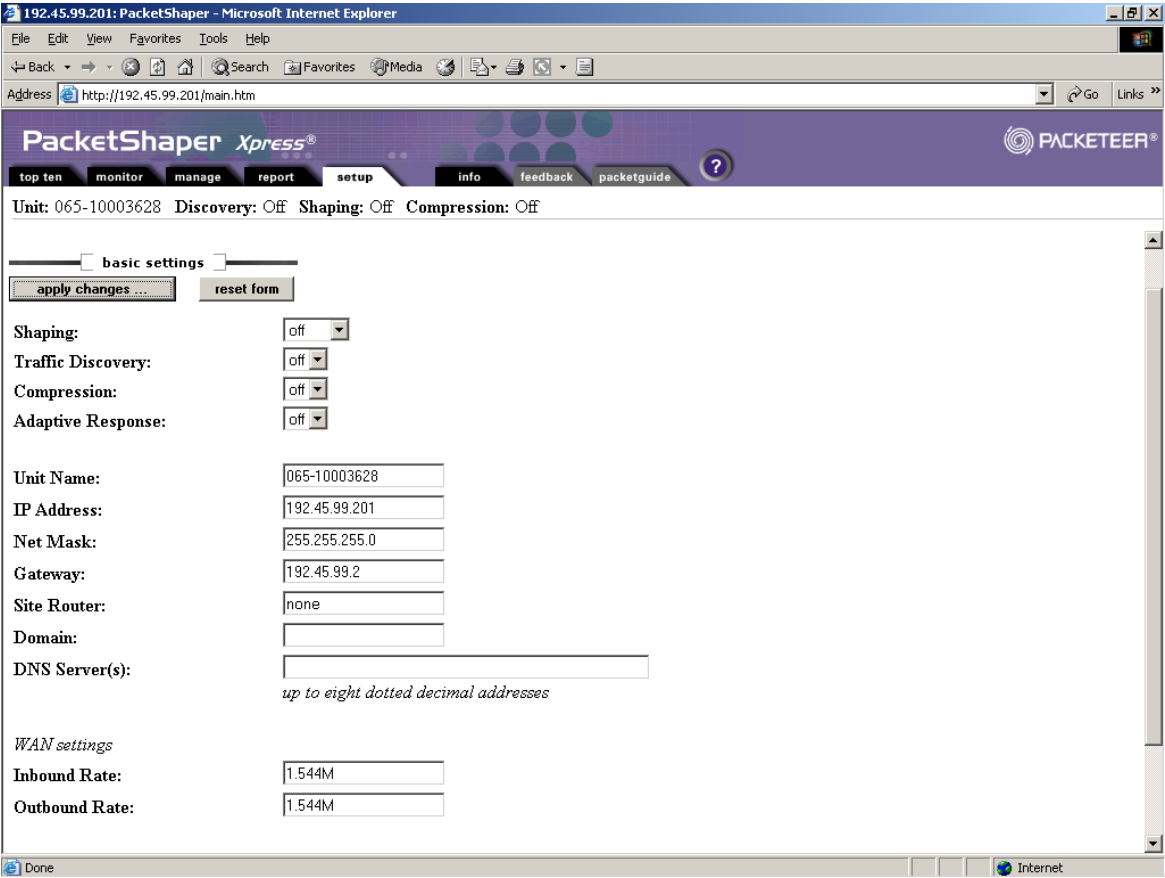
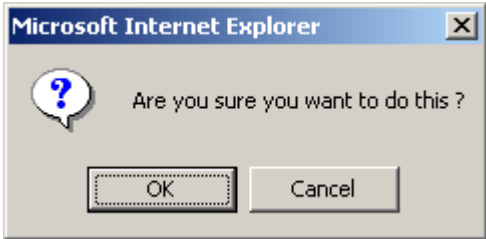
3. Configure Packeteer PacketShaper

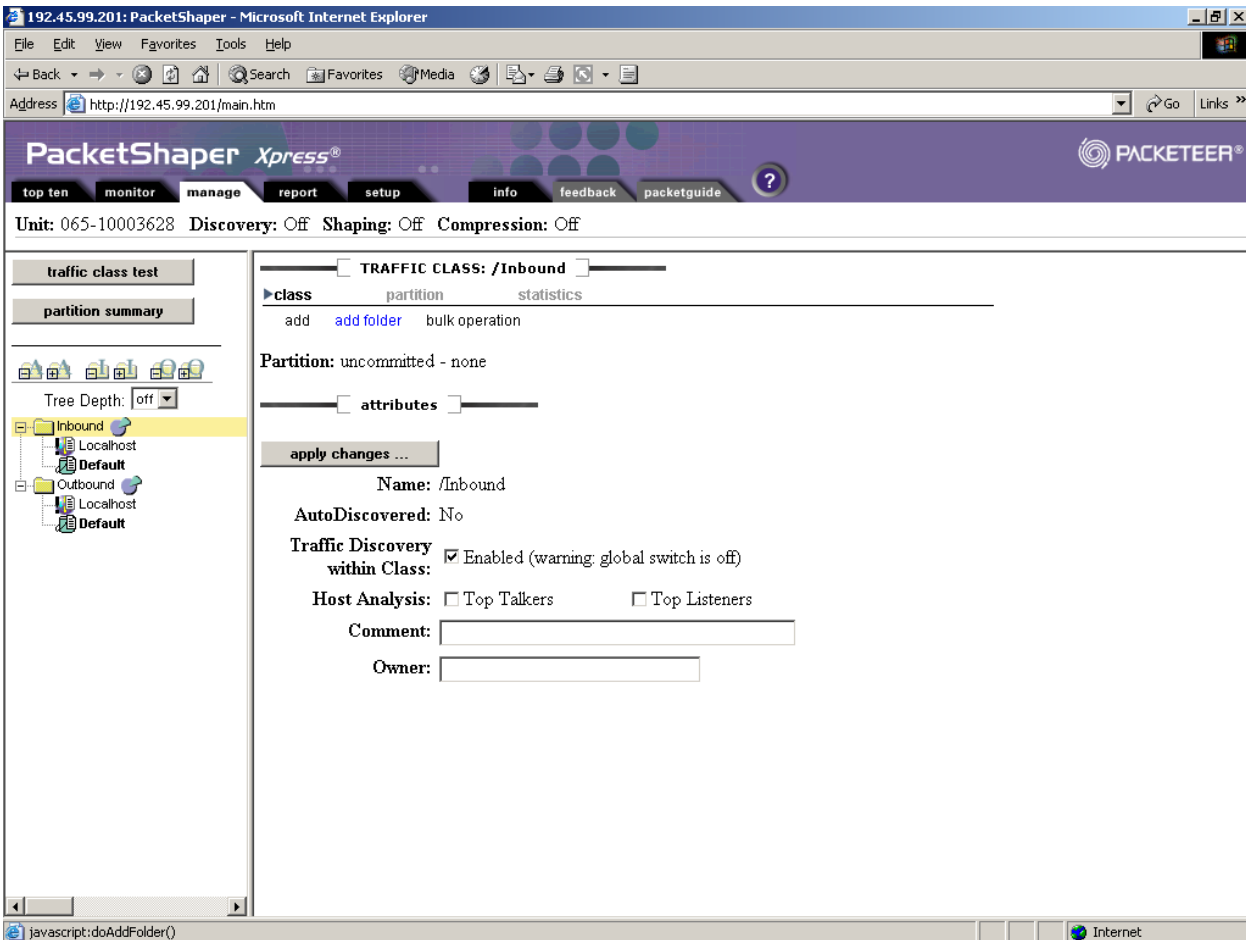
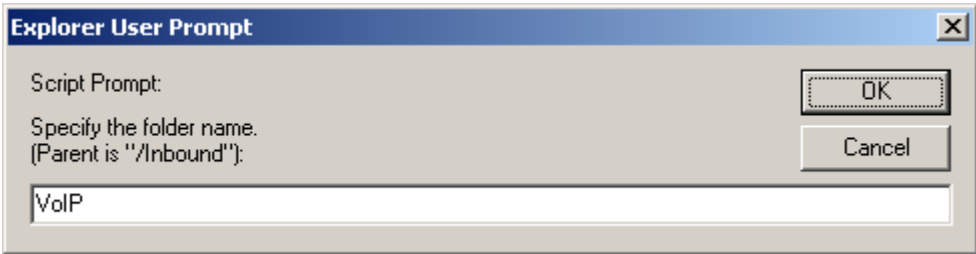
This section describes the steps for creating traffic classes for SIP VoIP protocols such as SIP, RTP, and RTCP, and configuring partitions and policies for those traffic classes on PacketShaper. The configuration is described in terms of inbound traffic, but is equally valid for outbound traffic. In fact, for the configuration of **Figure 1**, where the WAN link capacity (1.544 Mbps) is lower than the LAN capacity (10/100 Mbps), bandwidth management in the outbound direction towards the WAN is more critical than in the inbound direction towards the LAN.

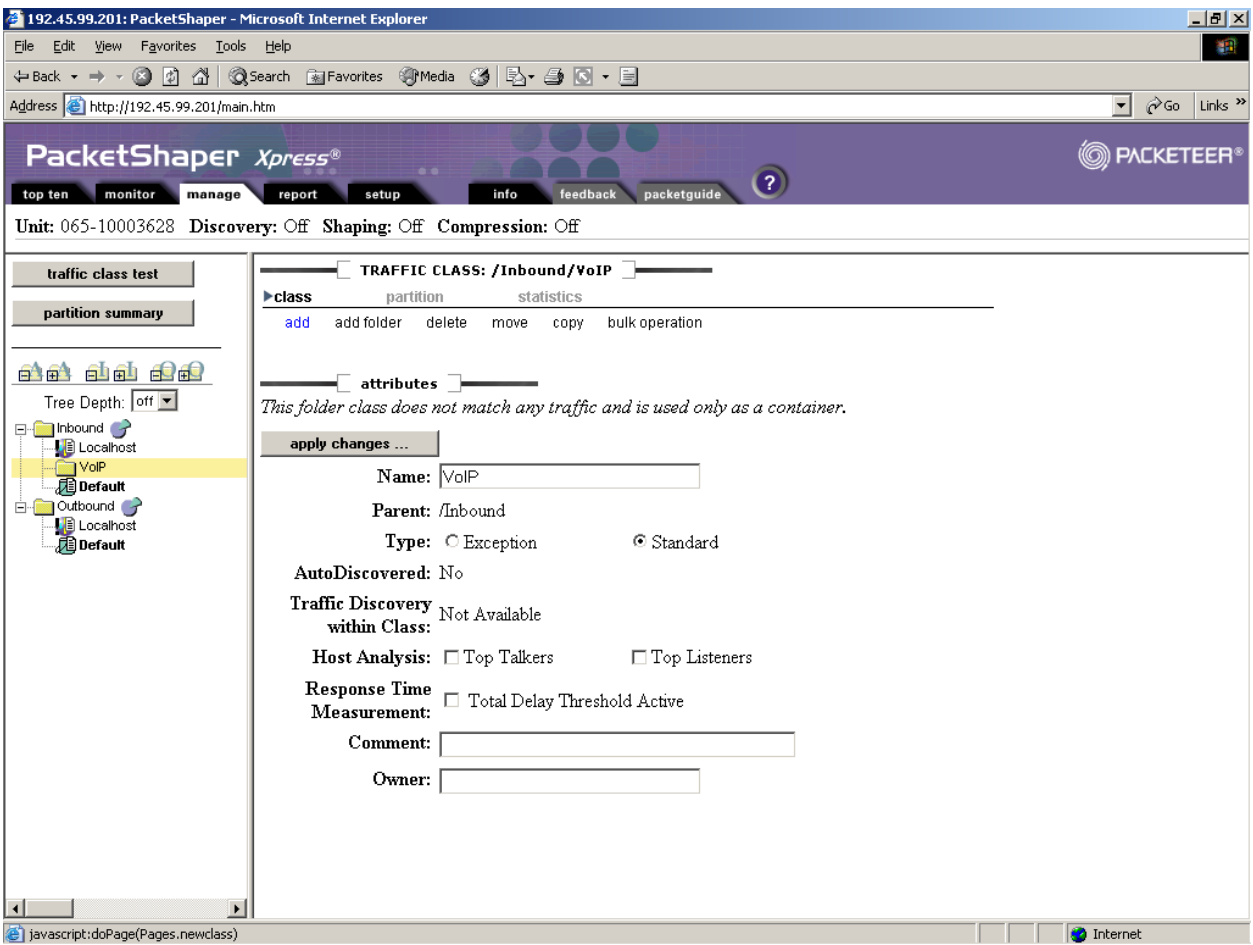
3.1. Create Inbound Traffic Classes

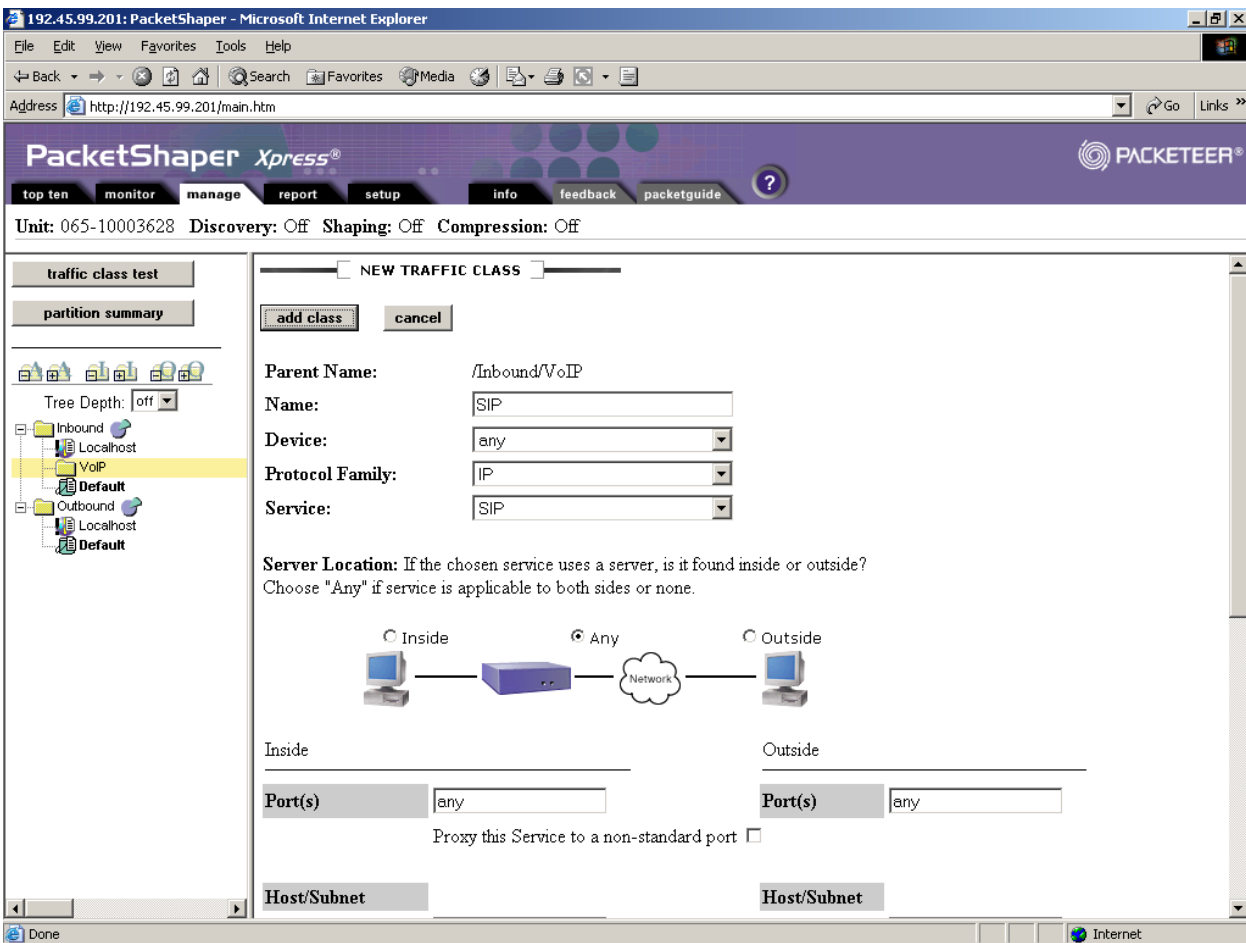
A traffic class identifies the traffic flows of specific IP addresses, protocols, or applications. Traffic classes may be either discovered or created. This section discusses the creation of traffic classes relevant to SIP VoIP.

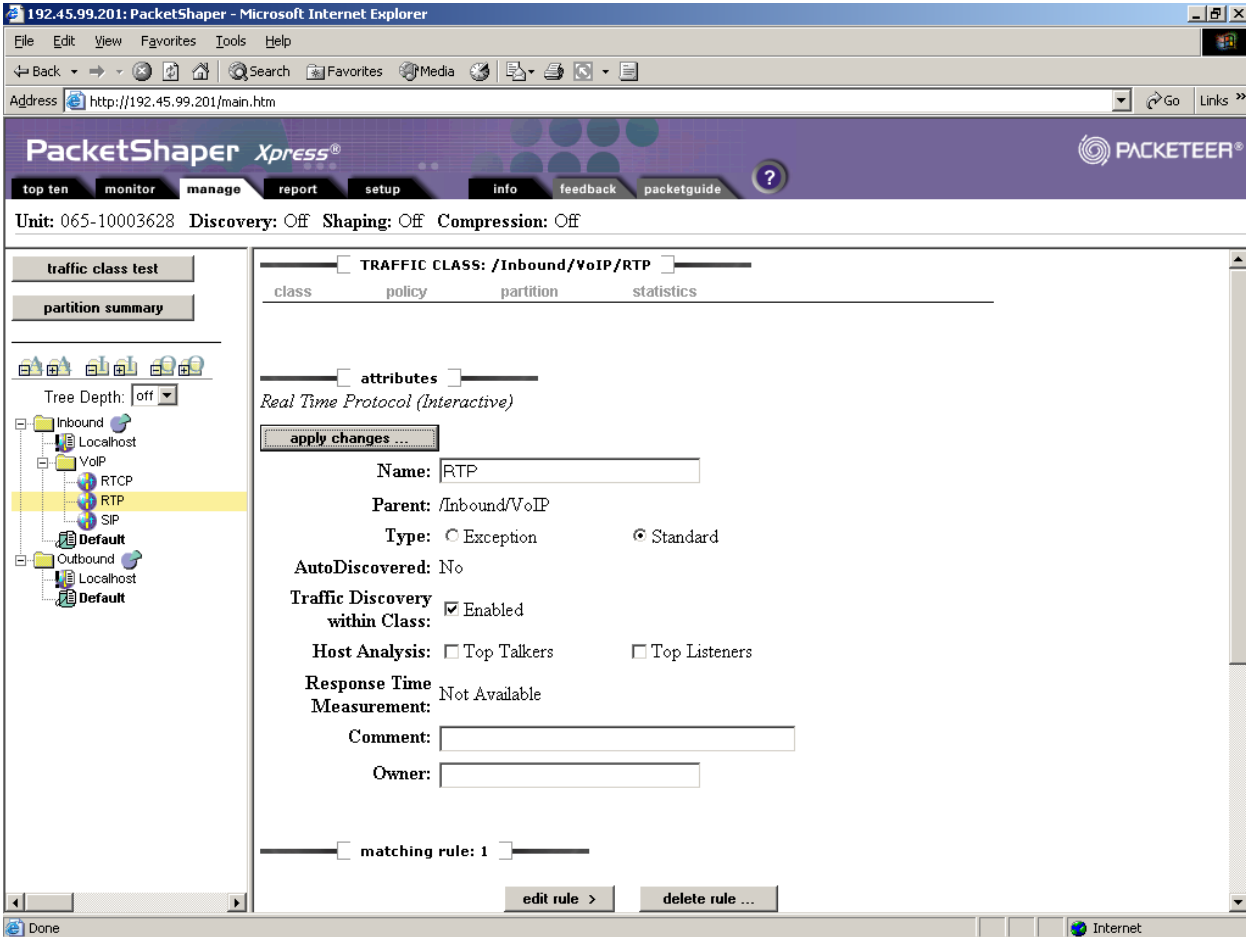
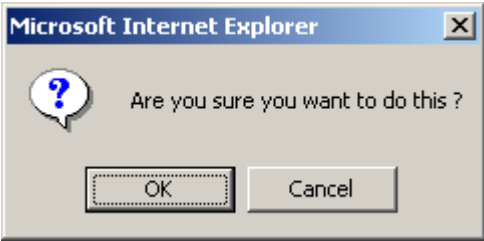
Step	Description
1.	Open a browser and enter <a href="http://<hostname or IP address of PacketShaper>">http://<hostname or IP address of PacketShaper> as the URL. Log in with the appropriate credentials.

Step	Description
2.	<p>Click on the setup tab. Enter the capacity of the WAN link in the Inbound Rate and Outbound Rate textboxes and click on “apply changes ...”. In the example below, a rate of 1.544M is used to approximate the T1 link rate.</p> 
3.	<p>Click on “OK” to confirm the changes.</p> 
4.	<p>Click on the manage tab and then the Inbound folder in the left panel.</p>

Step	Description
5.	<p>Select class->add folder to create a folder for inbound VoIP traffic classes.</p> 
6.	<p>Enter a descriptive name for the folder and click on “OK”.</p> 

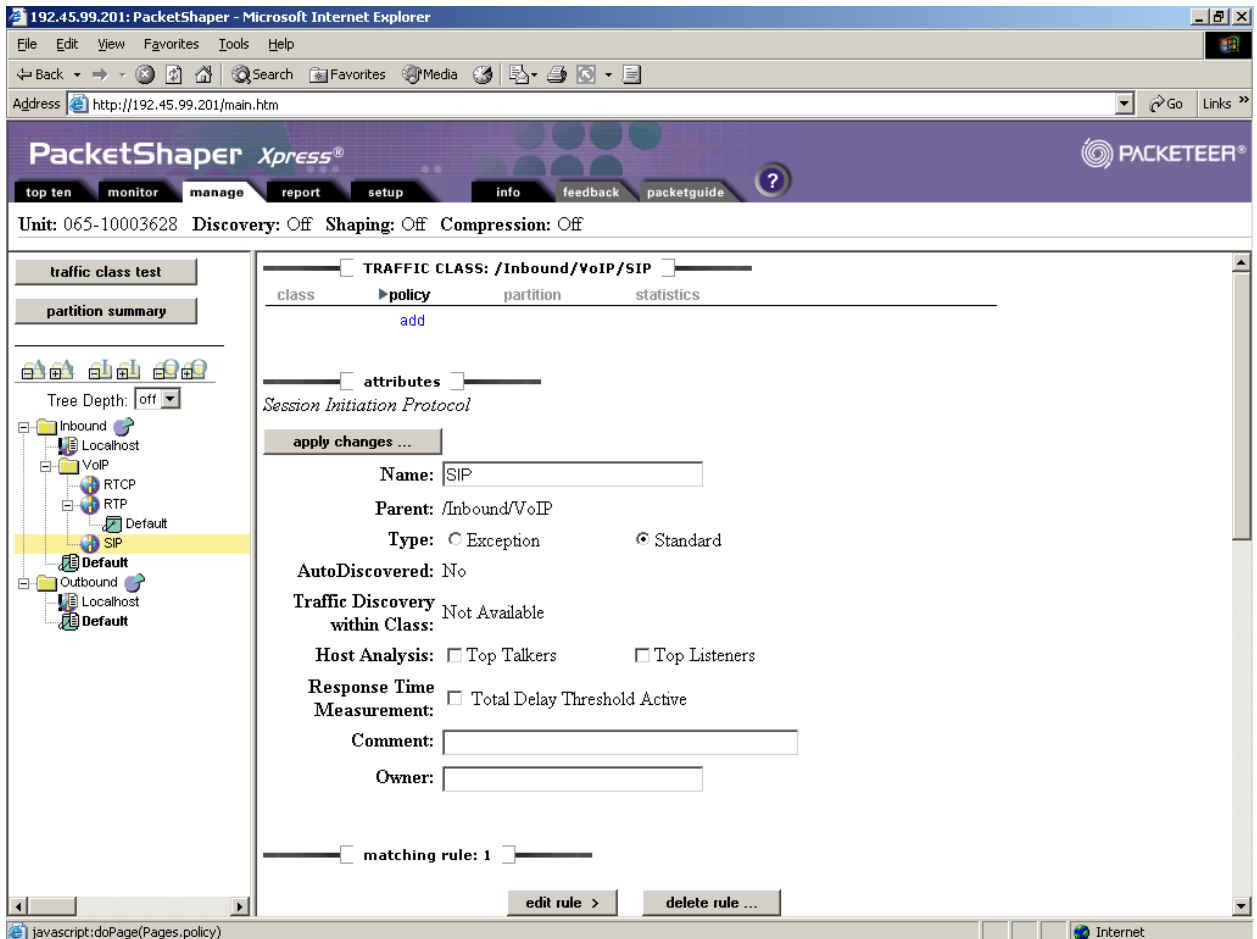
Step	Description
7.	<p>Click on the newly created folder in the left panel. Select class->add to create an inbound traffic class.</p> 

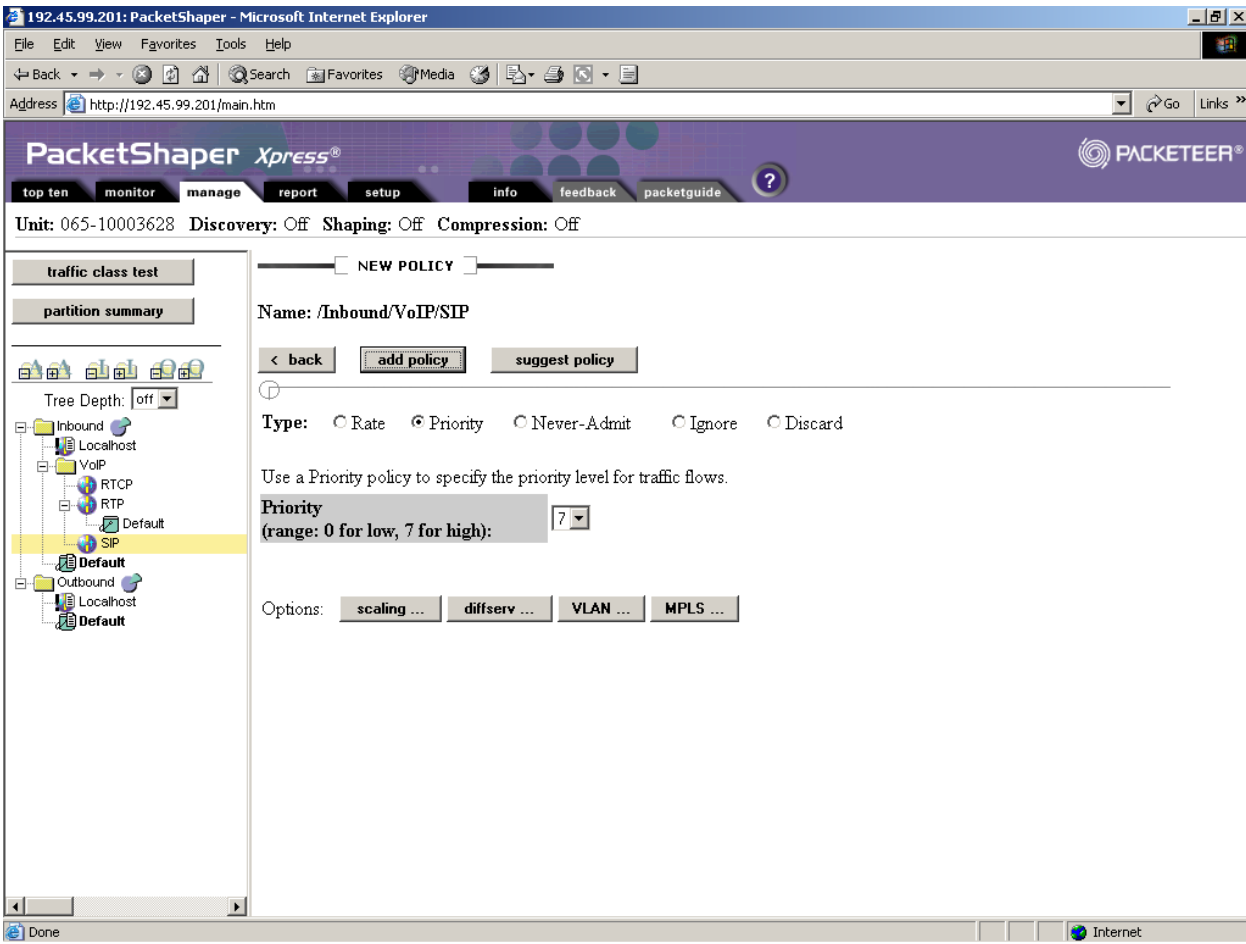
Step	Description
8.	<p>Enter a descriptive Name for the traffic class and select the Service corresponding to the traffic class. The example below shows the creation of a SIP traffic class. Click on “add class”.</p> 
9.	<p>Repeat Steps 7-8 for the following traffic classes:</p> <ul style="list-style-type: none"> • RTP: Select RTP-I from the Service drop-down menu box. • RTCP: Select RTCP-I from the Service drop-down menu box.

Step	Description
10.	<p>Click on the RTP traffic class in the left panel. Enable Traffic Discovery within Class to allow PacketShaper to automatically create sub-classes for discovered codecs. Click on “apply changes ...”.</p> 
11.	<p>Click on “OK” to confirm the change.</p> 

3.2. Assign Priority Policies to Inbound Call Control Traffic Classes

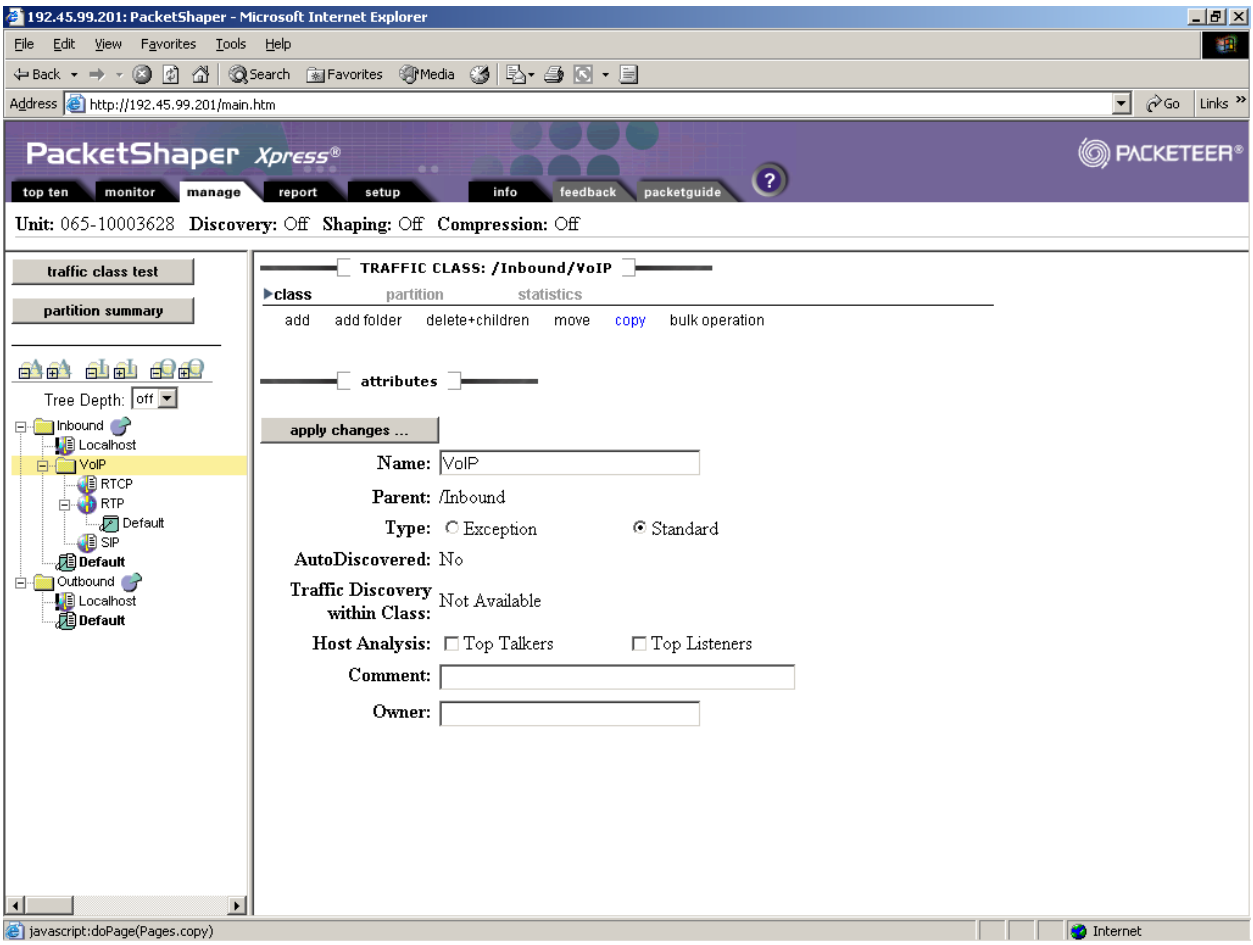
A priority policy applied to a traffic class specifies the precedence that packets within the traffic class should have relative to packets within other traffic classes. For example, a priority policy may be applied to each SIP call control traffic class so that SIP call control traffic takes precedence over less real-time sensitive data traffic.

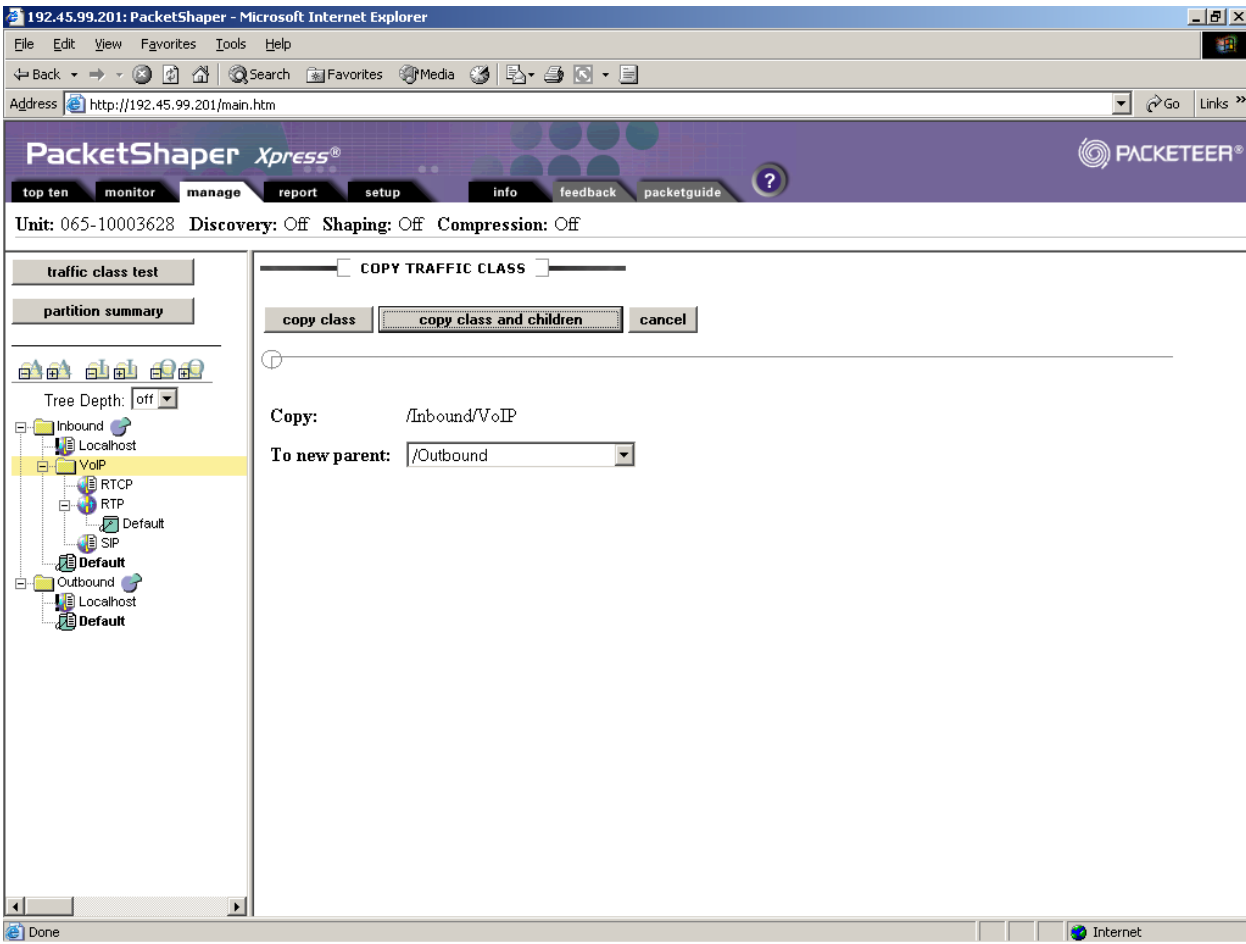
Step	Description
1.	<p>In the manage tab, click on the inbound SIP traffic class in the left panel and select policy->add.</p>  <p>The screenshot shows the PacketShaper Xpress web interface in a Microsoft Internet Explorer browser window. The address bar shows 'http://192.45.99.201/main.htm'. The interface has a navigation bar with tabs: top ten, monitor, manage, report, setup, info, feedback, and packetguide. The 'manage' tab is active. Below the navigation bar, there's a status line: 'Unit: 065-10003628 Discovery: Off Shaping: Off Compression: Off'. The main area is divided into two panels. The left panel shows a tree view of traffic classes: Inbound, Localhost, VoIP, RTP, Default, SIP, Outbound, and another Localhost/Default. The 'SIP' class under 'Inbound' is selected. The right panel shows the configuration for the selected traffic class, 'TRAFFIC CLASS: /Inbound/VoIP/SIP'. It has tabs for class, policy, partition, and statistics. The 'policy' tab is active, showing an 'add' button. Below that, there's an 'attributes' section with a 'Session Initiation Protocol' label and an 'apply changes ...' button. The configuration fields include: Name: SIP, Parent: /Inbound/VoIP, Type: Standard (selected), AutoDiscovered: No, Traffic Discovery within Class: Not Available, Host Analysis: Top Talkers and Top Listeners (both unchecked), Response Time Measurement: Total Delay Threshold Active (unchecked), Comment: (empty field), and Owner: (empty field). At the bottom, there's a 'matching rule: 1' section with 'edit rule >' and 'delete rule ...' buttons.</p>

Step	Description
2.	<p>Set Type to “Priority” and specify “7” for the Priority value. Click on “add policy”.</p> 
3.	Repeat Steps 1-2 for the inbound RTCP traffic class.

3.3. Configure Outbound Traffic Classes

Traffic classes may be configured for outbound traffic in the same manner as described in Sections 3.1 – 3.2 for inbound traffic. If the outbound configuration is to be the same as the inbound configuration, then perform the steps below.

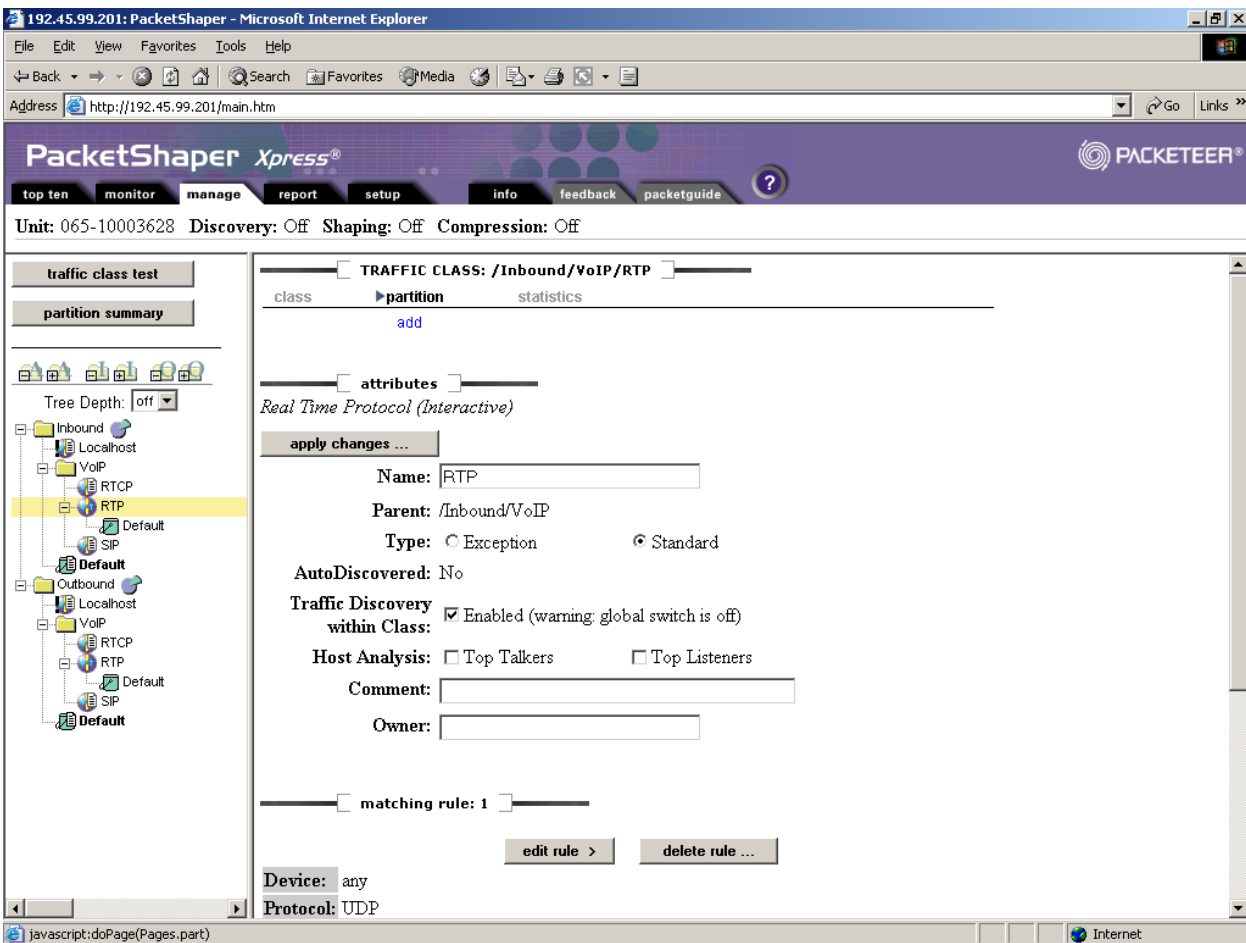
Step	Description
1.	<p>In the manage tab, click on the inbound VoIP folder in the left panel and select class->copy.</p>  <p>The screenshot shows the PacketShaper Xpress web interface in a Microsoft Internet Explorer browser window. The address bar shows 'http://192.45.99.201/main.htm'. The interface has a top navigation bar with tabs: top ten, monitor, manage, report, setup, info, feedback, and packetguide. The 'manage' tab is active. Below the navigation bar, there's a status bar showing 'Unit: 065-10003628', 'Discovery: Off', 'Shaping: Off', and 'Compression: Off'. The main content area is divided into two panels. The left panel is a tree view showing the hierarchy of traffic classes. The 'Inbound' folder is expanded, and the 'VoIP' folder is selected. The right panel shows the configuration for the selected 'VoIP' class. It includes fields for Name (VoIP), Parent (/Inbound), Type (Standard), AutoDiscovered (No), Traffic Discovery (Not Available), Host Analysis (Top Talkers, Top Listeners), Comment, and Owner. The 'copy' button is highlighted in the top navigation bar.</p>

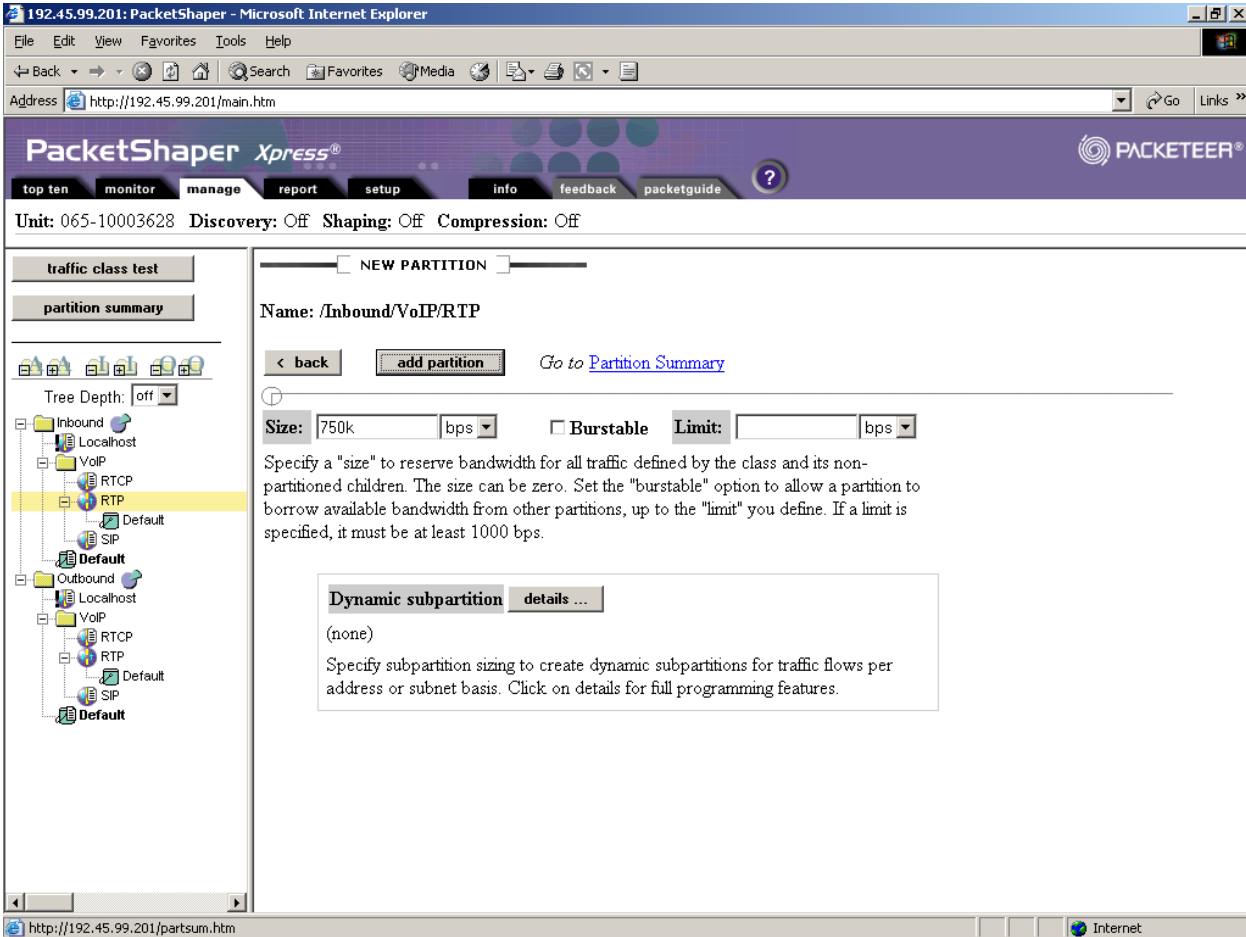
Step	Description
2.	<p data-bbox="277 233 1268 268">Set To new parent to “/Outbound” and click on “copy class and children”.</p> 

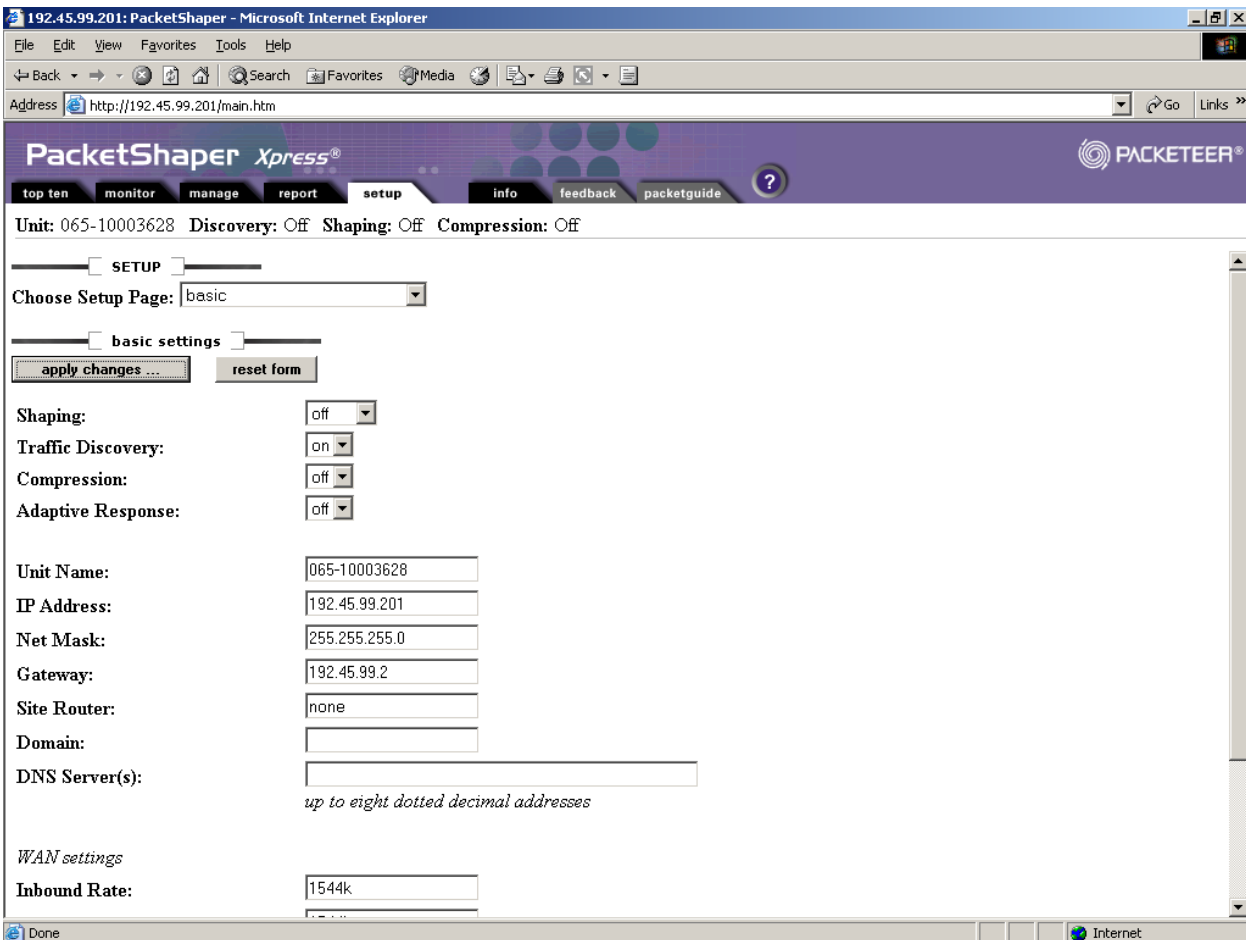
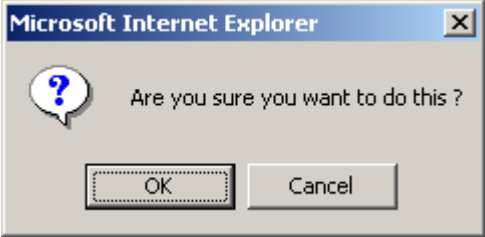
3.4. Bandwidth Partition for the RTP Traffic Class and Rate Policy for Individual RTP Flows

A partition specifies the total bandwidth reserved for all traffic within a traffic class. A partition applied to the RTP traffic class thus reserves a portion of the WAN link capacity for all RTP traffic. When combined with a rate policy (described below) on individual RTP flows, the partition effectively sets a maximum on the number of concurrent RTP flows going across the WAN link. Bandwidth above the reserved partition amount cannot be guaranteed, even with burstable limits defined.

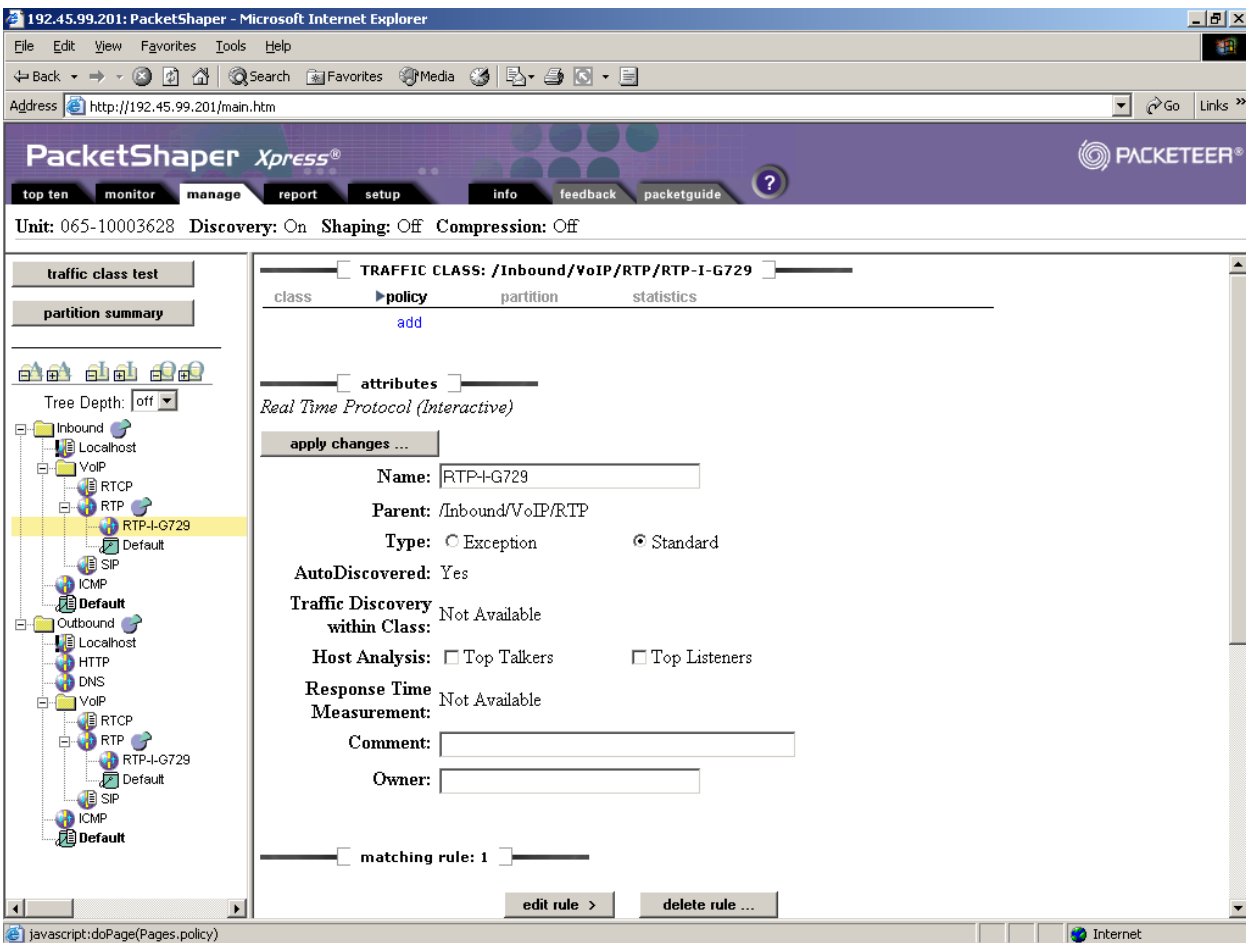
Since the audio (RTP) portion of a VoIP connection typically requires a constant bit rate, a rate policy should be applied to individual RTP flows within the RTP traffic class. A rate policy specifies the amount of bandwidth reserved for each RTP flow, and when properly defined, ensures that each RTP flow going to the WAN router receives enough bandwidth.

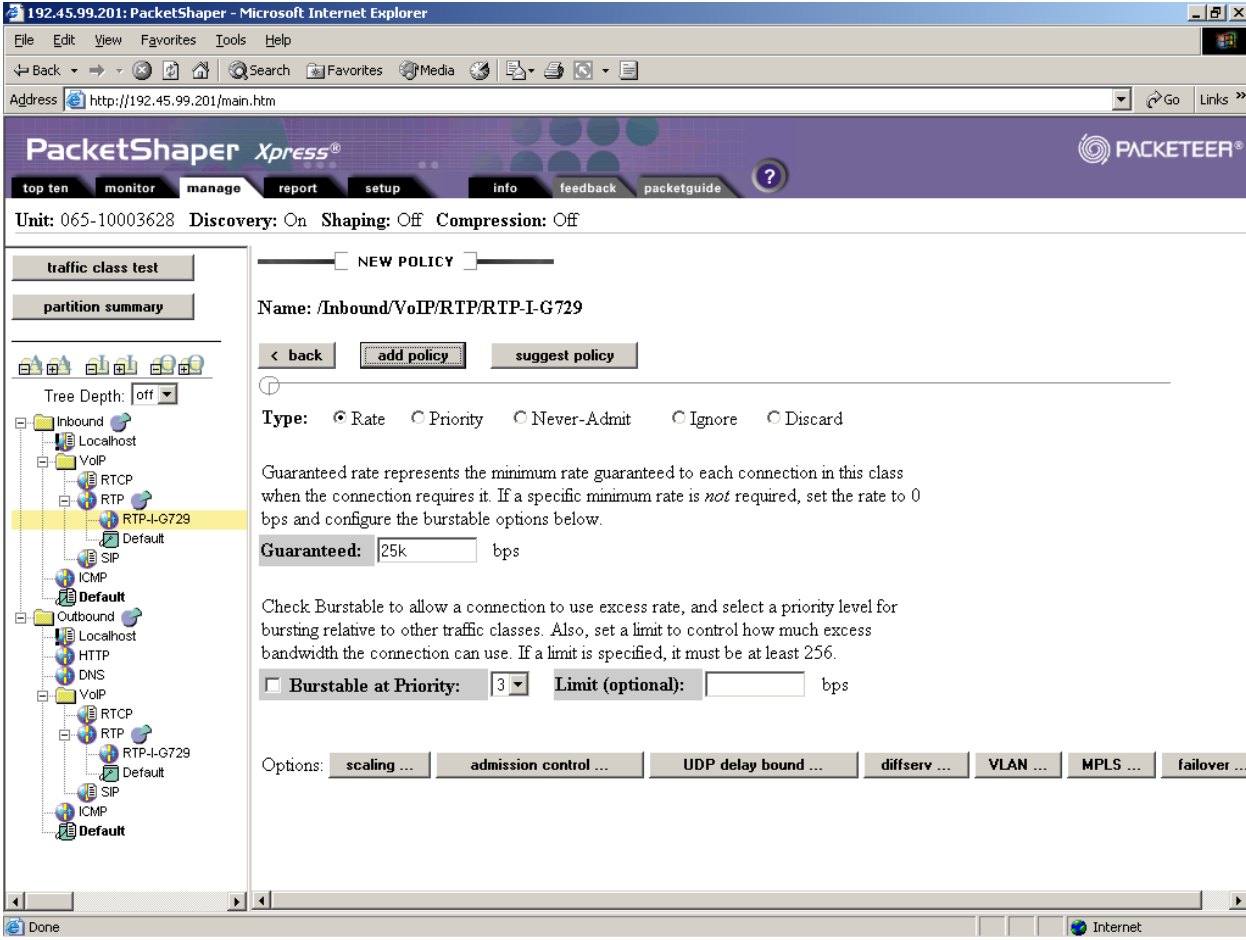
Step	Description
1.	<p>In the manage tab, click on the inbound RTP traffic class in the left panel and select partition->add.</p>  <p>The screenshot shows the PacketShaper Xpress web interface in a Microsoft Internet Explorer browser window. The address bar shows 'http://192.45.99.201/main.htm'. The interface has a navigation bar with tabs: 'top ten', 'monitor', 'manage' (selected), 'report', 'setup', 'info', 'feedback', and 'packetguide'. Below the navigation bar, it displays 'Unit: 065-10003628', 'Discovery: Off', 'Shaping: Off', and 'Compression: Off'. On the left side, there is a tree view under 'traffic class test' and 'partition summary'. The tree view shows a hierarchy: Inbound > Localhost > VoIP > RTP (selected). The main panel shows the configuration for the selected 'RTP' traffic class. It includes a 'TRAFFIC CLASS: /Inbound/VoIP/RTP' header, a 'partition' tab, and an 'add' button. Below this, there is an 'attributes' section with the following fields: 'Name: RTP', 'Parent: /Inbound/VoIP', 'Type: Standard' (selected), 'AutoDiscovered: No', 'Traffic Discovery within Class: Enabled (warning: global switch is off)', 'Host Analysis: Top Talkers' and 'Top Listeners' (both unchecked), 'Comment:', and 'Owner:'. There is also a 'matching rule: 1' section with 'edit rule >' and 'delete rule ...' buttons. At the bottom, it shows 'Device: any' and 'Protocol: UDP'. The status bar at the bottom of the browser window shows 'javascript:doPage(Pages,part)' and 'Internet'.</p>

Step	Description
2.	<p>Enter the Size (in bit rate or percentage) of the bandwidth partition for inbound RTP traffic and click on “add partition”. In the example below, approximately half of the inbound T1 link is reserved for RTP traffic.</p>  <p>The screenshot shows the PacketShaper Xpress web interface in Microsoft Internet Explorer. The address bar shows http://192.45.99.201/main.htm. The interface has a navigation bar with links: top ten, monitor, manage, report, setup, info, feedback, packetguide. Below the navigation bar, it shows Unit: 065-10003628, Discovery: Off, Shaping: Off, Compression: Off. The main area is titled "NEW PARTITION" and shows the Name: /Inbound/VoIP/RTP. There are buttons for < back, add partition, and a link to Go to Partition Summary. The Size is set to 750k bps, and the Limit is set to bps. A checkbox for Burstable is present. A text box explains: "Specify a 'size' to reserve bandwidth for all traffic defined by the class and its non-partitioned children. The size can be zero. Set the 'burstable' option to allow a partition to borrow available bandwidth from other partitions, up to the 'limit' you define. If a limit is specified, it must be at least 1000 bps." Below this is a section for Dynamic subpartition with a details link and a note: "(none) Specify subpartition sizing to create dynamic subpartitions for traffic flows per address or subnet basis. Click on details for full programming features." The status bar at the bottom shows the URL http://192.45.99.201/partsum.htm and an Internet icon.</p>
3.	Repeat Steps 1 – 2 for the outbound RTP traffic class.

Step	Description
4.	<p>Click on the setup tab. Set Traffic Discovery to “on” and click on “apply changes ...”.</p> 
5.	<p>Click on “OK” to confirm the change.</p> 

Step	Description																																																																																																																																																																																																																																																													
6.	<p>Click on the monitor tab. Place several SIP phone calls across the WAN link until the codec(s) is (are) discovered and displayed on the traffic monitor. Click on the update button as necessary.</p> <div><div>192.45.99.201: PacketShaper - Microsoft Internet Explorer</div><div><div>File Edit View Favorites Tools Help</div><div><div>Back Forward Stop Reload Home Search Favorites Media Print</div><div>Address http://192.45.99.201/main.htm Go Links</div></div><div><div>PacketShaper Xpress</div><div>PACKETEER</div></div><div><div>top ten monitor manage report setup info feedback packetguide</div></div><div>Unit: 065-10003628 Discovery: On Shaping: Off Compression: Off</div><div><div>MONITOR TRAFFIC</div><div>Click "clear stats ..." to reset values shown in GREEN. Sep 21 2005 - 17:00:51</div><div>Monitor: Traffic Display: All classes clear stats ... update Auto (Off) Stop Auto</div><div>Last cleared: Sep 21 - 16:59:22</div><div>Tree Depth: off</div><table><thead><tr><th>Traffic Class Name</th><th>Report</th><th>Class Hits</th><th>Policy Hits</th><th>Current (bps)</th><th>1 Min (bps)</th><th>Peak (bps)</th><th>Guar. Rate</th><th>Pkt Exch (ms)</th><th>Partition Min-Max</th><th>Policy Type (Pri.) Guar.</th></tr></thead><tbody><tr><td>Inbound</td><td></td><td></td><td></td><td>24k</td><td>63k</td><td>277k</td><td>0</td><td>NA</td><td>uncommitted - none</td><td></td></tr><tr><td> Localhost</td><td></td><td>382</td><td>382</td><td>18k</td><td>40k</td><td>223k</td><td>0</td><td>3</td><td></td><td>Priority (6)</td></tr><tr><td> VoIP</td><td></td><td></td><td></td><td>24k</td><td>23k</td><td>27k</td><td>0</td><td>NA</td><td></td><td></td></tr><tr><td> RTCP</td><td></td><td>0</td><td>0</td><td>395</td><td>169</td><td>605</td><td>0</td><td>NA</td><td></td><td>Priority (7)</td></tr><tr><td> RTP</td><td></td><td></td><td></td><td>24k</td><td>22k</td><td>25k</td><td>0</td><td>NA</td><td>750k - nonburstable</td><td></td></tr><tr><td> RTP-I-G729</td><td></td><td>2</td><td>NA</td><td>24k</td><td>12k</td><td>25k</td><td>0</td><td>NA</td><td></td><td></td></tr><tr><td> Default</td><td></td><td>1</td><td>NA</td><td>35</td><td>3003</td><td>25k</td><td>0</td><td>NA</td><td></td><td>Priority (7)</td></tr><tr><td> SIP</td><td></td><td>0</td><td>0</td><td>0</td><td>20</td><td>4101</td><td>0</td><td>NA</td><td></td><td></td></tr><tr><td> ICMP</td><td></td><td>10</td><td>NA</td><td>1</td><td>8</td><td>290</td><td>0</td><td>NA</td><td></td><td></td></tr><tr><td> Default</td><td></td><td>20</td><td>23</td><td>18</td><td>48</td><td>3613</td><td>0</td><td>50</td><td></td><td>Priority (3)</td></tr><tr><td>Outbound</td><td></td><td></td><td></td><td>25k</td><td>40k</td><td>156k</td><td>0</td><td>NA</td><td>uncommitted - none</td><td></td></tr><tr><td> Localhost</td><td></td><td>382</td><td>382</td><td>1980</td><td>12k</td><td>101k</td><td>0</td><td>29</td><td></td><td>Priority (6)</td></tr><tr><td> HTTP</td><td></td><td>1</td><td>NA</td><td>0</td><td>0</td><td>0</td><td>0</td><td>NA</td><td></td><td></td></tr><tr><td> DNS</td><td></td><td>3</td><td>NA</td><td>1</td><td>10</td><td>355</td><td>0</td><td>NA</td><td></td><td></td></tr><tr><td> VoIP</td><td></td><td></td><td></td><td>24k</td><td>21k</td><td>25k</td><td>0</td><td>NA</td><td></td><td></td></tr><tr><td> RTCP</td><td></td><td>1</td><td>1</td><td>69</td><td>80</td><td>1251</td><td>0</td><td>NA</td><td></td><td>Priority (7)</td></tr><tr><td> RTP</td><td></td><td></td><td></td><td>24k</td><td>21k</td><td>25k</td><td>0</td><td>NA</td><td>750k - nonburstable</td><td></td></tr><tr><td> RTP-I-G729</td><td></td><td>2</td><td>NA</td><td>24k</td><td>12k</td><td>25k</td><td>0</td><td>NA</td><td></td><td></td></tr><tr><td> Default</td><td></td><td>1</td><td>NA</td><td>20</td><td>2989</td><td>25k</td><td>0</td><td>NA</td><td></td><td></td></tr><tr><td> SIP</td><td></td><td>0</td><td>0</td><td>0</td><td>0</td><td>0</td><td>0</td><td>NA</td><td></td><td>Priority (7)</td></tr><tr><td> ICMP</td><td></td><td>4</td><td>NA</td><td>0</td><td>0</td><td>0</td><td>0</td><td>NA</td><td></td><td></td></tr><tr><td> Default</td><td></td><td>24</td><td>27</td><td>253</td><td>383</td><td>6353</td><td>0</td><td>163</td><td></td><td>Priority (3)</td></tr></tbody></table></div><div>(1 item remaining) Downloading picture http://192.45.99.201/images/minusbot.gif... Internet</div></div></div>	Traffic Class Name	Report	Class Hits	Policy Hits	Current (bps)	1 Min (bps)	Peak (bps)	Guar. Rate	Pkt Exch (ms)	Partition Min-Max	Policy Type (Pri.) Guar.	Inbound				24k	63k	277k	0	NA	uncommitted - none		Localhost		382	382	18k	40k	223k	0	3		Priority (6)	VoIP				24k	23k	27k	0	NA			RTCP		0	0	395	169	605	0	NA		Priority (7)	RTP				24k	22k	25k	0	NA	750k - nonburstable		RTP-I-G729		2	NA	24k	12k	25k	0	NA			Default		1	NA	35	3003	25k	0	NA		Priority (7)	SIP		0	0	0	20	4101	0	NA			ICMP		10	NA	1	8	290	0	NA			Default		20	23	18	48	3613	0	50		Priority (3)	Outbound				25k	40k	156k	0	NA	uncommitted - none		Localhost		382	382	1980	12k	101k	0	29		Priority (6)	HTTP		1	NA	0	0	0	0	NA			DNS		3	NA	1	10	355	0	NA			VoIP				24k	21k	25k	0	NA			RTCP		1	1	69	80	1251	0	NA		Priority (7)	RTP				24k	21k	25k	0	NA	750k - nonburstable		RTP-I-G729		2	NA	24k	12k	25k	0	NA			Default		1	NA	20	2989	25k	0	NA			SIP		0	0	0	0	0	0	NA		Priority (7)	ICMP		4	NA	0	0	0	0	NA			Default		24	27	253	383	6353	0	163		Priority (3)
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Localhost		382	382	18k	40k	223k	0	3		Priority (6)																																																																																																																																																																																																																																																				
VoIP				24k	23k	27k	0	NA																																																																																																																																																																																																																																																						
RTCP		0	0	395	169	605	0	NA		Priority (7)																																																																																																																																																																																																																																																				
RTP				24k	22k	25k	0	NA	750k - nonburstable																																																																																																																																																																																																																																																					
RTP-I-G729		2	NA	24k	12k	25k	0	NA																																																																																																																																																																																																																																																						
Default		1	NA	35	3003	25k	0	NA		Priority (7)																																																																																																																																																																																																																																																				
SIP		0	0	0	20	4101	0	NA																																																																																																																																																																																																																																																						
ICMP		10	NA	1	8	290	0	NA																																																																																																																																																																																																																																																						
Default		20	23	18	48	3613	0	50		Priority (3)																																																																																																																																																																																																																																																				
Outbound				25k	40k	156k	0	NA	uncommitted - none																																																																																																																																																																																																																																																					
Localhost		382	382	1980	12k	101k	0	29		Priority (6)																																																																																																																																																																																																																																																				
HTTP		1	NA	0	0	0	0	NA																																																																																																																																																																																																																																																						
DNS		3	NA	1	10	355	0	NA																																																																																																																																																																																																																																																						
VoIP				24k	21k	25k	0	NA																																																																																																																																																																																																																																																						
RTCP		1	1	69	80	1251	0	NA		Priority (7)																																																																																																																																																																																																																																																				
RTP				24k	21k	25k	0	NA	750k - nonburstable																																																																																																																																																																																																																																																					
RTP-I-G729		2	NA	24k	12k	25k	0	NA																																																																																																																																																																																																																																																						
Default		1	NA	20	2989	25k	0	NA																																																																																																																																																																																																																																																						
SIP		0	0	0	0	0	0	NA		Priority (7)																																																																																																																																																																																																																																																				
ICMP		4	NA	0	0	0	0	NA																																																																																																																																																																																																																																																						
Default		24	27	253	383	6353	0	163		Priority (3)																																																																																																																																																																																																																																																				

Step	Description
7.	<p>Click on the manage tab and then the Inbound folder in the left panel. Click on the discovered codec and select policy->add.</p>
	 <p>The screenshot shows the PacketShaper Xpress web interface in a Microsoft Internet Explorer browser window. The address bar shows 'http://192.45.99.201/main.htm'. The interface has a navigation bar with tabs: top ten, monitor, manage, report, setup, info, feedback, and packetguide. Below the navigation bar, it displays 'Unit: 065-10003628', 'Discovery: On', 'Shaping: Off', and 'Compression: Off'. On the left side, there is a tree view under 'Inbound' showing folders for Localhost, VoIP, and Outbound. Under 'VoIP', there are sub-folders for RTPCP, RTP, and Default. The 'RTP' folder is expanded, and 'RTP-I-G729' is selected. In the main panel, the 'TRAFFIC CLASS: /Inbound/VoIP/RTP/RTP-I-G729' is shown. Below this, there are tabs for 'class', 'policy', 'partition', and 'statistics'. The 'policy' tab is active, showing an 'add' button. Below the 'add' button, there is a section for 'attributes' with the text 'Real Time Protocol (Interactive)'. An 'apply changes ...' button is present. The 'Name' field is set to 'RTP-I-G729', and the 'Parent' is '/Inbound/VoIP/RTP'. The 'Type' is set to 'Standard' (radio button selected). 'AutoDiscovered' is set to 'Yes'. 'Traffic Discovery within Class' is 'Not Available'. 'Host Analysis' has 'Top Talkers' and 'Top Listeners' checkboxes, both of which are unchecked. 'Response Time Measurement' is 'Not Available'. There are input fields for 'Comment' and 'Owner'. At the bottom, there is a 'matching rule: 1' section with 'edit rule >' and 'delete rule ...' buttons. The status bar at the bottom shows 'javascript:doPage(Pages.policy)' and 'Internet'.</p>

Step	Description
8.	<p>Set Type to “Rate” and enter the Guaranteed bandwidth for each inbound RTP flow (for example, each phone call). The per-call bandwidth depends on the codec, packet size, and number of frames per packet. For example, for G.729 codec packets with 20 ms packet size and 2 frames per packet, the calculated per-call bandwidth including overhead is approximately 25kbps. Click on “add policy”.</p>  <p>The screenshot shows the PacketShaper Xpress web interface in a Microsoft Internet Explorer browser window. The address bar shows http://192.45.99.201/main.htm. The interface has a top navigation bar with links like 'top ten', 'monitor', 'manage', 'report', 'setup', 'info', 'feedback', and 'packetguide'. Below this, there's a status bar showing 'Unit: 065-10003628', 'Discovery: On', 'Shaping: Off', and 'Compression: Off'. The main area is divided into a left sidebar with a tree view and a right main panel. The tree view shows a hierarchy of traffic classes: Inbound, Localhost, VoIP, RTP, RTP-I-G729, Default, SIP, ICMP, Outbound, HTTP, DNS, VoIP, RTP, RTP-I-G729, SIP, ICMP, and Default. The main panel shows the configuration for the selected policy, /Inbound/VoIP/RTP/RTP-I-G729. It includes buttons for '< back', 'add policy', and 'suggest policy'. The 'Type' is set to 'Rate' (radio button selected). Below this, there's a description of the 'Guaranteed' rate and a text input field for 'Guaranteed' set to '25k' bps. There's also a section for 'Burstable at Priority' with a dropdown set to '3' and a 'Limit (optional)' field. At the bottom, there are several option buttons: 'scaling ...', 'admission control ...', 'UDP delay bound ...', 'diffserv ...', 'VLAN ...', 'MPLS ...', and 'failover ...'.</p>
9.	Repeat Steps 7 – 8 for the discovered outbound traffic codec.

3.5. Enable Traffic Shaping

To turn on traffic shaping, click on the **setup** tab, set **Shaping** to “On”, and click on “**apply changes ...**”.

192.45.99.201: PacketShaper - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Search Favorites Media Print

Address http://192.45.99.201/main.htm Go Links »

PacketShaper Xpress®

PACKETEER®

top ten monitor manage report **setup** info feedback packetguide ?

Unit: 065-10003628 Discovery: On Shaping: Off Compression: Off

SETUP

Choose Setup Page: basic

basic settings

apply changes ... reset form

Shaping: on

Traffic Discovery: on

Compression: off

Adaptive Response: off

Unit Name: 065-10003628

IP Address: 192.45.99.201

Net Mask: 255.255.255.0

Gateway: 192.45.99.2

Site Router: none

Domain:

DNS Server(s):

up to eight dotted decimal addresses

WAN settings

Inbound Rate: 1544k

4. Avaya Communication Manager and Avaya SIP Enablement Services

Basic administration of SIP stations, SIP trunking support, and SIP call routing in Avaya Communication Manager and Avaya SIP Enablement Services (SES) is assumed. In **Figure 1**, the Avaya SES Server routes calls originated by the registered SIP phones (IP telephones and softphones) in the enterprise and simulated WAN of **Figure 1**, to Avaya Communication Manager. Conversely, Avaya Communication Manager routes calls destined for the registered SIP phones to the Avaya SES server via SIP trunks. Note that for calls between the registered SIP phones, the call is routed from the Avaya SES server to Avaya Communication Manager and back to the Avaya SES server before terminating on the destination SIP phone.

5. Interoperability Compliance Testing

The interoperability compliance testing focused on assessing the impact that PacketShaper has on Avaya SIP VoIP traffic traversing a T1 link connecting an enterprise site to a simulated WAN. On PacketShaper, rate policies were applied for each RTP flow and bandwidth reservations were configured for the total RTP traffic. For SIP call control and RTCP traffic, bandwidth reservations were not made, but high priority policies were applied.

5.1. General Test Approach

The general approach was to attempt SIP phone calls between the enterprise and the simulated WAN with and without competing HTTP and FTP traffic. The competing FTP traffic traversed the WAN link and simulated an enterprise user performing an FTP download from a WAN host. An HTTP traffic generator was connected to the enterprise router (the Cisco 2900 router in **Figure 1**) and although the HTTP traffic did not traverse the WAN link, it did traverse the link between the enterprise router and the LAN. This arrangement allowed for more than 1.544 Mbps of HTTP traffic (in both directions) to go through the PacketShaper.

The main objectives were to verify that:

- Calls between SIP phones in the enterprise and in the WAN are successfully completed and maintained with good voice quality.
- Calls between an H.323 phone in the enterprise and a SIP phone in the WAN are successfully completed and maintained with good voice quality.
- Multiple SIP phone calls between the enterprise and the WAN up to the reserved amount are successfully completed and maintained with good voice quality.
- Non-VoIP traffic (HTTP and FTP) does not encroach upon the bandwidth reserved for SIP RTP traffic.
- SIP RTP traffic does not encroach upon the bandwidth reserved for other traffic.
- The solution is valid for G.711 and G.729 codecs.

5.2. Test Results

The test objectives of Section 5.1 were verified. With the appropriate bandwidth reservations and rate policies, PacketShaper was able to guarantee bandwidth for all calls up to the reservation amount during varying levels of competing HTTP and FTP traffic on the WAN link.

Calls above the expected number of calls allowed by the bandwidth reservation were established, but without audio. Since SIP call control traffic was assigned a high priority policy in PacketShaper, SIP signaling packets were able to traverse the WAN link unimpaired, and thus successfully establish the additional calls. However, the total bandwidth reservation for RTP traffic combined with the per-flow rate policy for RTP traffic limited the number of RTP flows, and thus the RTP (audio) streams of the additional calls were denied.

6. Verification Steps

The following steps may be used to verify the configuration:

- On PacketShaper, verify that the partitions, rate policies, and priority policies are configured correctly.
- From each side of the WAN link, ping SIP endpoints (Avaya SES server, Avaya SIP Telephones, and Avaya SIP Softphones) on the other side of the link.
- Place SIP phone calls across the WAN link and verify good voice quality when the WAN link is unsaturated and saturated with competing traffic.

7. Support

For technical support on the Packeteer PacketShaper, consult the support pages at <http://www.packeteer.com/support> or contact Packeteer Technical Support at 408-873-4550.

8. Conclusion

These Application Notes described the procedures for configuring the Packeteer PacketShaper to guarantee WAN link bandwidth to SIP Voice over IP (VoIP) RTP traffic generated by Avaya SIP Telephones and Softphones, and give high priority to SIP VoIP signaling traffic generated by Avaya SIP Enablement Services (SES) servers, and Avaya SIP Telephones and Softphones. During compliance testing, SIP phone calls traversing the WAN link were successfully established and maintained while non-VoIP traffic such as HTTP and FTP traffic was sharing the WAN link.

9. Additional References

Product documentation for Avaya products may be found at <http://support.avaya.com>.

Product documentation for Packeteer products may be found at <http://www.packeteer.com/support>.

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