interwise

InterwiseConnect™

Quality of Service (QoS)

Version 7.2

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Overview

Interwise Connect is a true Voice Conferencing product aimed at offering a replacement for companies with various hosted conferencing services. As such, the product is required to provide end users with the high quality voice conferencing experience to which they are used.

A Voice Conferencing application has strong network quality requirements which a Best Effort IP Network usually cannot provide in reality, where different applications compete for the same finite network resources.

Why is VoIP (and Voice Conferencing based on VoIP) sensitive to network quality?

There are three main areas that affect voice quality over IP.

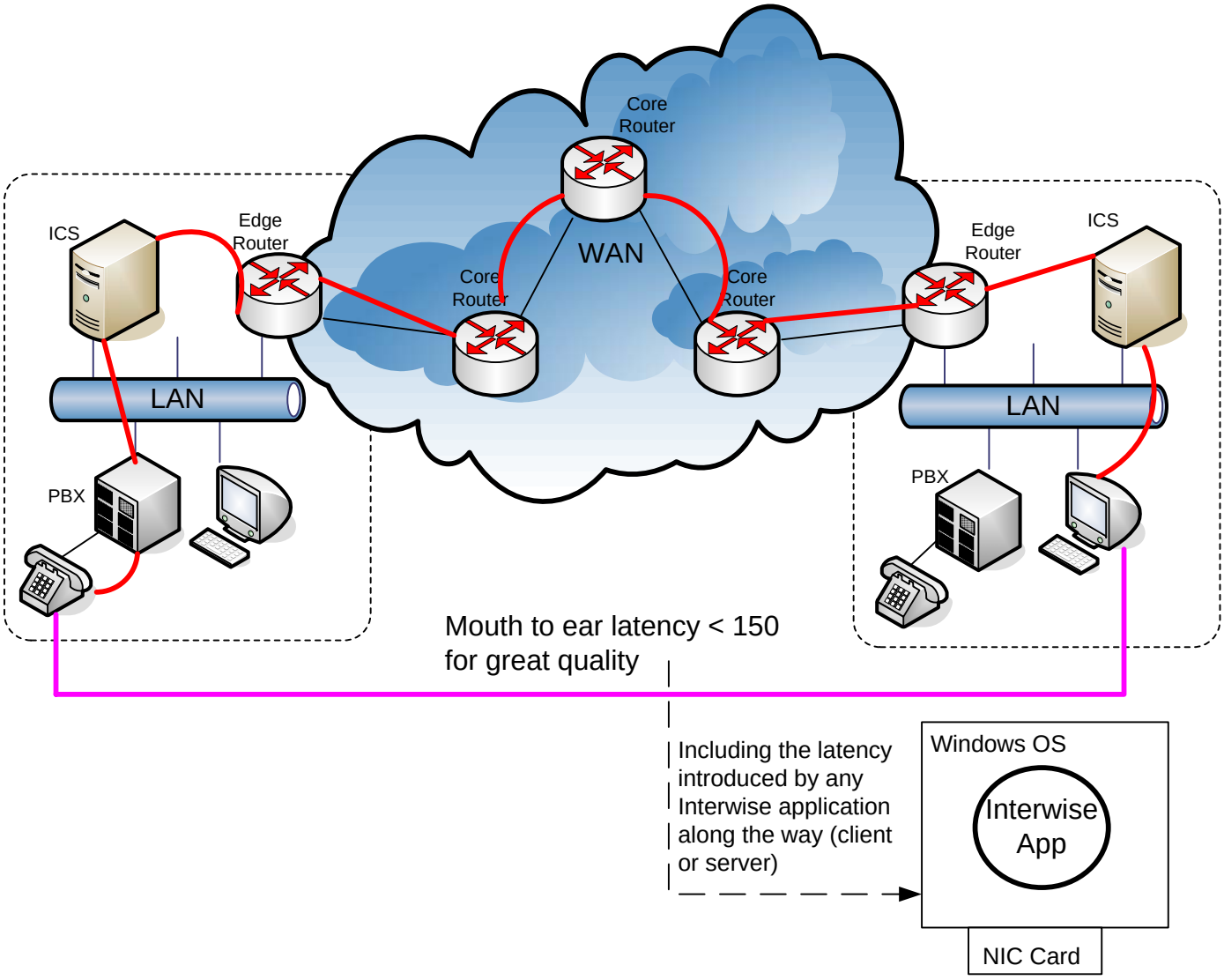
- **Latency:** ITU recommends (Recommendation G.114) the following delay for one way mouth to ear voice:
  - < 150 results in good quality and satisfies users (it is considered today that 200 ms gives the same result). PSTN meets these criteria.
  - 150-400 – acceptable quality. This means there is a percentage of users who are not satisfied. The recommendations suggest that at this delay level an administrator had to be aware of the situation (possibility of voice quality degradation).
  - > 400 – not acceptable.
- **Jitter:** Jitter or the delay variance between arriving packets should be under 30ms for good voice quality. Jitter that increased this value would cause cut offs / choppiness of the voice or if large jitter buffers (that compensate and smooth out jitter) are introduced they would cause the overall latency/delay time to increase, which would degrade voice quality.
- **Packet Loss:** It is recommended that packet loss is kept under 1% for high voice quality. Packet loss at a higher rate would degrade voice quality as UDP does not retransmit packets and with many packets missing the voice will be choppy. If TCP is used as the transmission protocol instead of UDP, packet loss will trigger TCP's congestion control mechanisms – namely a slow start and retransmit which will introduce considerable delays, thus degrading voice quality.

The real challenge is in providing high voice quality over an IP network when the IP network was designed as a Best Effort delivery network without service guarantees.

When many applications compete for the same limited network resources, situations arise where routers simply cannot send out packets fast enough. This can be due to outbound bandwidth limitation or CPU limitation. In order to handle this, routers maintain queues where packets are kept until the router can send them. These queues are the ones introducing latency and jitter and when queue limits are reached, packets are dropped and create the packet loss conditions.

The real world cases where these types of congestions occur are in the network links that connect local LAN segments with the WAN. As WAN resources are much more limited and expensive than LAN resources, the relay of packets from LAN to WAN often hits congestion situations where packets are queued or in extreme cases dropped. However, congestion on LAN resources may still occur, though not as often.

On the LAN there is another issue to consider, when if all packets are treated equally in a Best Effort scheme, and a user sends a file on the network, the file packets which are large chunks of data will be ahead of voice packets in the router queues. So both the LAN and WAN poses challenges in keeping voice quality as high as possible.



What is QoS and how is it achieved?

Quality of Service (QoS) refers to the mechanisms and practices implemented to guarantee services (applications) the network resources they require, such as bandwidth and latency.

QoS usually refers to network QoS and the network level practices. The major practices are:

- **Classification and Marking:** A service stream is identified (according to IP, port, Diffserv code, etc) then marked and classified as associated to a specific class of service. The classification is usually done on the WAN edge node routers (though can be done on local LAN routers as well to provide QoS on local LAN) and marking can be done on local routers or at the application/service level.
- **Queuing and Prioritization:** Once service streams are separated to classes of service, different treatment can be applied to different classes through different queues. Packets belonging to a specific class of service will be placed in a relevant queue. The different queues enable a router to prioritize between the various classes of service. Real time interactive voice is usually implemented using a priority queue.

It is important to understand that these are all network layer QoS practices and not at the application level. The need from the application level is explained in the following section.

Interwise support for QoS

Stream Separation

Separate Streams

Interwise enables the option to send packets using separate streams for voice, video and data.

Each server in the Interwise network will define whether a client/child PX connection to it, will be established using a separate stream for voice, video and data and on which port. A client/child PX receiving this information will establish relevant connection to the parent server using the destination ports specified by the parent server.

The server itself will retrieve the information from the **ISM** as part of the standard machine level details it already gets.

- **Data stream:** will carry the 'data' information which means application sharing, notes, etc. and basically any information that is not voice or video including all session control information (such as login, presenting rights, etc)
- **Voice Stream:** will carry only voice packets.
- **Video Stream:** will carry only video packets.

Components

Separate streams apply to ALL Interwise live connections.

- PC to ICS
- ICS to ICS
- ITS to ICS – As Interwise sends only voice and some negligible amount of control data this does not need to be supported at this stage.

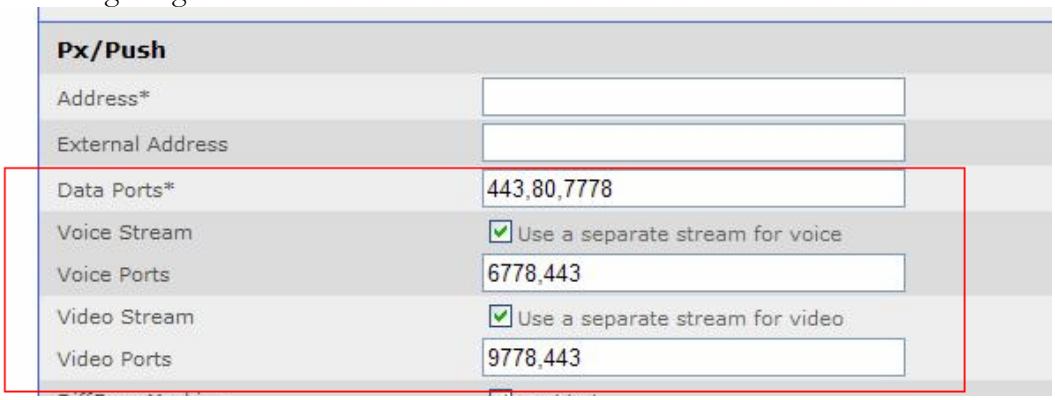
ISM Configuration

In the ISM you can define the following for each ICS:

1. Enable voice in a separate stream.
2. Enable video in a separate stream. This option is available only if voice is in a separate stream.
3. For each stream, define a port list. This port list will have existing default port numbers.

   NOTE: When only voice is enabled as a separate stream video shares the data stream.

The following image shows how to define streams in the server details in the ISM:

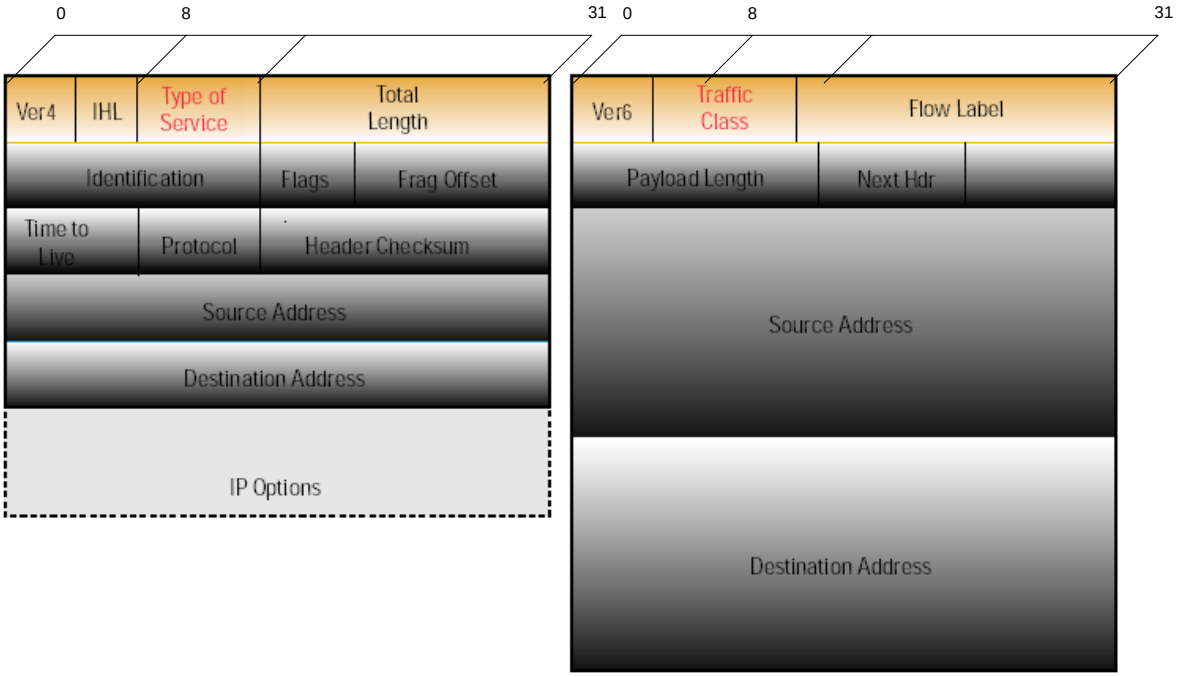


Diffserv Marking

Interwise supports Diffserv Marking in accordance to RFC 2475/2475.

Each stream is marked with its own Diffserv marking code point (DSCP), meaning that every packet will have a specific DSCP code (0-63) in its IP header TOS field.

IP V4 Header: IP V6 Header:



With Diffserv marking applied to Interwise traffic, a network administrator is able to identify/classify Interwise traffic on the routers and apply relevant prioritization.

DSCP Values

- **Voice Stream** - Interwise mark voice packets with the default DSCP of 46 (decimal)
- **Video Stream** - Interwise mark video packets with the default DSCP of 34 (decimal)
- **Data Stream** - Interwise mark data packets with the default DSCP of 26 (decimal)

Components

Diffserv marking applies to ALL Interwise live connections and for each node it applies to all sent packets, including packets sent to parent servers and child servers.

- PC to ICS
- ICS to PC
- Child ICS to parent ICS
- Parent ICS to child ICS
- ICS to ITS
- IP Gateway or IP PBX to ITS/IVR (This is part of the Interwise software)

Configuration

Each stream (voice, video, data) can be configured to have Diffserv marking enabled and to have a specific DSCP code in the range of 0-63.

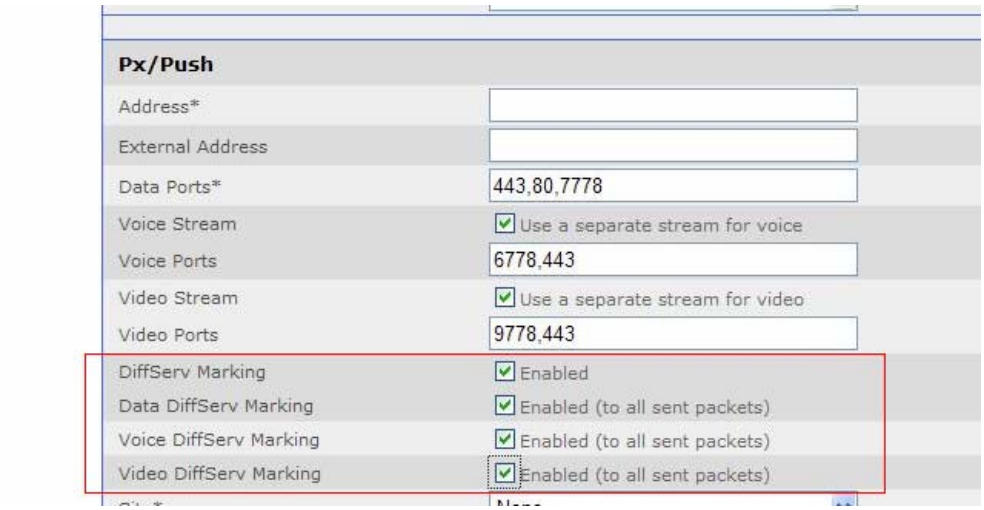
These settings apply to all sent packets of that specific component (PC, Server).

- **Server** (ICS, ITS) - Diffserv parameter will be defined in the ISM for each server.
- **Participant** - Diffserv parameters will be defined locally on the PC machine through a group policy editor.

ISM configuration for the ICS:

Diffserv marking configuration is applied at the ICS level and affects the marking on outgoing packets that this ICS will be sending to parent servers, child servers or Participants.

The following image shows how to define Diffserv marking in the server details in the ISM:



1. We require the ability to turn Diffserv marking on/off (by default the Diffserv marking is off).
2. Diffserv marking is defined per stream type (relating to the above requirement for stream configuration) and may have different values for every stream type.
3. DSCP values can be set in the range between 0-63.

The server retrieve from the ISM if Diffserv is enabled for a stream and then it uses a relevant DSCP code as defined in the Windows registry for the specific service type.

The default DSCP values are:

i. Data stream = 26 (Assured forwarding)
ii. Voice stream = 46 (Expedited forwarding)
iii. Video stream = 34 (Assured forwarding)

These default values can be manually changed by the administrator on the ICS machine.

Participant application configuration:

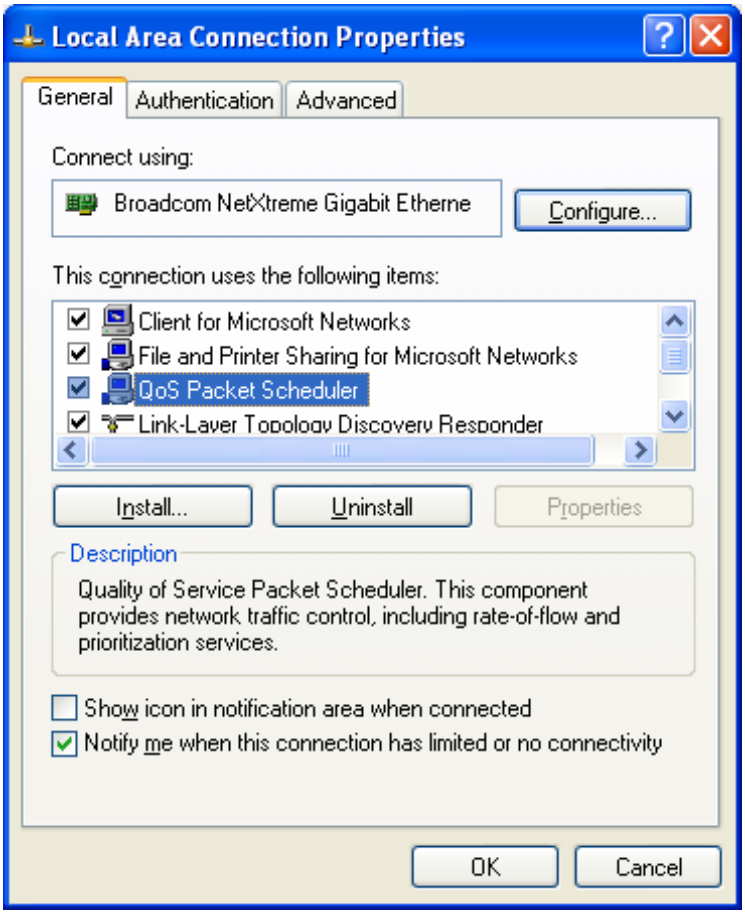
The Participant application needs to be able to apply Diffserv marking on sent packets and to use a specific DSCP value for each stream.

Whether the Diffserv marking will be on or off, should be defined in the Participant .ini file.

The DSCP values that will be associated with each stream are taken from the local OS according to the service type the Participant application specified when creating a stream (under Windows OS).

A 'QoS packet scheduler' must be installed on each of the end user machine and should be turned on.

This service comes with Windows XP OS and above.


Local Area Connection Properties
General
Authentication
Advanced
Connect using:
Broadcom NetXtreme Gigabit Etherne
Configure...
This connection uses the following items:
Client for Microsoft Networks
File and Printer Sharing for Microsoft Networks
QoS Packet Scheduler
Link-Layer Topology Discovery Responder
Install...
Uninstall
Properties
Description
Quality of Service Packet Scheduler. This component provides network traffic control, including rate-of-flow and prioritization services.
Show icon in notification area when connected
Notify me when this connection has limited or no connectivity
OK
Cancel