# ARITARI
# VOICE OPTIMISATION

### Routing and Technology

## OVERVIEW

Aritari is the leader in Voice Optimisation and provides a unique and powerful range of software capabilities that measurably improve the quality of voice calls while reducing the network bandwidth requirements for delivery.

It's becoming common to deliver voice over the Internet, providing an even better reason to adopt Aritari. Our software helps voice deal with the challenges of the Internet namely packet loss and high latency.

- Simplify Routing
- Protect Voice Quality
- Up to 90% Bandwidth Reduction
- Voice Gateways

## ARITARI VOICE OPTIMISATION

As network strategies evolve it becomes harder to maintain good voice quality. Aritari has developed a Voice SD WAN capability that will support a simpler voice network, improved quality and at a reduced cost over any network strategy.

### Simplify Voice Routing

One of the largest challenges to voice delivery today relates to complex routing due to a separate voice and data provider and resulting support and quality challenges. Some of the main challenges include:

- Poor voice quality over data network
- Poor voice quality of Internet
- Voice routing issues between voice and data networks

Regardless of the size or complexity of your network, with Aritari you can direct all branch voice traffic to an Aritari CPE or VM driving all voice down a voice VPN through the network to a central head end (DC/HQ). This design eliminates complex voice routing while providing voice packets with software that improves voice quality even in challenging network conditions.

ARITARI
NEW WORLD NETWORKS

## Protect Voice Quality

Aritari has been designed to improve voice quality in the toughest bandwidth conditions. Our customers have thousands of branches with limited bandwidth or who struggle to get consistent voice quality from the Internet to their chosen VOIP provider.

Aritari delivers two specific capabilities to address poor network conditions including:

- Forward Error Correction
- Packet Loss Correction

Forward Error Correction is a recognized technique which sees Aritari sending single voice packets multiple times when packet loss is detected in the network. This allows Aritari to reduce the effects of bandwidth congestion in an MPLS or Internet network.

Packet Loss Correction is based on the Aritari proprietary VPN tunnel (not TCP IP) which is more efficient at delivering lost packets in the network than TCP IP. This ensures that when congestion occurs the network is able to resend and accept lost packets faster, improving voice quality.

## Voice Compression

Aritari reduces the bandwidth requirements to deliver your voice by up to 90% over traditional TCP IP networks. This allows customers with large branch networks to deliver strong quality voice minutes over limited asymmetrical links like ADSL.

An example of this is reducing a VOIP call of 86kbps to 8.36kbps.

Aritari helps customers manage and reduce network and bandwidth cost without compromising on voice quality.

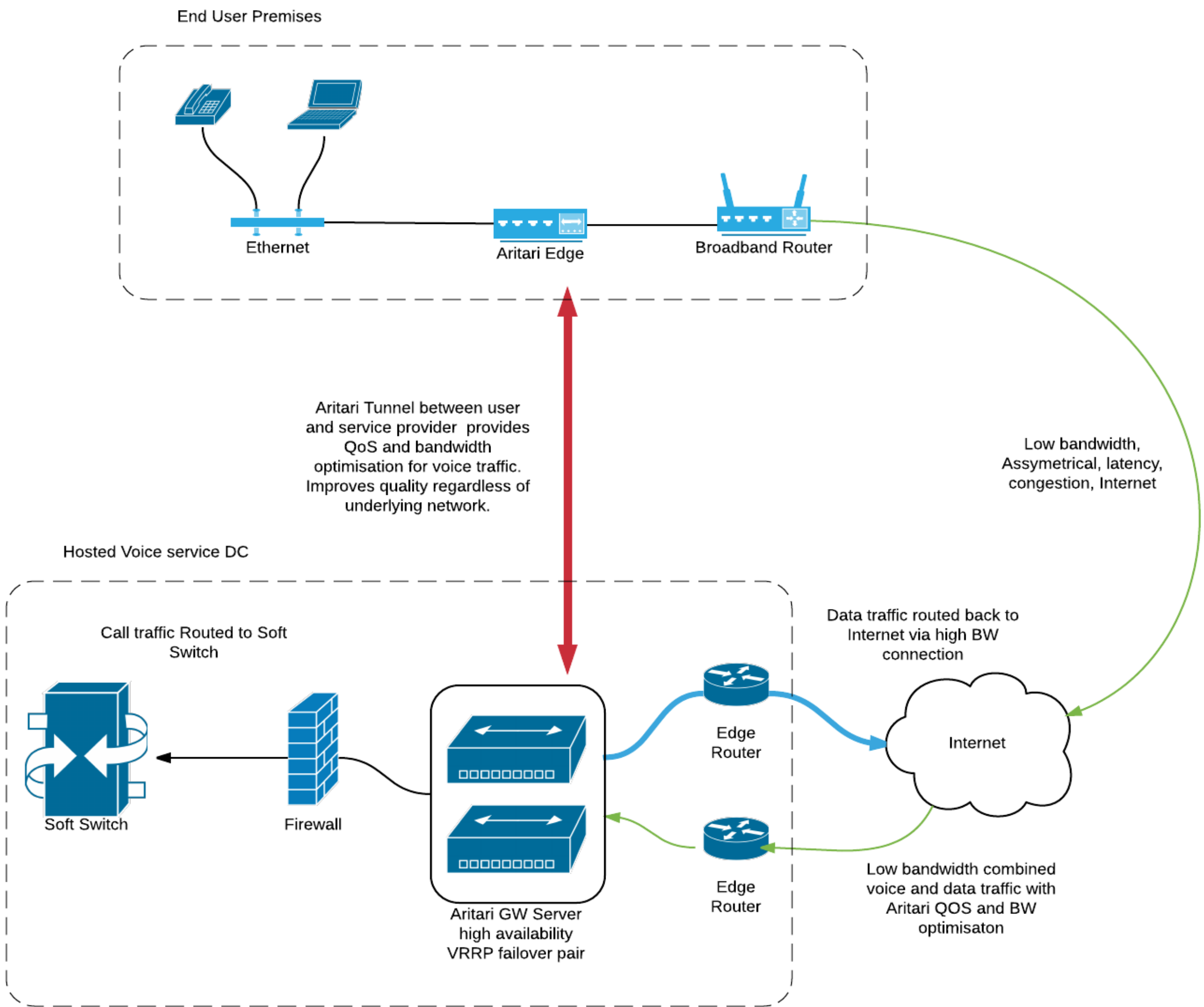
## VOIP Gateways

Aritari VOIP Gateways can be deployed to help clients aggregate their voice services to a central DC or to a public VOIP provider network anywhere in the world.

The VOIP Gateway extends the benefits of Aritari Voice Optimisation all the way to your VOIP provider or selected network point to ensure that you get the best network performance available.

Aritari also provide public managed VOIP Gateways in major locations in the world including the US and Europe.

Diagram 1.1 VOIP Gateway Deployment




## Additional Features Included

VOIP Optimisation from Aritari includes the following additional technology:

- Quality of service parameters
- Link bonding and bandwidth aggregation
- Transparent failover

## Technical Challenges – SD Voice

This section describes some of the current challenges facing organisations and VOIP providers in delivering quality voice over the Internet.

Main barriers to deploying VoIP include:

- Excessive bandwidth consumption of even so-called efficient CODECs such as G.729.
- High load on transit routers due to the large number of packets per second involved where there are many calls being carried
- Latency and jitter which arises as the result of larger data packets using the same links (even evident if the only data traffic is session data such as that provided by SIP)
- The cost of high bandwidth WAN links which are needed to solve these issues using traditional methods
- Lack of CODEC support in devices
- Complexities involved for enterprises wishing to deploy VoIP between sites across the public Internet or non-private links
- The cost of providing backup solutions to avoid the WAN link being a single point of failure

The features of Aritari SD Voice which eliminate or significantly reduce these barriers:

- Bandwidth used by voice is reduced by as much as ten times
- Jitter introduced using router queues is reduced to virtually zero
- Classes of data can receive as little as 0.4kbits/s
- Interactive traffic remains responsive
- There is no need to reduce the maximum transmission unit (MTU) of the WAN
- Aritari SD Voice optionally supports real-time and invisible transcoding of G.711 to more efficient CODECs with higher MOS scores than G.729 (the most widely supported low bandwidth CODEC)
- Backup links can be switched to in less than a second and without losing calls in progress
- Multiple links can be combined to both increase bandwidth available and eliminate single points of failure
- Sites can be privately linked across the public Internet

## Bandwidth Consumption and Router Load

The essential portion of a voice packet is normally very small. For example, a 20mS G.729 encoded voice packet contains only 20 bytes of useful data, transmitted at a rate of 50 packets per second.
This results in:
**50 x 20 x 8 bits per second = 8000 bits per second (8kbit/s).**

To be able to carry this packet across an IP network you also need some additional data, which is broken down in the table below:

| Protocol | Bytes | Description |
|---|---|---|
| IP | 20 | Standard data required by all Internet traffic. Source/Destination/Length etc. |
| UDP | 8 | Source and destination port etc. |
| RTP | 12 | Media type, timestamp, source ID, etc. |

Adding all of this to the standard 20mS G.729 packet our data rate now becomes:

**(20 + 20 + 8 + 12) x 50 x 8 = 24 000 (24kbit/s)**

Now a voice call is using three times more bandwidth than it needs to. In addition, at layer 2, or the network layer, there is more information required to transfer data packets across physical links.
At best, this link will be Ethernet, which "only" adds another 14 bytes for every packet (making our G.729 voice channel use 29.6kbits/sec.), however ATM based technologies such as most broadband links are less efficient. These networks use fixed sized "cells" of data to carry information, and typically these cells are 53 bytes in length. The G.729 packet above would require two such cells, and now have:

**2 x 53 x 50 x 8 = 42400 (42kbit/s)**

Therefore each G.729 voice channel that is carried across a broadband link consumes 42.4kbit/s of bandwidth, only 8kbit/s being used for the information that we really want to be transmitted.

How Aritari SD Voice solves this problem:

The information that is added to a voice packet to transmit it across a network is mostly superfluous. This is because all of it is either fixed for the entire duration of the stream, (such as destination address and port), or can be worked out from previous packets, (such as sequence numbers and time stamps). SD Voice creates a VPN tunnel between two SD Voice enabled devices. Within this tunnel, voice packets from multiple channels are combined into a single data stream, which has several effects:

- SD Voice only needs to send one set of IP and UDP headers for each "super-packet" that it sends

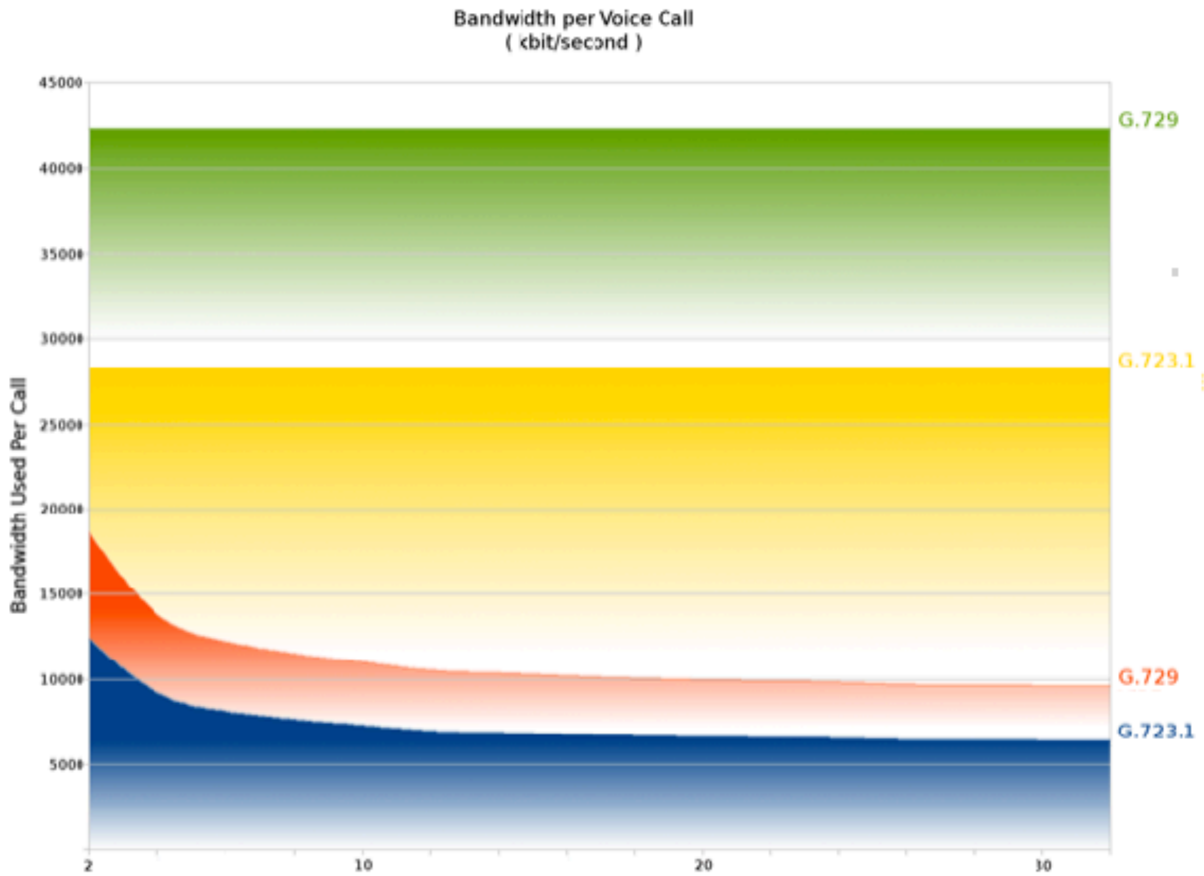
- It allows SD Voice to not transmit the superfluous information for each voice channel at all. In fact, the SD Voice system is so efficient that the total overhead for each channel within the stream is 2.287 bits
- The cell padding effect which is present on broadband networks is eliminated.

All this processing is entirely transparent to other devices on the network since the original data streams are reconstructed before they are sent on their way from the receiving SD Voice device. The only effects are a much lower bandwidth consumption and far fewer packets per second across the WAN, reducing load on routers and the possibility of network congestion.

The following table illustrates just how much bandwidth is saved using the G.729 CODEC across an ADSL network.

| Simultaneous Calls | Bandwidth w/o ViBE | Bandwidth With ViBE |
|---|---|---|
| 1 | 42.4 kbit/s | 29.2 kbit/s |
| 10 | 424 kbit/s | 110 kbit/s |
| 50 | 2.12 Mbit/s | 468.8 kbit/s |
| 100 | 4.24 Mbit/s | 982.4 kbit/s |

To illustrate the bandwidth savings that this represents, below is a graph which shows the average bandwidth used per call, including all overheads, when sent over a broadband network.




## Latency and Jitter Reduction

The Problem

This section could easily be summarised as “traditional traffic shaping does not work at low bandwidths.” It is also the area that all competing technologies fail to address. Essentially, traditional quality of service (QoS) paradigms treat voice as just another class of traffic, they may give it priority over other types of packet as best they can, but they do this in the same way that they would when prioritising any other class. The problem is that voice is not the same as any other class of traffic, it has specific requirements in terms of latency and jitter which need to be honored.

As an illustration, consider a 256kbit WAN link. This link has typical QoS techniques applied, and voice packets are prioritised.
Look at the illustration below:

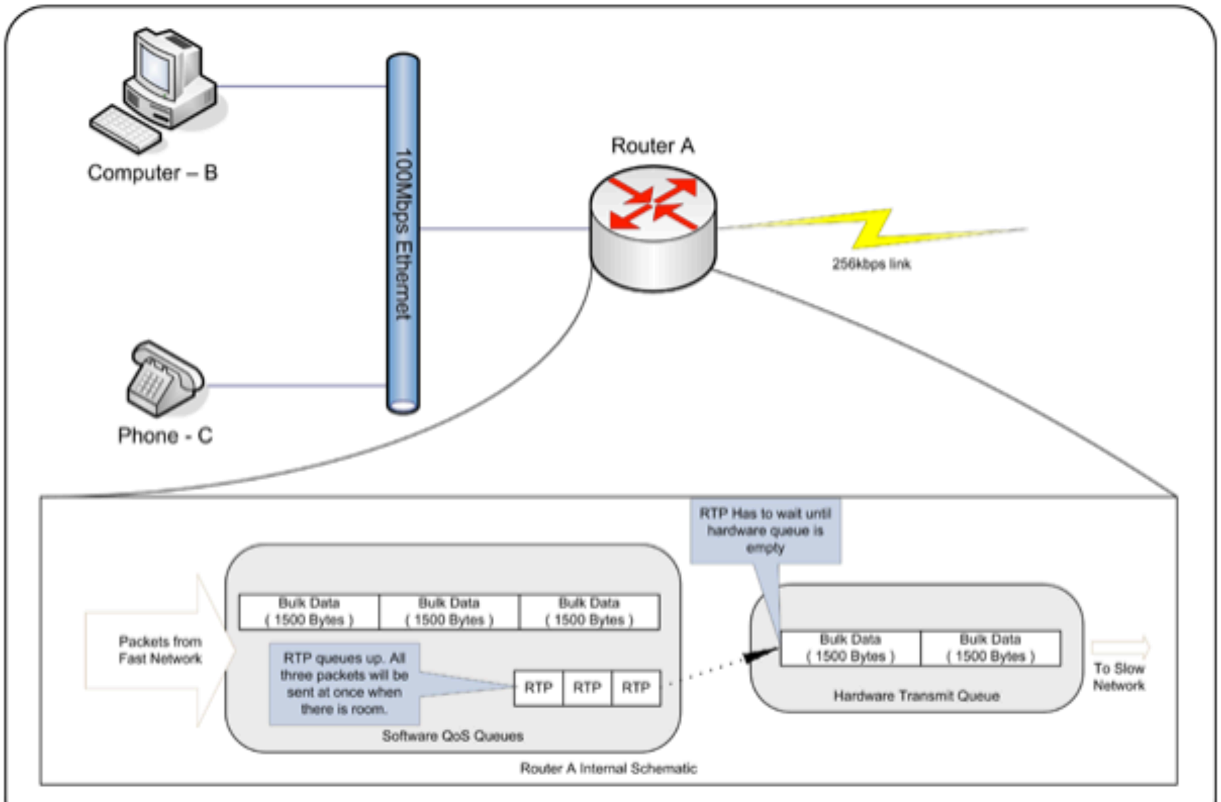



Illustration of Standard QoS

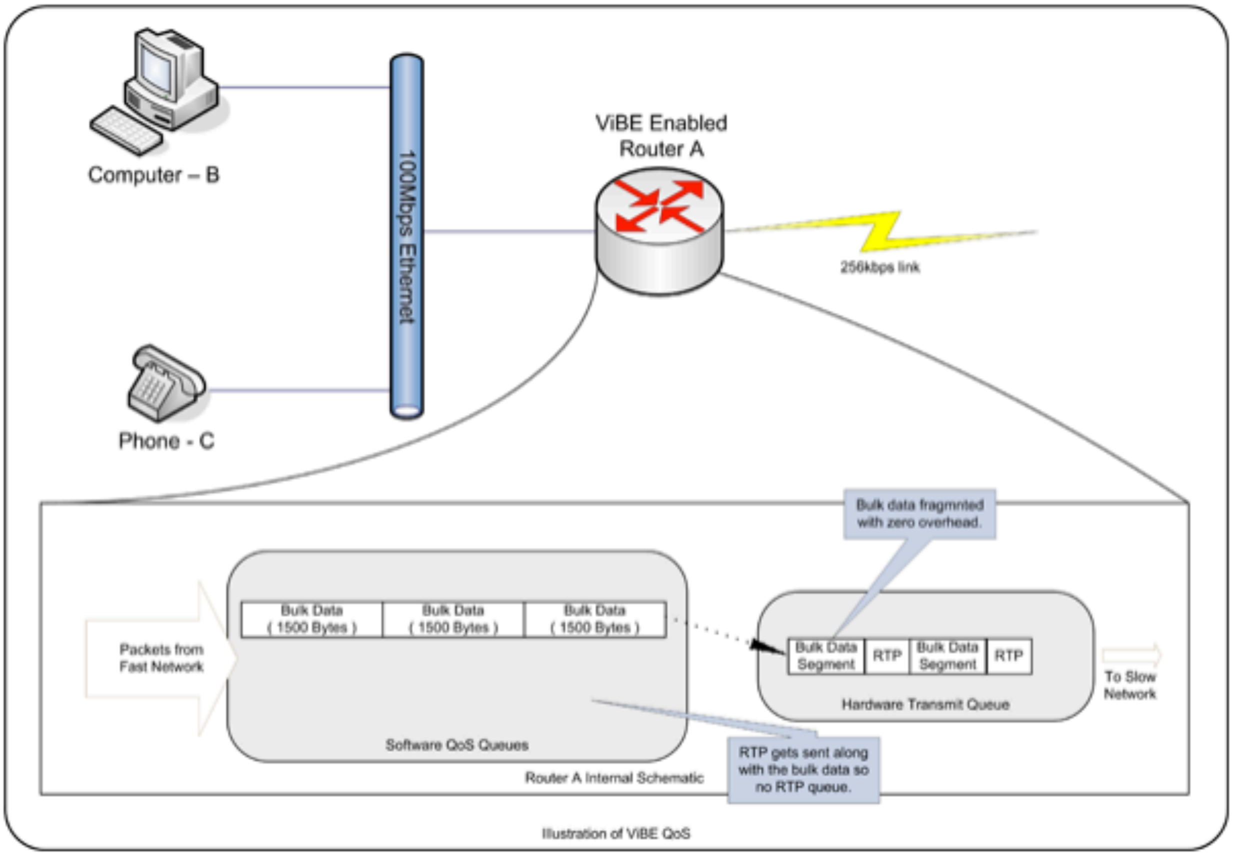
Data enters the router (A) from a 100Mbit/s LAN interface. A computer (B) on the LAN is sending an e-mail with a large attachment, and since the LAN is 100Mbit/s his computer sends many large (typically 1500 byte) packets to router A in quick succession.
The router will continue to queue packets for sending to the WAN until its buffer fills up. At the same time, another device © is being used for a voice call. It is sending small (60 byte ) packets every 50th of a second. Router A allocates a separate buffer to these packets, and when a decision needs to be made as to which packets to send to its hardware transmit buffer, it will send the packets from C first.
However, that decision is only made when there is a space in the hardware buffer, which only happens when a previously queued packet is sent. What all of this means is that the chances are that there will already be two 1500 byte packets in the hardware buffer at the time when a voice packet really should be sent on the WAN. At 256kbit/s, these 3000 bytes will take 94mS to send, meaning that as many as four voice packets could queue up waiting for a space on the network.

The normal way of making this situation a little better is to reduce the maximum size of packets that can be sent on the WAN. The drawback of this approach is that the WAN is then made less efficient, and in any case the smallest “maximum packet size” allowed is 576 bytes, cutting the router induced jitter to around 36mS, still nearly twice the amount of time allocated to a single voice packet.

How SD Voice Solves This Problem

SD Voice technology approaches this problem differently. The SD Voice tunnel created between enabled devices only sends packets at set intervals, which at 256kbit/s will be the same interval that voice packets are being sent. SD Voice knows how much bandwidth is available on the link, and so it also knows how large its packets can be. Any space which is not allocated to voice will simply be used by portions of data packets as required, in a very efficient manner.

This is illustrated below:




## CONCLUSION

Aritari Voice Optimisation ensures you achieve the best results under the worst conditions. A powerful software stack that simplifies your voice routing, reduces your network and bandwidth costs and protects your voice quality from unexpected network congestion.