VIRTUAL ACCESS
Service Provider Solutions

Service Managed Gateway™

How to Configure Clearway Technology for Converged Voice and Data Services

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1 Introduction

Virtual Access Clearway technology enables data and voice traffic to share one DSL line. Clearway technology dynamically monitors the amount of voice traffic on the line and implements Quality of Service (QoS) settings according to the volume of traffic. The Clearway quality assurance mechanism ensures high voice quality and maintains standard data flows.

1.1 The benefits of Clearway technology

A call agent controls VoIP phones. To operate, the VoIP phones must be able to contact the call agent.

The Service Managed Gateway (SMG) can be configured as the following proxy servers.

- MGCP proxy server
- SIP proxy server

These configurations have the following advantages:

- You can put all the VoIP phones on the LAN, so the phones do not need live IP addresses.
- The firewall on the SMG protects the VoIP phones.
- Each phone is simply configured to request an IP address, which makes the phones easy to manage and deploy.
- Converged voice and data services are possible.

The SMG can also be configured as a Skinny Client Control Protocol overview (SCCP) application level gateway (ALG).

Clearway can also be applied to propriety VoIP protocols by analysing Real-time Transfer Protocol (RTP) traffic. This is very useful when tunnelling voice traffic in an IP PBX environment

1.2 How Clearway technology manages Quality of Service

Quality of Service is critical to VoIP. To implement QoS, Clearway technology:

- controls all VoIP calls with the proxy server;
- determines the amount of bandwidth that is required based on the volume of VoIP calls, and
- prioritises traffic through four Diffserv queues that are associated with virtual routes.

Each virtual route and its associated queue has a specific purpose.

Virtual Route (VR)	**Purpose**
First VR	Prioritises inbound VoIP traffic.
Second VR	Limits the volume of inbound data traffic.

Third VR	Prioritises outbound VoIP traffic.
Fourth VR	Limits the volume of outbound data traffic.

Table 1: The purpose of the virtual routes

The inbound queues are configured on the LAN interface (eth-0) of the SMG. The outbound queues are configured on the WAN interface (ppp-1) of the SMG.

The inbound traffic queues prioritise VoIP traffic. The inbound traffic queues limit non-VoIP traffic by limiting the bandwidth that is available for data traffic.

1.2.1 The script for voice bandwidth management

A script monitors the amount of voice data on the line at one time by monitoring the number of active voice calls. To do this, four parameters must be passed to the script when the script is scheduled. The parameters are:

- the maximum bandwidth,
- the bandwidth per call,
- the Diffserv queue that the script operates on, and
- the initial reservation of bandwidth for the first call.

If the maximum transmit rate is the total available bandwidth, when there is no voice call all the bandwidth is available to transfer data. When a voice call starts, bandwidth is allocated to the voice call and the amount of bandwidth for data is reduced. When the voice call ends, the amount of bandwidth for data increases back to the total available bandwidth. Figure 1 shows how the script manages bandwidth changes.

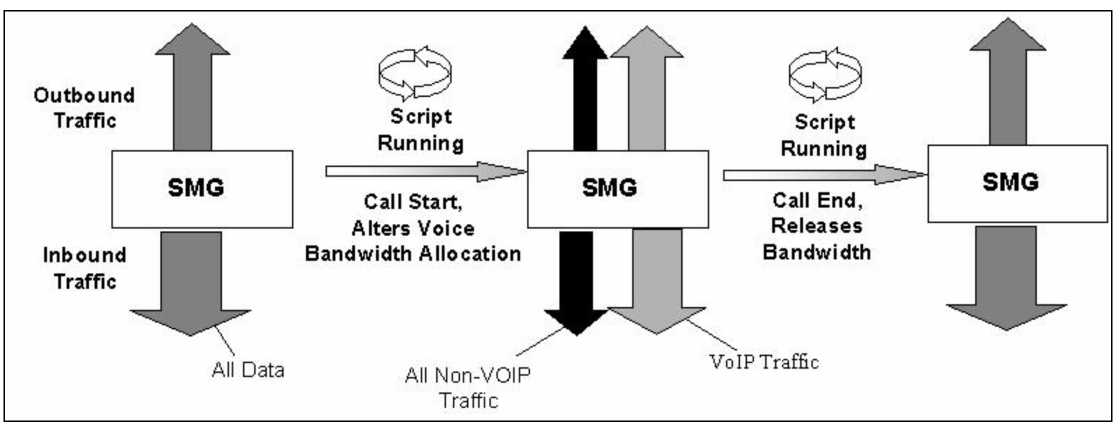

Figure 1: The function of the script for voice bandwidth management

The bandwidth that is available per call depends primarily on the VoIP codec that is used. If G.729 codec is used, the bandwidth per call is 50Kb on an ADSL line. If G.711 codec is used, the bandwidth per call is 110Kb.

The script also adjusts the size of the Transmission Control Protocol (TCP) window when a call is active. Adjusting the TCP window size improves any latency that is caused by bursts of inbound traffic when a TCP socket is first set up. Normally, the TCP window size is 65535kb. This means that a burst of traffic of 65535kb can be sent before there is any flow control. The burst temporarily causes a queue of data to be introduced at the ISP end of the link, and latency is introduced. Latency will lead to jitter on a phone call. But latency is avoided if the TCP window is adjusted before there is any TCP data flow.

Jitter is the term used to describe the degree of variance in the latency of the network. Transmitting high quality voice services over IP networks requires low degrees of jitter; total network latency is not as significant.

1.2.2 Reducing jitter

Two outbound queues help reduce jitter, but significant jitter can still be introduced into the outbound stream.

For example, an email with a large attachment is sent. Typically, this is sent in 1500-byte packets. As the SMG begins transmitting a 1500-byte packet on the ADSL interface, a VoIP packet arrives from the LAN. The VoIP packet is not transmitted until after the 1500-byte packet is transmitted. If the outbound bandwidth is 256Kbps, it takes (1500*8)/256000=46ms to transmit the 1500-byte package. This means that 46ms of jitter is introduced into the outbound traffic.

Jitter is reduced by limiting the maximum segment size of the outbound TCP packets. The maximum segment size (MSS) is a parameter in the TCP SYN packets that is used to negotiate the maximum packet sizes that are sent between the two ends of the TCP session. When the MSS is specified, the SMG dynamically modifies the MSS values on the TCP packets as they pass through the SMG.

The default MSS is 1460 bytes, plus 40 bytes for the TCP header and IP header. If the MSS is 730, the maximum packet size is 770 bytes. Maximum jitter is now (770*8)/256000 = 24ms. 24ms is a more acceptable jitter value.

2 Configuring Clearway technology

2.1 Preparing to configure Clearway technology

Before you configure Clearway VoIP, you must know how much bandwidth is available on the ADSL line. Use this information to decide the maximum transmission rate for the inbound traffic priority queue.

Suppose your ADSL line has 512Kb of bandwidth. Configure the four virtual routes and Diffserv queues with the settings shown in Table 2.

	Virtual Route			**Diffserv Queue**			
	Interface	**Redirect queue**	**Protocol**	**Interface**	**Priority**	**Minimum transmission rate**	**Maximum transmission rate**
Inbound traffic priority	Eth-0	1	UDP	Eth-0	1	0	0
Inbound traffic limitation	Eth-0	2	--	Eth-0	2	0	512000
Outbound traffic priority	PPP-1	3	UDP	AAL5-1 bound to PPP-1 in the configuration	1	0	0
Outbound traffic limitation	PPP-1	4	--	AAL5-1 bound to PPP-1 in the configuration	2	0	0

Table 2: Settings for an ADSL line with 512Kb of bandwidth

We recommend the settings in Table 2 for any Clearway configuration. If your bandwidth is a different size, the interface, redirect queue, protocol, priority settings, and minimum transmission rate stay the same; only the maximum transmission rate will be different.

2.2 Configuring DHCP options

2.2.1 Enable the DHCP server

1. In the Expert View of the SMG web, select **advanced configuration -> system -> local servers -> dhcp server-> system**.
2. On the DHCP Server page, select **yes** in the Enabled drop-down box.
3. Click **Update**.

2.2.2 Specify the DHCP options

1. In the Expert View of the SMG web, select **advanced configuration -> system -> local servers -> dhcp server-> dhcp options -> tftp server name**.
2. Click **Add** in the row of the file name that you want to specify.
3. On the DHCP Server TFTP Filename Entry page, select **yes** in the Configured drop-down list.
4. Type the name of the boot file in the Filename field.
5. Click **Update**.

2.3 Enabling the MGCP proxy server

1. In the Expert View of the SMG web, select **advanced configuration -> system -> local servers -> mgcp proxy server**.
2. On the MGCP proxy server page, select **yes** in the Enabled drop-down list.
3. Type the IP address of the call agent in the Call Agent IP Address fields.
4. Under Advanced, the relevant port numbers can be entered
5. Click **Update**.

2.4 Enabling the SIP proxy server

1. In the Expert View of the SMG web, select **advanced configuration -> system -> local servers ->clearway voip -> sip default proxy server**
2. On the SIP default proxy server page, select **yes** in the Enabled drop-down list.
3. Type the IP address of the call agent in the Call Agent IP Address fields.
4. Enter the relevant port numbers for the call agent and media gateway
5. Click **Update**.

2.5 Enabling the Skinny (SCCP) ALG

1. In the Expert View of the SMG web, select **advanced configuration -> system -> local servers -> clearway voip -> skinny application gateway**
2. On the MGCP Proxy Server page, select **yes** in the Enabled drop-down list.
3. Type the Skinny signalling port number in the Port field.
4. Click **Update**.

2.6 Configuring virtual routes and Diffserv queues for QoS

You must configure four virtual routes and associate Diffserv queues with each of them. We recommend that you configure the first virtual route and associate a Diffserv queue with it. Then configure the second virtual route and associate a Diffserv queue with it, and so on.

2.6.1 Add a virtual route

The priority of a virtual route is determined by its position in the virtual route list. We recommend that you add a route at row 10 or higher. If you add a route at row 10 or higher, you can add a virtual route with a higher priority in the future.

1. In the Expert View of the SMG web, select **advanced configuration -> system -> filters -> virtual routes**.
2. Click **Add** in the row of the virtual route that you want to add. Add a route at row 10 or higher.
3. On the Virtual Route Entry page, click **Advanced** to display the advanced virtual route options.
4. Specify the settings relevant to the traffic type that needs to be classified

Table 3 below lists the parameters that are required.

5. Click **Update**.

Option	Setting
Configured	Select **yes** from the drop-down list.
Name	Type a name for the virtual route. This name appears in the virtual route list.
Interface	Select **eth-0** from the drop-down list. Select ppp-1 for upstream virtual routes.
Redirect Interface	Select **eth-0** from the drop-down list. Select ppp-1 for upstream virtual routes.
Redirect Queue	Type the number of the Diffserv queue that you will associate with this virtual route.
Protocol	Select the relevant protocol from the drop-down list. UDP for voice and TCP for data.
Diffserv Value	If voice packets are marked (TOS Byte), enter the full Diffserv value here.
Diffserv Mask	Enter 255 under Diffserv Mask if a Diffserv Value has been entered.

Table 3: The settings for a virtual route

2.6.2 Add a Diffserv queue

1. In the Expert View of the SMG web, select **advanced configuration -> system -> Diffserv -> queue properties**.
2. Click **Add** in the row of the Diffserv queue that you want to add.
3. On the Queue Properties Entry page, click **Advanced** to display the advanced virtual route options.
4. Specify the settings that are listed in Table 4.
5. Click **Update**.

Option	Setting
Enabled	Select **yes** from the drop-down list.

Name	Type a name for the Diffserv queue. This name appears in the virtual route list.
Interface	Select **eth-0** from the drop-down list. Select aal5-1 for upstream queues.
Priority	Type the priority number for the queue.
Min Transmit Rate	Type the guaranteed data transfer rate for the queue.
Max Transmit Rate	Type the maximum data transfer rate for the queue.

Table 4: The settings for a new Diffserv queue

2.7 Configuring the script for voice bandwidth management

2.7.1 Install the script

1. In the Expert View of the SMG web, select **advanced configuration -> system -> scripts -> script editor**.
2. On the Script Editor page, paste in the script shown in section 2.7.1.1 for MGCP and the script shown in section 2.7.1.2 for SIP
3. Click **Update**.

2.7.1.1 The script for MGCP voice bandwidth management

```
!echo off
!arg maxbw, voicebw, queuenum, initdec
!unique
$port = $5
echo Monitoring voice traffic for dynamic bandwidth adjustment
!if port = ''
  $port = ppp-1
!endif
$x = 0
$curbw = $maxbw
$firststep = $initdec
  !while x < 10
  $ds($x) = $curbw
  !if $x = 0
    !sub curbw $firststep
  !else
    !sub curbw $voicebw
    !if $curbw < 0
      $curbw = 0
    !endif
  !endif
  !inc x
!endwhile
!while 1
  !we MGCP:DLCX|CRCX 60
  !endevent
  $numcalls = `wc -l mgcp connections`
  !dec numcalls
  !if numcalls < 1
```

```
    quiet set IP Address translation interface TCP window Adjustment
enabled $port,no
    quiet commit
  !endif
  !if numcalls = 1
    quiet set IP Address translation interface TCP window Adjustment
enabled $port, yes
    quiet commit
  !endif
  !if numcalls > 9
    $numcalls = 9
  !endif
  !if $curbw <> $ds($numcalls)
    $curbw = $ds($numcalls)
    quiet set Diffserv queue token bucket maximum average rate $queuenum
$curbw
    quiet commit
  !endif
!endwhile
```

2.7.1.2 The script for SIP voice bandwidth management

```
!echo off
!arg maxbw, voicebw, queuenum, initdec
!unique
[voicebw]
  !speed 20
  !echo off
  !arg maxbw, queuenum,port
  !unique
  $port = $5
  echo Monitoring voice traffic for dynamic bandwidth adjustment
  !if port = ''
    $port = ppp-1
  !endif
  $x = 0
  $curbw = 0
  !while 1
    !we SIP:ACK|BYE 60
    !endevent
    !pause 1
    $z = `find current_local_call_bandwidth sip stats`
    $localbw = $z[2]
    $z = `find current_active_call_bandwidth sip stats`
    $totalbw = $z[2]
    !sub totalbw, $localbw
    !mul totalbw, 1000
    !if totalbw < 1
      $tcpadjust = no
    !else
      $tcpadjust = yes
    !endif
    !if '`show ip Address translation interface tcp window adjustment
enabled $port`' <> '$tcpadjust'
      `set IP Address translation interface TCP window Adjustment enabled
$port,$tcpadjust`
      `commit`
```

```
     !endif
     !if $totalbw > $maxbw
         $totalbw = $maxbw
     !endif
     !if $curbw <> $totalbw
       $curbw = $totalbw
       $newbw = $maxbw
       !sub newbw, $totalbw
       !echo $newbw
       `set Diffserv queue token bucket maximum average rate $queuenum
$newbw`
       `commit`
     !endif
   !endwhile
```

2.7.2 Schedule the script

1. In the Expert View of the SMG web, select **advanced configuration -> system -> scheduler -> scheduler tasks**.
2. Click **Add** to add a task.
3. On the Scheduler Task Entry page, specify the settings that are listed in Table 5,
4. Click **Update**.

Option	**Setting**
Enabled	Select **yes** from the drop-down list.
Name	Type a name for the script. This name appears in the virtual route list. We recommend that you use the name **Voice management**.
Frequency	Select **startup** from the drop-down list.
Script	Type **voicebw 512000, 50000, 2, 50000,**

Table 5: The settings for script schedule

2.8 Configuring the maximum segment size to reduce jitter

1. In the Expert View of the SMG web, select **advanced configuration -> interfaces -> ppp-1 -> ip -> ip**.
2. On the IP Interface on ppp-1 page, click **Advanced** to display the advanced interface options.
3. In the TCP Largest MSS field, type the number of the largest maximum segment size.
4. Click **Update**.