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# Multimedia Over FDDI

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# Multimedia Over FDDI

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## Abstract

*This paper focuses on issues in multimedia networking over FDDI and simulation results demonstrating the feasibility of multimedia over FDDI. Previous work in this field has concentrated on the asynchronous channel in FDDI. Other examinations of the FDDI synchronous channel have applied bursty traffic to the synchronous channel. Multimedia traffic is typically stream oriented. Our simulations were focused on simulating typical configurations with a range of applications and traffic models. The advantages of FDDI as an integrated voice/video/ data network are demonstrated.*

## 1: Introduction

The paper is organized as follows.

A brief discussion on the basics of FDDI and multimedia is followed by an overview of the various multimedia applications and issues in networking of multimedia applications such as voice and video. The importance of parameters such as network delay, intra-packet jitter, and bandwidth is highlighted.

The simulation models are discussed. The methodology involved testing the effect of variations in TTRT, network size, packet size, and offered load on the multimedia streams of voice and video, using the IRI Planyst<sup>+</sup> tool.

Finally, the simulation results for running multimedia over the FDDI synchronous channel are discussed. A comparison of multimedia over FDDI synchronous and asynchronous channels is made demonstrating the ability of the synchronous channel to provide a near-constant latency and bandwidth under various loads and configurations.

### 1.1: FDDI Basics

FDDI is a 100 Mbps high speed local area network standard developed under the auspices of American National Standards Institute (ANSI) X3T9.5 committee. Unlike other LANs whose origins were proprietary products, FDDI was developed by a group of people whose interest was to create a reliable fault-tolerant, high-speed network connecting numerous stations over greater distances than existing standards. The ANSI X3T9.5 committee thus developed a specification for a network based on a dual counter-rotating fiber optic ring using timed-token protocol, which is capable of transmitting data at 100 Mbps in each ring and which can extend to 500

stations over a total fiber length of 200 km with full system performance.

The FDDI MAC supports multiple classes of service: Synchronous, Asynchronous and Restricted Asynchronous. The asynchronous class is used for normal, bursty data traffic. For more constant, isochronous traffic, the synchronous channel can be used and it provides guaranteed bandwidth to each station on each token rotation. The restricted asynchronous service provides a mechanism for extended and protected dialog between a limited number of stations. This feature is rarely used.

There have been several publications on FDDI [10], [11], [12], [13], [14], [15], [16] which explain the protocol in further detail.

## 2: Multimedia

Multimedia is a term used to denote a set of applications, products, and technologies [1]. We define it as the use of multiple means to communicate information via a computer. Whereas computers today are mainly used for textual data information presentation and storage, multimedia uses the computer for text, natural and animated images, rendered graphics, auditory stimuli and realistic sound. There are two key components to multimedia: multiple output media and interaction.

Television as we know it today exemplifies one of the components of multiple output media; it delivers 'realistic' pictures, animation and sound. However, the content of what is viewed, and the way it is presented is largely beyond the control of the user. The user interaction is limited to turning on or off, switching channels, volume control and some image control.

Personal computers demonstrate the other key concept in multimedia: interaction. Unfortunately, most current personal computers exploit only a limited number of media such as text, graphics and a few sound tones. It is only now that the personal computers' ability to present clearer, sharper images is being exploited by multimedia applications.

### 2.1: Multimedia applications

The concept of multimedia is only as interesting as the applications it can support. It is the scope of multimedia applications which is drawing such a tremendous response from the information technology industries.

The various multimedia applications are shown in table 1 below.

**Table 1: Multimedia applications**

| APPLICATION                     | MEDIA<br>A=Audio, V=Video,<br>T=Text, I=Images,<br>G=Graphics | DESCRIPTION                                                                                                                                                                                               |
|---------------------------------|---------------------------------------------------------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Education/ Training             | A, V, T, I                                                    | A live or taped video instructional session locally at a station or across a network. This involves transfer of audiovisual and textual instructions.                                                     |
| Interactive Education/ Training | A, V, T, I, G                                                 | A live or taped video instructional session locally at a station or across a network. This involves transfer of audiovisual and textual instructions. The interaction may be via voice, text or graphics. |
| Information kiosks              | A, V, T, G                                                    | This is dispensing of information such as legal, tourist, consumer catalogs, dictionaries etc... These kiosks typically have minimal interaction with the user; mainly instructions and commands.         |
| Banking                         | T, I                                                          | Number transfers along with documents such as checks, money orders, internal papers etc...                                                                                                                |
| Medical Info. systems           | V, I, T, A                                                    | Stored or real-time capture of X-Rays, CAT scans, reports etc... with ability to do history, compare and micro and macro searches.                                                                        |
| Library                         | I, T, V                                                       | Search and retrieval of texts, audio and video cassettes.                                                                                                                                                 |
| Real Estate                     | A, V, I, T                                                    | Ability to view a neighborhood, block, street, house and interior of home remotely.                                                                                                                       |
| Electronic Mail                 | T, A, I                                                       | Ability to leave audiovisual messages along with textual messages.                                                                                                                                        |
| Home Video Distribution         | V, A                                                          | Cable company maintains library of movies which can be selected and played-back at viewer's convenience.                                                                                                  |
| Travel Agents                   | A, V, I, T                                                    | Similar in concept to real estate with ability to book travel, lodging and boarding in one call.                                                                                                          |
| CAD/CAM                         | T, G, A                                                       | Standard engineering CAD / CAM with the ability to voice-annotate and have on-line graphics help and notes as pop-up boxes.                                                                               |
| Electronic Collaboration        | A, V, T                                                       | Video conferences, concurrent CAD/CAM etc...                                                                                                                                                              |
| Advertising                     | V, A, I, T                                                    | Ability to reconstruct, edit and create images annotated with audio and text. The images may be sequenced into a video clip.                                                                              |
| Weather                         | I, T                                                          | Ability to parse satellite images of the atmosphere into weather reports.                                                                                                                                 |

There are several issues with multimedia, most notably *compression*, *synchronization* and *network transparency*.

Multimedia applications typically consume a lot of disk space for storage and current storage devices such as Hard Disks, floppy disks, CD-ROMs are not capable of storing massive amounts. Moreover, these storage devices have a very slow transfer rate (read, write). Motion video requiring 75 Mbps transfer rate cannot be sustained using a CD-ROM without some compression techniques.

Synchronization of multimedia objects is an important issue [8]. An object is a unit of data which may be a pixel, encoded audio, the multimedia document itself, etc... Synchronization can be at different granularity's. An example of synchronization is voice and video streams; the voice accompanying a video clip needs to be

synchronized to the picture. This issue is currently being addressed in the multimedia community.

Most multimedia applications will need network support [9]. Videoconferencing [7], distributed training, etc... require the use of a network. Insofar as the network is concerned, the only difference between multimedia and other applications is the integration of voice, video and data on the same network. Imaging can be modeled as bursty data and hence does not require any additional service of the network.

Almost all data transmission is asynchronous in nature. It is unpredictable and of varying duration. Most local area networks are optimized for high throughput, bursty data transmissions over a shared medium with little or no latency constraints. If the network is lightly loaded, and a station applies a large load to the network, it will be able to transmit the load with minimal delays. However, if the

network is heavily loaded, a station applying a large load will encounter significant delays before transmission. Thus, although Ethernet under light load offers excellent average latencies, at high load it offers little or no bound on the network access time.

Public networks or the telecom networks, on the other hand, are typically optimized for circuit-switching applications such as voice which requires low bandwidth and low latency. These networks typically cannot provide low access delay to bursty data.

In order to evaluate the feasibility of using existing LANs in multimedia applications, we examine the issues in sound and video transmissions such as bandwidth requirements, latency, jitter and maximum number of sessions. The characteristics of sound and video and the requirements of multimedia on networks are examined. Finally, the feasibility of multimedia over FDDI is explored.

### 3: Definitions

*Available bandwidth* is the bandwidth which is actually available for valid transmissions. Available bandwidth can also be measured in terms of network efficiency. Thus, in FDDI efficiency  $h = (T - D) / T$ ; where  $T$  = target token rotation time in ms and  $D$  = ring latency in ms. If  $D=0.1$  ms and  $T = 10$  ms, then  $h = 99\%$  and available bandwidth is 99 Mbps whereas total bandwidth is 100 Mbps.

*Latency* is the average end to end message delay which includes time for A/D conversion (if any), sample and encode, packetization, queuing delay, transmission delay, propagation delay, receive delay, decode, and presentation.

*Jitter* is the maximum instantaneous variation in object presentation time. If the object is a packet, then the maximum inter-packet arrival time variation is defined as packet jitter.

*Session* is defined as an interactive communications dialogue between two or more users. Thus a telephone conversation between two people is a session which consumes a portion of the available bandwidth.

### 4: Characteristics of Sound

We classified sound as human speech and music. Human speech or voice is typically in the 0-4 khz spectrum. The bandwidth of music discernible by the human ear is 20-25 khz (high fidelity systems have a bandwidth of 22 khz).

Conversational sound (speech) consists of talk-spurts followed by silence periods [3], [5], [6]. The ratio is typically 35:65 respectively, with only one person speaking at a time.

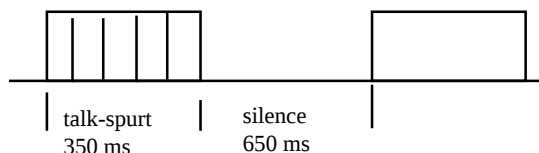


Figure 1: Speech pattern

In digitized voice, the voice signal is sampled at the Nyquist rate (twice the signal rate) in order to recover the original signal correctly.

Therefore the voice sampling rate is  $2 \times 4 \text{ khz} = 8 \text{ khz}$  or one sample every 125 ms in telephony. For stereo sound, the audio frequencies extend up to 22 khz. Hence sampling frequencies of up to 44 khz are also used in digital stereo sound. Each sample is coded into a bit-stream and the number of code-bits varies depending upon the coding scheme used.

Table 2: Compressed digital audio streams

| TRANSMISSION RATES          | bits/sample |
|-----------------------------|-------------|
| 2400 bps compressed         | 1-2         |
| 64 kbps standard PCM        | 6-10        |
| 40-16 kbps ADPCM scheme[22] | 5-2         |
| 300-400 kbps stereo         | 8-10        |

Although telephone voice is not very bandwidth intensive, stereo sound requires up to 0.4 Mbps bandwidth. At lower bit rates, the audio-quality deteriorates. Determining the quality of digitized audio is a very subjective phenomenon and studies have indicated a wide range of acceptable quality. It also depends on the application. Spoken voice can be compressed significantly before it becomes incomprehensible. The study of stereo quality sound is even more subjective.

There are several coding and compression techniques available for voice [2], [18], [21], [22]. Adaptive Differential Pulse Code Modulation (ADPCM) and Digital Speech Interpolation (DSI) are two of the popular mechanisms for voice compression. DSI is well-suited for packetized voice transmission as it conserves bandwidth during silence intervals in a conversation.

#### 4.1: Issues in sound transmissions

In voice transmissions [2],[3],[4],[5],[19], bandwidth is not an issue in the LAN environment, although it may be in stereo sound. A critical issue in interactive speech or music (or voice mail) played back over the network is latency. Once a person starts speaking, or music starts to play, intermittent delays during the speech or music

session, are often noticeable, and sometimes annoying. Therefore, once the session starts, it is desirable to maintain a continuous stream of sound. In fact, as shown in the table below, several studies have indicated that maximum tolerable latencies for speech are of the order of 600 ms. Our experiences with satellite communications has demonstrated that even 250 ms (one way) delays are

annoying though coherence is not impaired. For music, the latencies may be more noticeable and hence the delays may be required to be even less in order to be imperceptible. Several studies have been conducted in order to determine effects of network delays on voice transmissions [3], [4], [5], [21], [31], [32].

**Table 3: Effects of latency on human ear perception**

| One-way delay | Effect of delay                                                                                          |
|---------------|----------------------------------------------------------------------------------------------------------|
| >600 ms       | Speech becomes incoherent and unintelligible.                                                            |
| 600 ms        | Speech is barely coherent.                                                                               |
| 250 ms        | Annoying. Conversation style has to be changed.                                                          |
| 100 ms        | Imperceptible if listener hears from network only and not off the air.                                   |
| 50 ms         | Imperceptible even if the listener in same room and can hear naturally off the air and from the network. |

Using interactive speech as a model, we decided that the maximum end-to-end tolerable latency was 100 ms. These latencies would be acceptable for a large spectrum of multimedia applications.

The only effect of token jitter is on the need to buffer. In a truly isochronous network (providing constant latency), there would be zero buffering requirement in the the maximum buffering required for a 64 kbps digital voice stream would be 120 bytes, which is very small.

network. In an asynchronous network, there is a need to buffer. So long as the buffering is not excessive, the jitter has minimal impact on system design. For example, if the maximum packet latency and hence the maximum jitter between packets were to be restricted to less than 15 ms (as we targeted for the simulation), then

## 5: Characteristics of video

Video is moving pictures. It is different from imaging and graphics mainly in the motion component. Video represents motion scenes as a rapid sequence of still

frames. An NTSC compatible video is 640 x 480 at 30 frames per second. A PAL compatible video is 768 x 516 at 25 frames per second. The smaller the window size (fewer number of lines scanned), the lower the bandwidth requirement.

**Table 4: Effect of frame rate on human eye perception**

| Frames per second (fps) | Effect on human eye                                                               |
|-------------------------|-----------------------------------------------------------------------------------|
| <10 fps                 | Eye cannot discern motion. Each frame appears disjointed                          |
| 12-15 fps               | Eye can discern motion but is jerky.                                              |
| 30 fps                  | Television quality. Cannot discern high motion component (blurred); e.g. baseball |
| 60-75 fps               | HDTV quality. Can discern motion in high-motion games; e.g. ice-hockey            |
| 90 fps                  | Limit of human eye perception                                                     |
| 1000 fps                | Scientific video quality; e.g. shuttle blast-off recording                        |

The video signal can consume tremendous bandwidth. An uncompressed digital NTSC video can consume anywhere from 90 Mbps to 200 Mbps depending on the encoding. This is enough to overrun any existing network capacity. Compressed video [[28], [29], [30], [32]] offers a significant reduction in the bandwidth needs while

maintaining similar picture quality. Video bandwidth can be reduced in several ways:

- Variable resolution
- Variable frame rate
- Reduced color fidelity
- Removing intra-frame and inter-frame redundancy

**Table 5: Video compression techniques, rates and bandwidth requirements**

| Compression technique | frame rate (fps) | bandwidth             |
|-----------------------|------------------|-----------------------|
| MPEG [30]             | 30               | 1.5 Mbps video stream |
| MPEG II               | 60-75            | 4-10 Mbps             |
| P x 64 [28]           | 6-15             | 64 kbps - 2 Mbps      |

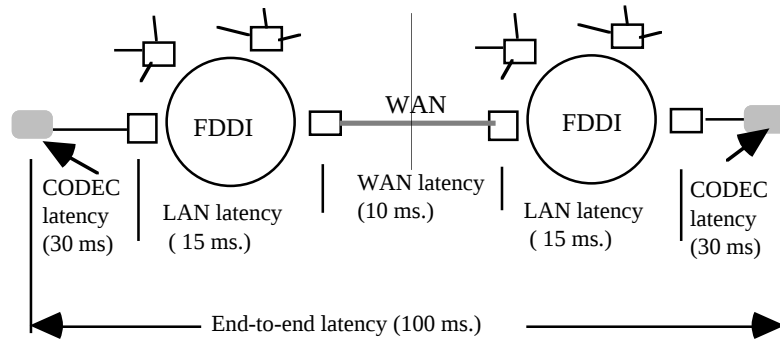
### 5.1: Issues in video transmissions

Digital video characteristics have not been studied as well as digital voice. It is difficult to characterize compressed video, video codecs and effect of network delay. The issue of compression algorithms and implementations is beyond the scope of this paper and the parameter of most interest to us is the network delay.

Using a well-known maxim that "*the human eye is more forgiving than the human ear*", we can apply the restriction that video must meet the same delay constraints

as those of voice. Depending on the video application, this delay may vary.

We selected 100 ms as an acceptable end-to-end latency [31], [32]. We assumed that codecs at each end will consume 30 ms in processing each frame and outputting data (at MPEG rates). This leaves an effective 40 ms latency for the network component of the latency. Of this, 10 ms is the latency across the WAN, and the rest is the LAN component of the delay. Assuming symmetrical LANs, this leaves 15 ms per LAN acceptable latency.



**Figure 2: Latency distribution across the network**

If the maximum packet latency and hence the maximum jitter between packets were to be restricted to less than 15 ms (as we targeted for the simulation), then the maximum buffering required for a 1.5 Mbps stream would be approximately 3 Kbytes, which is usual in an FDDI adapter.

## 6: Use of FDDI Synchronous class of service for video/voice applications

The purpose of our study was to examine the feasibility of multimedia applications over the FDDI synchronous channel when the FDDI is shared with normal, bursty data traffic. The synchronous channel offers a protected, low-latency bandwidth, which when unused is available to the normal asynchronous transmissions.

A portion of the FDDI bandwidth is allocated to synchronous services, either at startup or later by a bandwidth allocator. Up to 100% of the network bandwidth can be allocated to the synchronous service. In other words, it is possible to implement a synchronous only network. This allocation can be fixed, dynamically allocated at session initiation, or on any granularity preferred by the network administrator. A standard allocator such as CCITT Q.931, may be used to perform call-setup, tear-down and bandwidth allocation and monitoring. SMT defines a protocol and several MIB attributes which can be used to monitor and control the bandwidth allocation [27].

Each multimedia station is allocated a portion of the synchronous bandwidth. In order to allocate the bandwidth, each station needs to characterize the application requirements in terms of overhead and payload, where overhead includes token capture, framing and higher layer protocol headers, and payload is the actual synchronous data (e.g. voice, video). This should be calculated in units of bytes per 125 microseconds. An application of 1.5 Mbps would require  $1.5 \times 10^6 \times 125 \times 10^{-6} / 8 = 23.4375$  rounded up to 24 units of bandwidth. A similar calculation should be done for the overhead. The total bandwidth available is of the order of  $100 \times 10^6 \times 125 \times 10^{-6} / 8 = 1562$  units. Following the bandwidth allocation, it is necessary to select the packet sizes for the negotiated TTRT. For example, with a TTRT of 8 ms, the packet size for the above synchronous traffic (1.5 Mbps stream) can be calculated as the number of bytes that the stream will produce in 8 ms. This is 1500 bytes. Hence, with an 8 ms TTRT, and a 1.5 Mbps video stream, 1500 byte packets will be transmitted per token rotation.

The above method of allocating bandwidth was selected so that an application would not have to change bandwidth allocation every time that the TTRT value changed. Only the packet size would change while maintaining a constant overhead. There are other mechanisms for allocating bandwidth which may be simpler and more suitable for different network configurations.

For the purposes of the simulation we allocated synchronous bandwidth based on the TTRT value. Thus an 8 ms TTRT with a 1.5 Mbps application would require 1500 byte transmission time or  $1500 \times 80 \text{ ns} = 0.12 \text{ ms}$  per TTRT.

## 7: Network operation

An FDDI network can be operated in three ways:

- 1) asynchronous only;
- 2) mixed asynchronous and synchronous;
- 3) synchronous only.

We decided to test the network in all modes of operation with a special emphasis on the mixed asynchronous and synchronous mode. The offered network load was varied from 80% to about 150% of capacity. The traffic was a mixture of various applications such as voice, video, imaging, file servers, and interactive data.

We wanted to test the network not for its maximum configurations but for its typical configurations. After conducting a survey of the existing implementations and practices, we concluded that a maximum *unsegmented* network was in the range of 40-60 nodes. An unsegmented network has nodes on the same physical cable-plant with no intervening bridges, routers or some such interworking units. Usually, networks do not exceed 50 nodes because of issues such as loading, administrative domains, and traffic isolation. We selected a network with nodes in the range of 48 to 55 as the representative maximum of the typical network.

### 7.1 Topology

The following topology was adopted as the model for the study. A LAN-WAN-LAN model was seen as appropriate for typical multimedia services. We assumed that the two LANs were symmetrical in behavior.

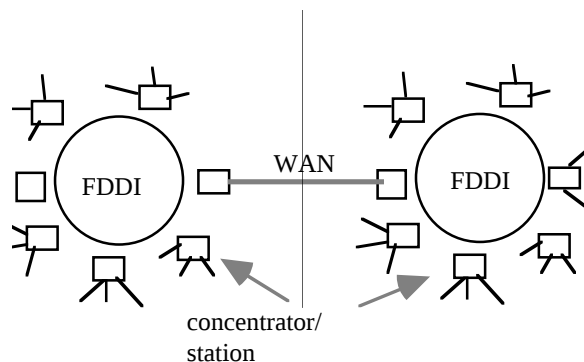


Figure 3: Topology for simulation

A single hierarchy of connections was selected because of the linearity of each LAN segment. Therefore, the delay characteristics for a single LAN could be linearly scaled to represent multiple-level LAN hierarchy. The WAN connection was assumed to offer a fixed latency path. The WAN could consist of ISDN, fractional T1, T1 or T3 lines depending on the bandwidth required.

### 7.2: Ring latency

A ring latency of 84 ms was used to test a small ring. This corresponds to roughly 7 kilometers of cable. To test the behavior of the network with larger ring latencies, a ring latency of 1 ms was selected. This corresponds to roughly 190 kilometers of cable.

### 7.3: Target Token Rotation Time

We decided to operate with three values of TTRT- 8 ms, 16 ms, and 24 ms TTRT values less than 8 ms were not selected because network efficiency drops significantly [33]. TTRT values larger than 24 ms were not selected because the packet latencies would be unacceptable for the multimedia traffic. Additionally, for a synchronous only network, a TTRT of 26 ms was used.

**Table 6: Traffic distributions**

| TRAFFIC TYPE              | INTER-ARRIVAL TIME (in ms) | PACKET LENGTH (in bytes) <sup>1</sup> | Peak OFFERED LOAD (in Mbps) <sup>2</sup> | Avg. OFFERED LOAD (in Mbps) | BUFFER SIZE (in packets) <sup>3</sup> |
|---------------------------|----------------------------|---------------------------------------|------------------------------------------|-----------------------------|---------------------------------------|
| Imaging host              | 0.33                       | 4K + 256                              | 106.25                                   | 10                          | 1000                                  |
| Imaging workstation       | 3.6                        | 4500                                  | 10                                       | 2                           | 10                                    |
| file data                 | 37                         | 1500 + 256                            | 10                                       | 3                           | 50                                    |
| interactive terminal data | 40                         | 500                                   | -                                        | 0.1                         | 10                                    |
| voice <sup>4</sup>        | 16, 84.5                   | 64, 2028 + 20 + 256                   | 0.032, 0.218                             | 0.0112, 0.076               | 10, 10                                |
| video                     | 8,14,16,24                 | 1500, 2304, 3000, 4500                | 1.5, 1.316, 1.5, 1.5                     | 1.5, 1.316, 1.5, 1.5        | 10                                    |
| gateway <sup>5</sup>      | -                          | -                                     | 10                                       | 10                          | 50                                    |

Overall, we stressed the network with a variety of traffic models to ensure that the network is robust under extreme operating conditions.

#### 7.4.1: Video streams

Each video source represents a compressed video stream. An MPEG-type stream rate of 1.5 Mbps is used. This consists of a train of packets ranging from 4500 bytes to 1500 bytes. The overall video loading is characterized through a parameter which represents the number of simultaneously active video sources.

A prescribed fraction of the video stream is directed through a bridge or a gateway to another FDDI or WAN network.

The target video latency for a video packet across a single FDDI is set to be around 15 ms (99% of packets) and of the order of 10 ms in the average.

### 7.4: Traffic and service models

Of the 50 stations, 26 were set-up to be voice/video stations, 3 were low-rate imaging sources, 10 were interactive data terminals, and 10 were file-servers. Optionally, high burst-rate imaging sources (1 to 7 stations) were used in place of the 3 low-rate imaging sources. The traffic was thus split into voice/video, imaging and data. Imaging and data were further subdivided. There were two models for the imaging and two models for the data.

The following table shows the characteristics of the various traffic models that we selected.

#### 7.4.2: Voice streams

Associated with each video stream is a packet voice stream. This stream was characterized as interactive and stored (MPEG CD-ROM specifications).

The interactive voice is characterized as a sequence of 32 kbps voice packets, each 64 bytes in size. The voice source is modeled as a sequence of on-off periods, representing the talk-spurts and silence periods typical in a voice conversation. The ratio of talk to silence periods is 35: 65. The resulting offered load of the voice stream is 11.2 kbps.

The stored voice is characterized as a sequence of 2 Kbyte packets (2028 + 20 MPEG headers), with an inter arrival rate of 84.5 ms. These packets are typically interleaved with the video stream in a ratio of one voice every six video packets. On the FDDI, these packets are repacketized if necessary to smaller packet sizes.

<sup>1</sup> The length denotes the data + headers. The headers were deliberately chosen to be a large number.

<sup>2</sup> The arrival rate for some traffic had a distribution model rather than a constant rate, which led to peak offered loads and average offered loads.

<sup>3</sup> The buffer size corresponds to the buffering at the transmit and receive queues. If an incoming packet finds the buffer full, it is dropped. This corresponds to blocking.

<sup>4</sup> An interactive voice model (32 Kbps ADPCM), and a MPEG stored voice-stream model were selected for modelling.

<sup>5</sup> The gateway loading was approximately equal to 4 voice/video stations and 1.3 file servers.

The desired latency for voice packets across a single FDDI is also set to be approximately 15 ms (99% of packets) and of the order of 10 ms in the average.

#### **7.4.3: Data : Interactive**

This traffic consists of short packets (500 bytes) arriving at random with low response time requirements of 50 ms (99% of packets) and 25 ms in the average.

#### **7.4.4: Data -File transfer**

This is modeled as file data to/from Ethernet hosts with the FDDI being used as a backbone to Ethernet clients. The file length is assumed to be uniformly distributed between 1500 and 25000 bytes. Each file on average consists of 8 packets of 1500 bytes + 256 bytes headers, each arriving at 10 Mbps once the file transfer is initiated. The file inter-arrival time is exponential. The offered load is 3.6 Mbps.

#### **7.4.5: Imaging : low-end (workstation)**

This traffic source was used in some of the simulation runs.

This is modeled as images coming off Ethernet hosts into the FDDI host at 10 Mbps. The image size distribution is uniform over 1.25 - 5 Mbytes. This image stream is packetized into maximum length FDDI packets (4096 + 256 bytes). The image inter-arrival time is varied and a default value of 20% on-time and 80% off-time is assumed. This assumes that a host is busy with imaging only 20% of the time. Thus the peak offered load is 10 Mbps, but the average offered load is 2 Mbps.

The maximum acceptable delay in transmitting an image and receiving it at the receiver is 1 s (99% of packets) and 0.5 s average delay. A buffer size of 10 packets is used. Any over-flow leads to packet dropping.

#### **7.4.6: Imaging : High-end (host)**

This traffic source is used to simulate the impact of a very bursty load on the network. The image distribution is uniform over 1.25 to 5.625 Mbytes. A single image stream consists of regularly arriving maximum sized packets (4096 + 256 bytes). The average image inter arrival rate is 0.32768 ms. The peak offered load is 106.25 Mbps and the average offered load is 10 Mbps.

### **8: Results**

The following sections summarize the results of the simulations.

#### **8.1: Case 1- Asynchronous only network**

In an asynchronous only network, with no synchronous bandwidth allocated or used, the voice and video are treated as data. No separate queue is allocated on transmit or receive. In such a network also it is possible to maintain a bound on the delay suffered by the packets. The following observations refer to figures 4 to 19.

##### **8.1.1: Effect on 99% latencies**

Due to the unpredictable nature of the traffic (asynchronous and bursty), the delay cannot be tightly bounded. As can be seen from the figures 8 and 10, the 99% latencies suffered by video packets is as high as 48 ms when the network is not overloaded but running close to capacity (90% load). When the network is overloaded, the latencies can be as high as 252 ms.

In a more typical environment, where the traffic does not consist of such high burst sources (imaging at 100 Mbps), it is possible to obtain low latencies. We were able to verify this in our simulation (see figure 6) where in an asynchronous only network, with 86% network loading, and a TTRT of 24 ms voice and video packet latencies were restricted to under 15 ms.

##### **8.1.2: Effect of TTRT on latencies**

In an overloaded network the higher the TTRT value, lower the latencies. In the 8 to 24 ms range, it was observed that the 24 ms TTRT value consistently offered lower delays for all traffic types when the ring was large. (figure 8 and 10).

##### **8.1.3: Effect of ring size**

Increasing ring latencies had an adverse effect on the packet latencies. This was reflected in the increase in the mean and maximum latencies for voice and video (figure 6 and 10). This effect was less noticeable in the 99% latencies.

##### **8.1.4: Effect of buffer sizes**

Buffer sizes were allocated to the individual queues, asynchronous and synchronous at different stations. Therefore the asynchronous buffer size at the imaging station was varied from 10 to 1000 packets, at the file server stations it was 50 packets, and at the interactive terminals it was 10 packets. Every synchronous station had a 10 packet buffer.

These buffers never overflowed except in the overload scenario. Then too, the blocking or buffer overflow was occurring only at the imaging station. This result is intuitive because the imaging stations were offering instantaneous overload. This burst would fill up the buffers and since the network was not faster than the application, the buffers would not empty out fast enough. At heavy loads these buffers are unable to empty and hence lead to overflow. Low burst-rate applications such

as file service and interactive terminal traffic were not capable of exceeding the network capacity and hence the network could always clear the buffers faster than the application could fill them.

The percentage of traffic blocked due to overflow decreased slightly with increasing TTRT. This is again intuitive because the larger TTRT values allowed longer transmission times which in turn allowed the buffers to be drained more often.

#### **8.1.5: Effect on gateway**

For traffic from FDDI to other LANs or WANs, the latency was gated by the characteristics of the other LAN or WAN. So long as the outbound traffic was less than the capacity of the WAN or LAN, there was little or no queuing delay. The major component of the delay was then the transmission delay.

The effect of the changing of the various network parameters on the gateway (figures 4-18) was similar to that of other end-stations except that in the overload case the gateway suffered significant blocking. The gateway was the bottleneck for voice/video sessions spanning the LAN-WAN-LAN connection. The minimum latencies observed when the network loading was 90% and with at least one high burst-rate source on the network, was 24 ms at 8 ms TTRT.

When the high burst-rate source (imaging with peak offered load of 100 Mbps and average of 10 Mbps) was removed, the gateway provided acceptable performances with latencies less than 15 ms.

### **8.2: Case 2- Asynchronous plus synchronous network**

In this network the voice and video are transmitted over the synchronous channel. The synchronous bandwidth is allocated per station based on the application requirement. The voice and video packet sizes are varied according to the TTRT requirements. If 64 byte interactive voice packets are used, then the packet size is constant for the different TTRT. For video, the packet-size is 1500 bytes for 8 ms, 3000 bytes for 16 ms and 4500 bytes for 24 ms TTRT. The gateway is used for voice/video and file server data forwarding. The gateway is allocated synchronous bandwidth in proportion to the traffic leaving the LAN. If four video streams are leaving the LAN, then the gateway is allocated bandwidth equal to four video streams (e.g. at 0.75 ms per video stream the gateway is allocated 3 ms).

#### **8.2.1: Effect on 99% latencies**

We observe that the 99% latencies for voice/video streams is fairly constant and under all circumstances- 90% and 150% loading on a small ring, 90% and 150% loading on a large ring, and different TTRT - the 99%

latencies are within 24 ms. For 8 ms and 16 ms TTRT values, the latencies are always within 16 ms. Even under extreme stress, the synchronous channel offered a low-latency path for time-critical applications such as multimedia.

#### **8.2.2: Effect of TTRT on latencies**

Within the range of TTRT values yielding acceptable latencies, it was more difficult to isolate the better TTRT. We observed that while 8 ms TTRT yields excellent values for voice/video traffic, the asynchronous bursty traffic suffered lower latencies at the higher TTRT values. Considering all traffic streams, we observed that a 16 ms TTRT offered better all-round latencies in a mixed synchronous and asynchronous network (figure 5, 7, 9 and 11).

#### **8.2.3: Effect of ring sizes**

To observe the effect of ring sizes on the network performance, we simulated with a small ring size of 84 ms latency and a large ring size of 1000 us latency. The effect of increased ring sizes on voice / video (VV) is readily apparent on the mean delays. The mean latencies increased by as much as 40% whereas the 99% latencies increased by about 10-15% only. Since the important parameter for system design is the 99% latency rather than the mean latency, this implies that in the range of typical ring sizes (50 ms to 400 ms ), the 99% latencies are fairly constant.

#### **8.2.4: Effect of buffer sizes**

The effect of buffer sizes is slightly higher blocking than in the asynchronous only network. This is due to the synchronous traffic effectively blocking some portion of the bandwidth. If 20% of the bandwidth is allocated to the synchronous channel, then for the asynchronous applications FDDI appears to be a 80 Mbps asynchronous data pipe. This is an excellent result because it implies that if the asynchronous application capacity requirement is known, the rest of the FDDI bandwidth can be allocated to the synchronous channel and there will be little or no effect on the asynchronous applications.

#### **8.2.5: Effect on gateway**

For traffic from FDDI to other LANs or WANs, the latency was gated by the characteristics of the other LAN or WAN. The major component of the delay was then the transmission delay.

The effect of the changing of the various network parameters was similar to that of other end-stations (figures 5, 7, 9,11, 13, 15, 17). It was observed that for the incoming traffic from the gateway, the latencies were less than 16 ms for TTRT values of 8 and 16 ms, under overload. For 24 ms TTRT the delay was of the order of

23 ms under overload. In light to heavy loading conditions the latency was 8 ms.

We also observed that changing the number of outbound sessions had little or no impact on the latency if the appropriate synchronous bandwidth was allocated.

#### **8.2.6: Synchronous bandwidth allocation**

It was observed that incorrect bandwidth allocation could dramatically affect the performance. Allocating the appropriate bandwidth resulted in a guaranteed low-latency channel for the application.

#### **8.3: Synchronous only network**

We simulated a synchronous only network with 70 voice/video stations and a ring latency of 122 ms (figure 19). In this configuration, the only traffic on the network was voice/video. Each station was generating voice and video traffic at a combined rate of 1.5 Mbps. Each station was allocated 0.37 ms per token rotation leading to a TTRT of 26 ms. The total load was near the maximum sustainable by the network.

The results showed that the 99% latencies were less than 7 milliseconds. In fact, even though the TTRT was 26 ms (in order to accommodate 70 stations), the actual maximum delay suffered by any packet was of the order of 10 milliseconds.

It was also observed that there was a sharp drop in performance with a small increase in the number of stations. This is because the network is operating at maximum capacity and even a slight increase in load can cause the synchronous bandwidth to be over allocated. We concluded that it is possible to calculate the maximum number of synchronous stations that can be supported and provide acceptable latencies. For the given application (1.5 Mbps MPEG stream), it is between 55-65 stations, depending on the safety factor desired.

It is not advisable to apply bursty and asynchronous traffic to the synchronous channel as it can lead to extremely high delays [34], [35]. This is because a burst offers instantaneous load which can exceed the allocated bandwidth causing the queuing component of the delay to increase significantly.

### **9: Conclusion**

The FDDI asynchronous mode provides excellent multimedia services at normal loads. However, it has limitations under heavy and extremely bursty loads which may not be acceptable for voice/video services. The FDDI synchronous mode of transmission provides a near constant, low latency service under various loads and configurations. Our results demonstrate that a large number of simultaneous MPEG and Px64 sessions can be supported even when the network is in over-load. Audio,

video, imaging and data communications services can be integrated over FDDI without compromising any service requirement in all but the most extreme cases. These results demonstrate the feasibility of FDDI as an integrated network and are part of a continuing study on multimedia over FDDI.

### **10: Acknowledgments**

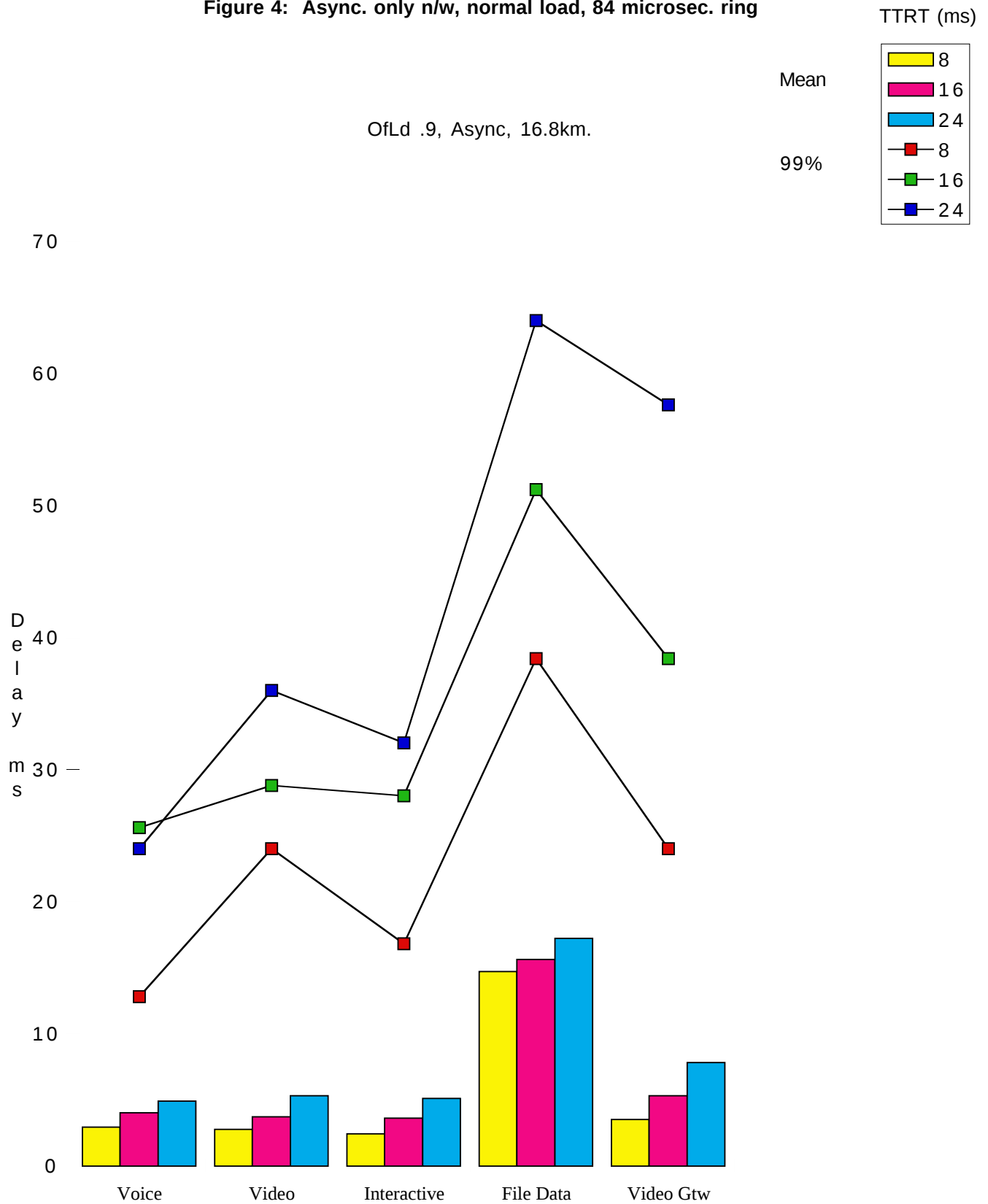
Our acknowledgments to Bob Grow (XDI), J. D. Russell (IBM), Bob O'Hara, Basil Alwan and Dave Roberts (Advanced Micro Devices) for their invaluable help and suggestions in the model development.

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Mean,99% Delay (20,21,22- Image WS Not Shown)

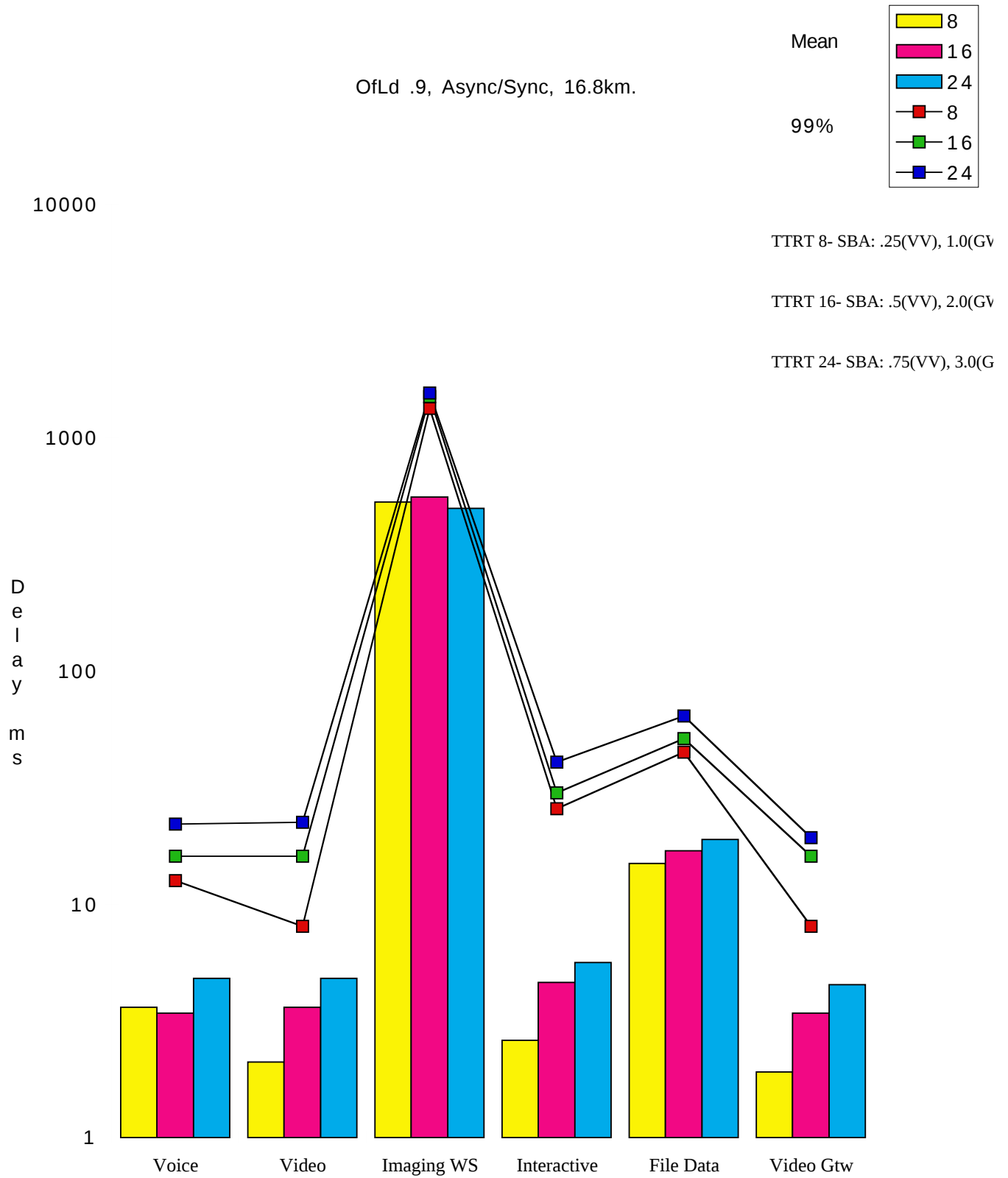
Figure 4: Async. only n/w, normal load, 84 microsec. ring



Mean,99% Delay (23,24,25)

Figure 5: Sync./Async. n/w, normal load, 84 microsec. ring with imaging ws

TTRT (ms)



Mean,99% Delay (20,21,22)

**Figure 6: Async. only n/w, normal load, 84 microsec. ring with Imaging WS**

TTRT (ms)

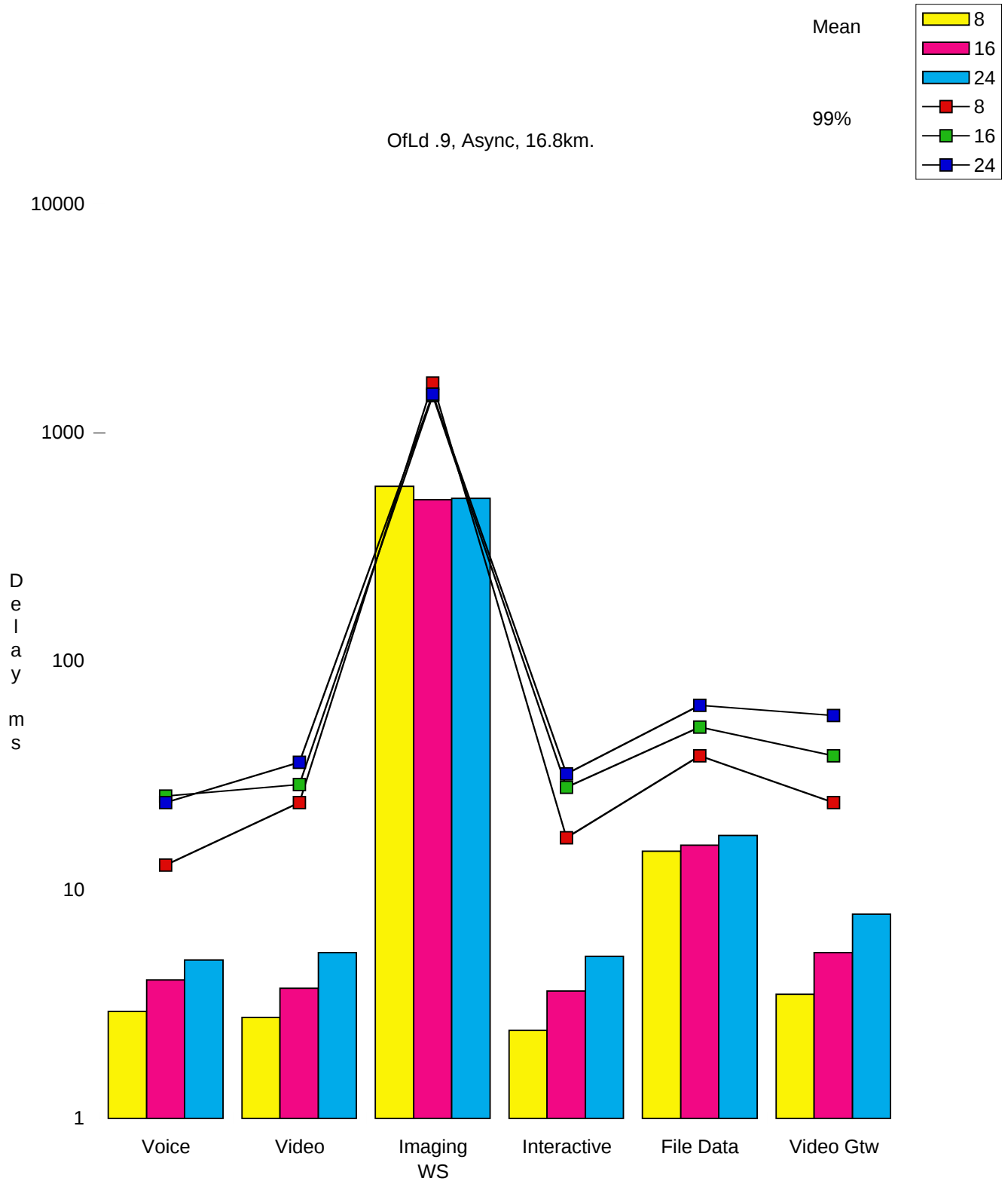


Figure 7: Sync./Async. n/w, normal load, 84 microsec. ring

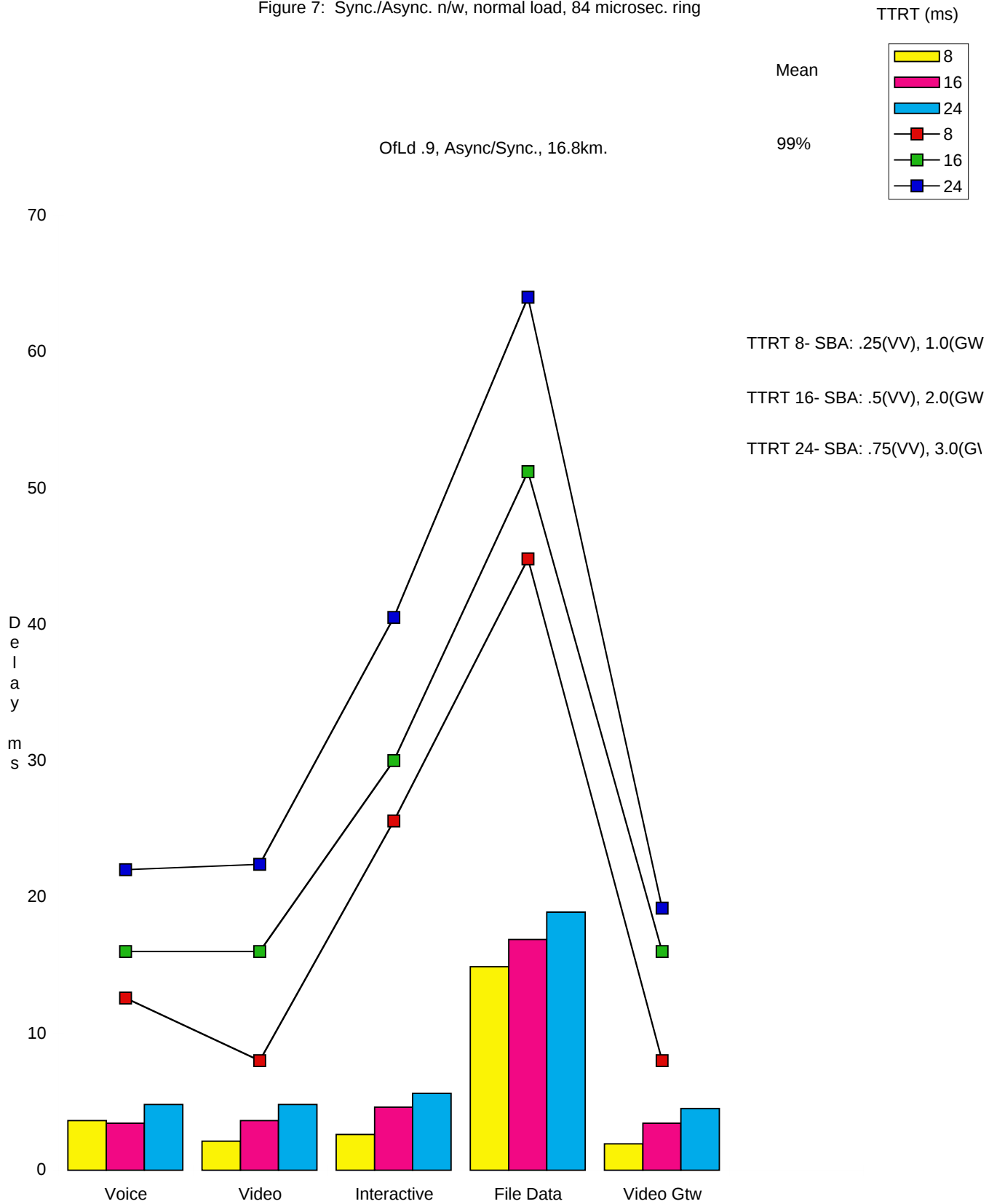
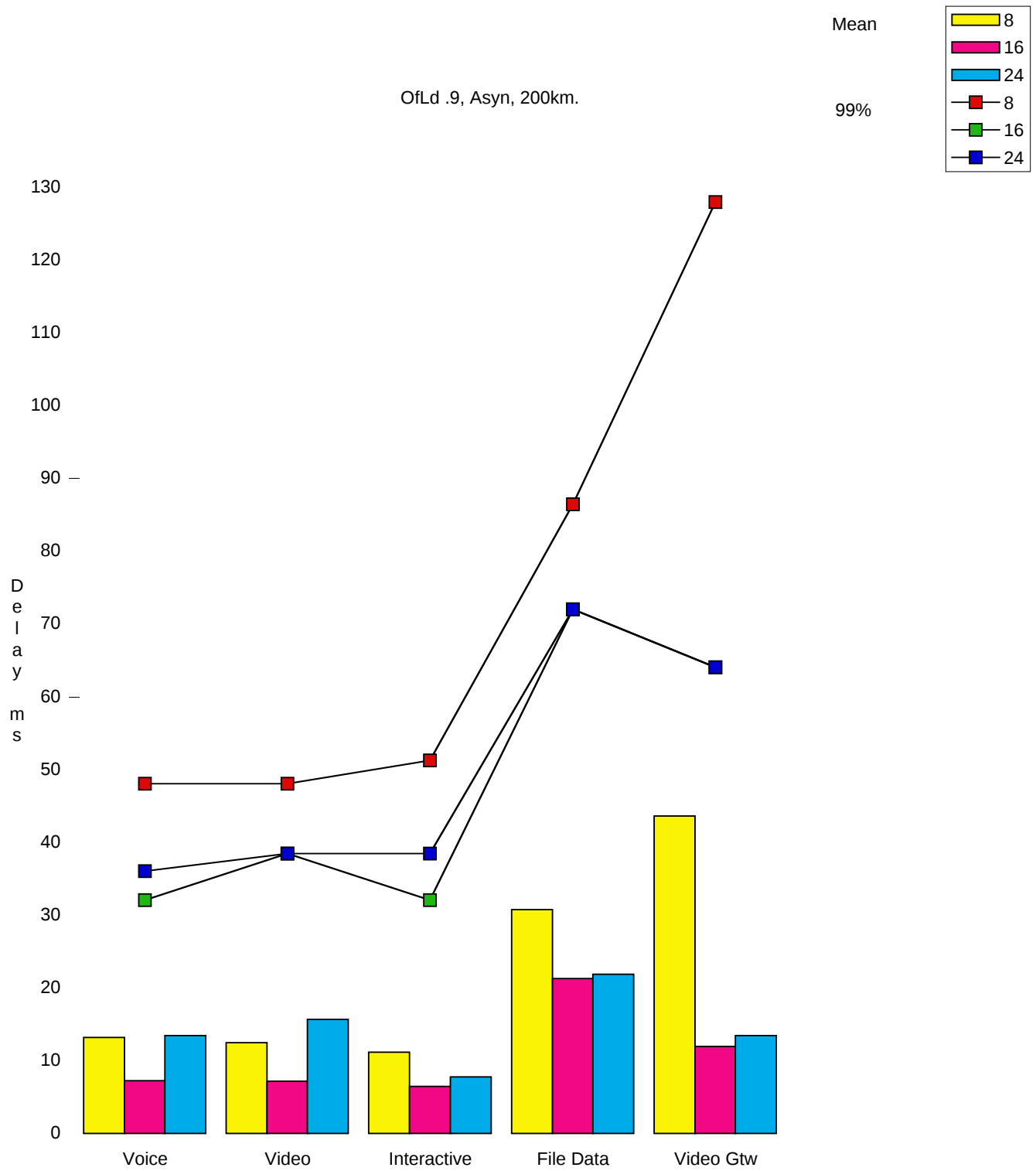


Figure 8: Async only n/w, normal load, 1000 microsec. ring

TTRT (ms)



Mean, 99% Delay (35,36,37- Image WS not shown)

Figure 9: Sync/Async n/w, normal load, 1000 microsec. ring

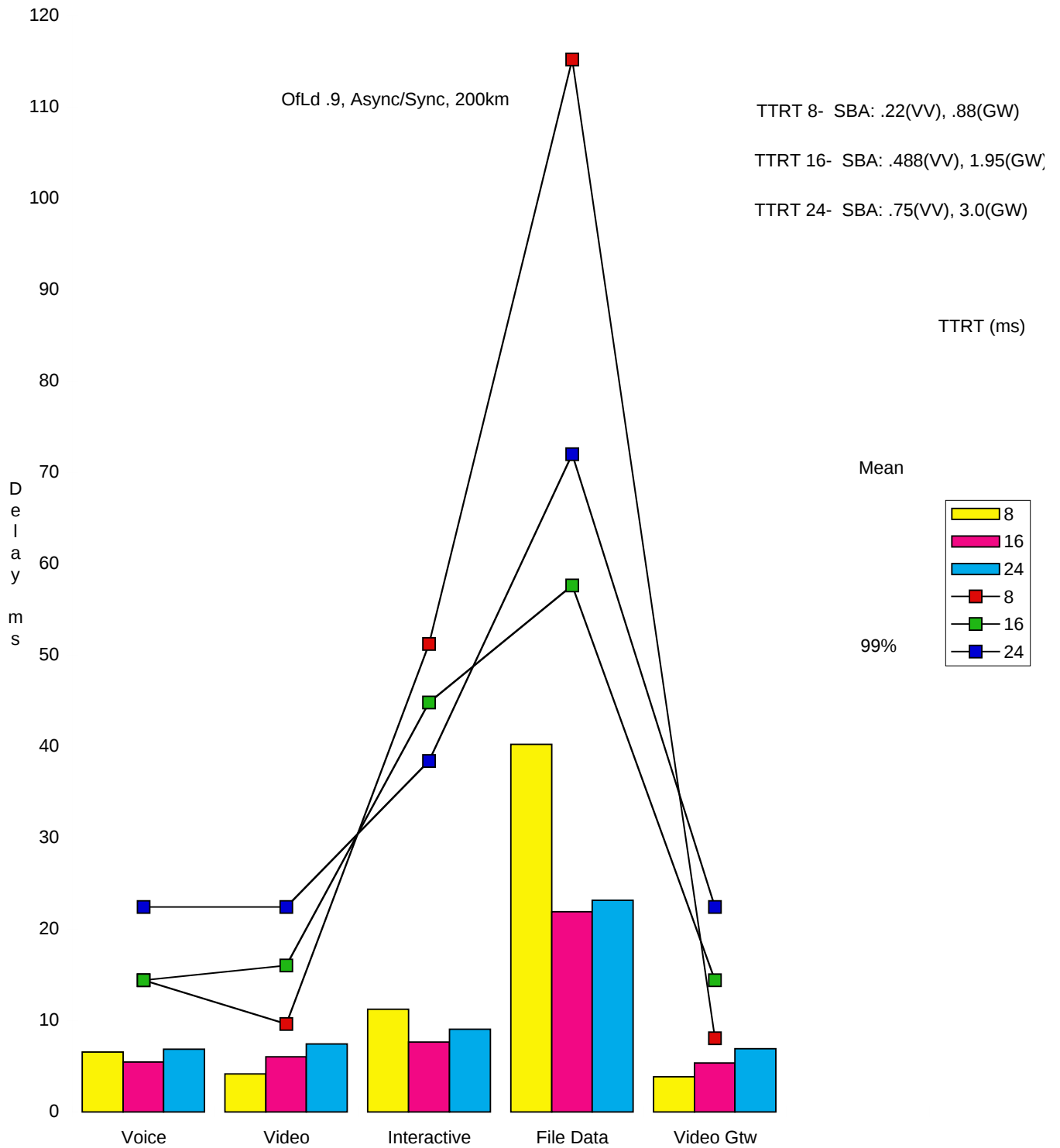


Figure 10: Async only n/w, normal load, 1000 microsec. ring with Imaging WS

TTRT (ms)

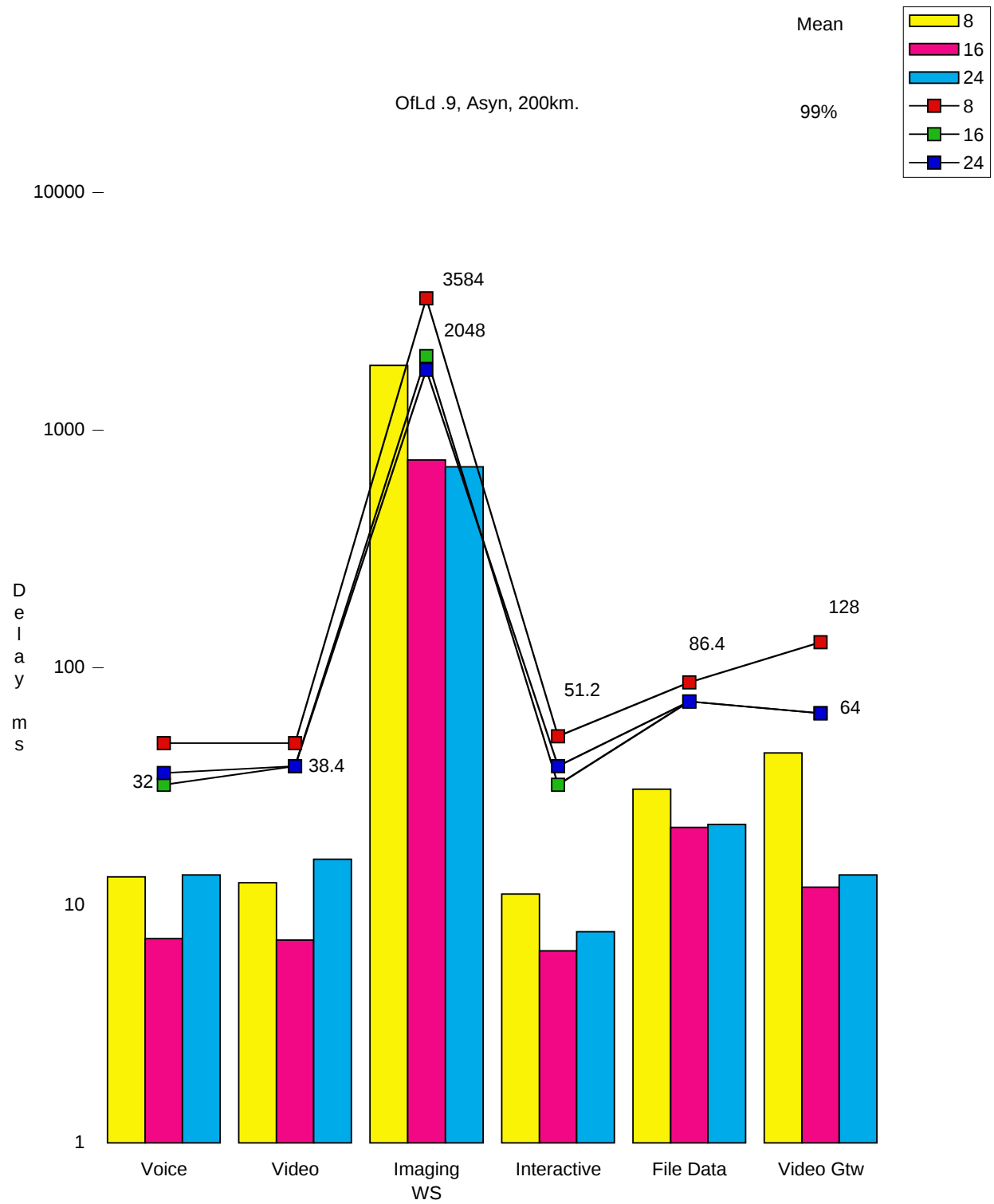


Figure 11: Sync + Async n/w, normal load, 1000 microsec. ring with Imaging WS

TTRT (ms)

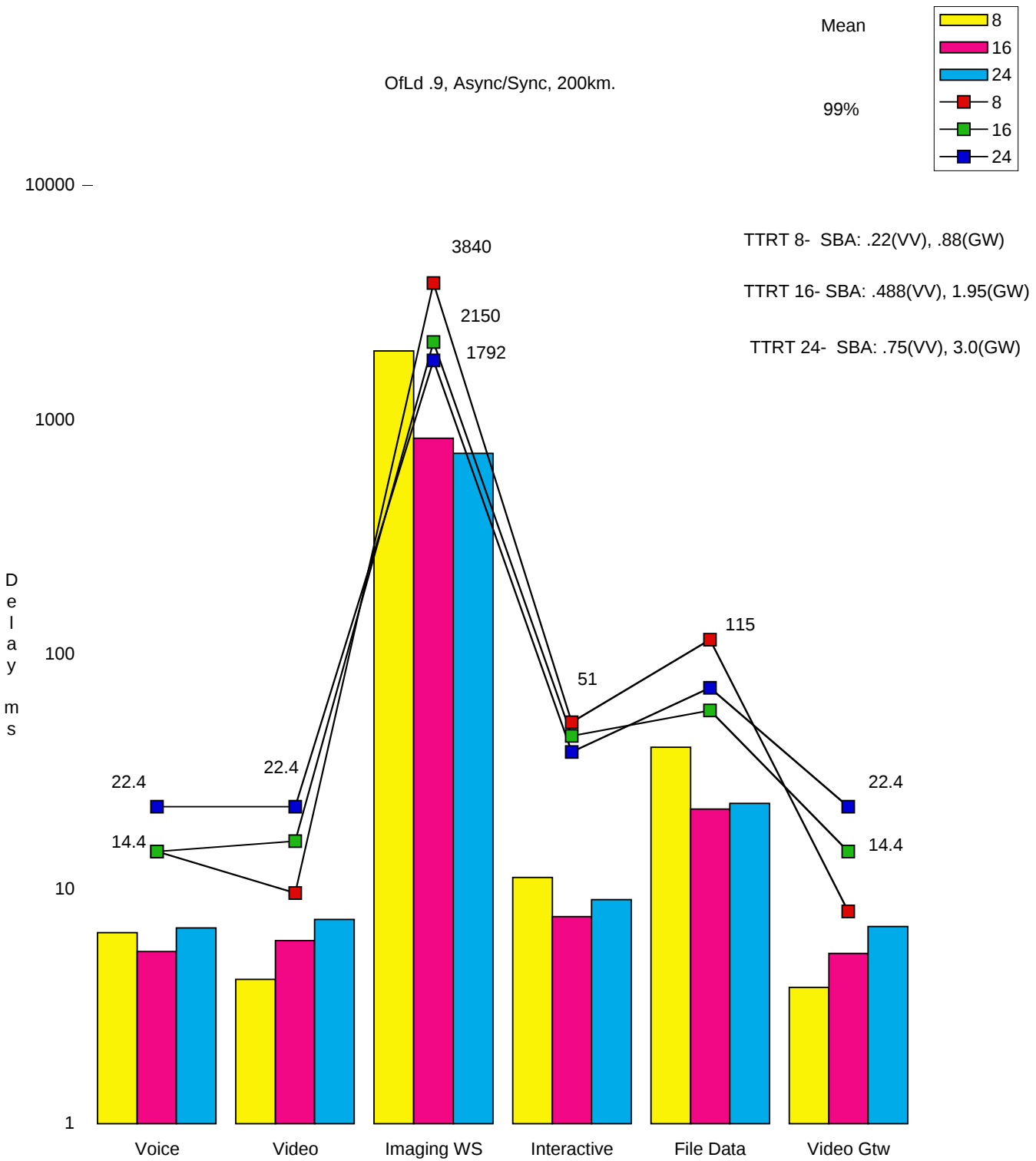


Figure 12: Async. only n/w, overload, 84 microsec. ring

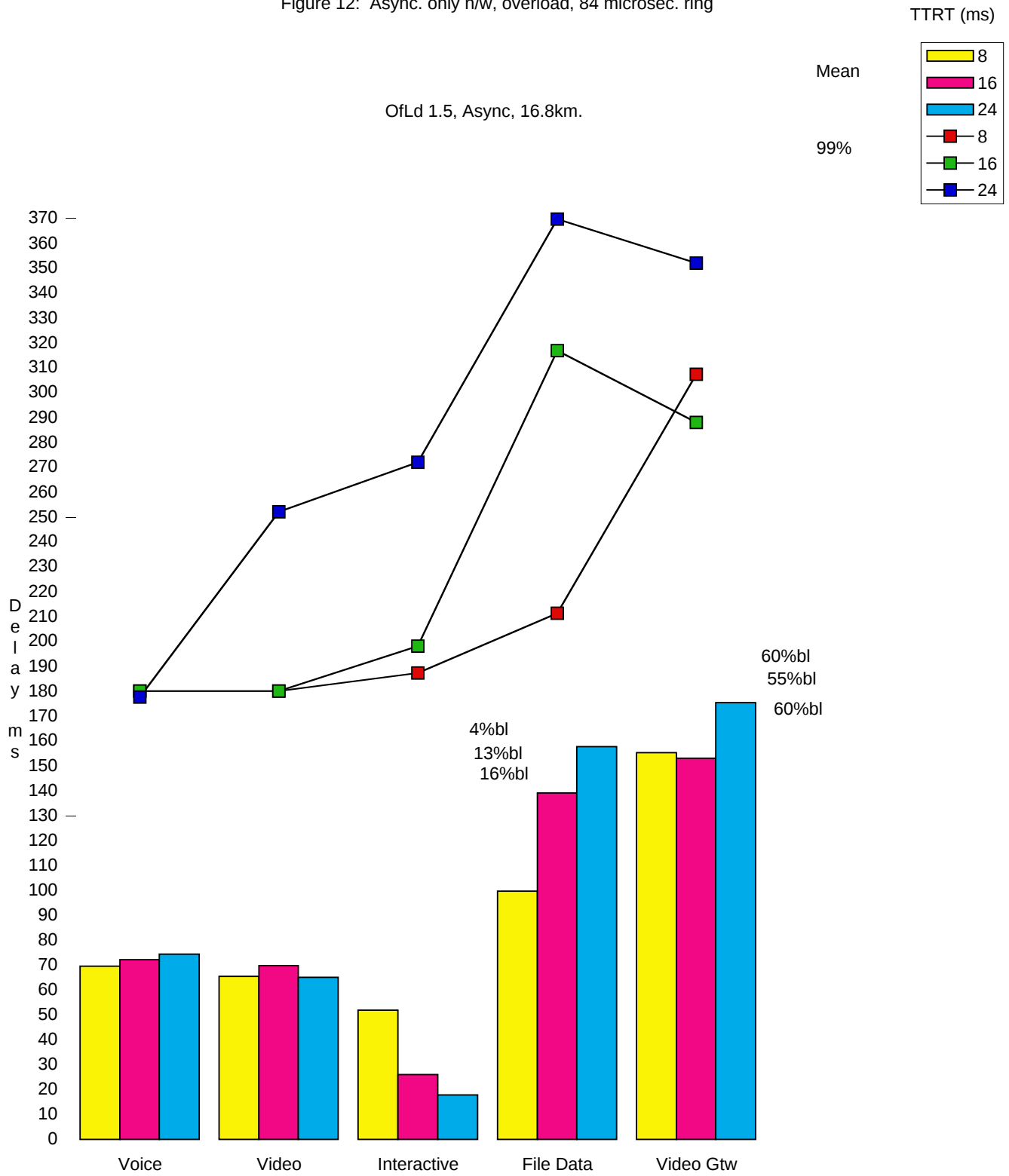


Figure 13: Sync + Async n/w, overload, 84 microsec. ring

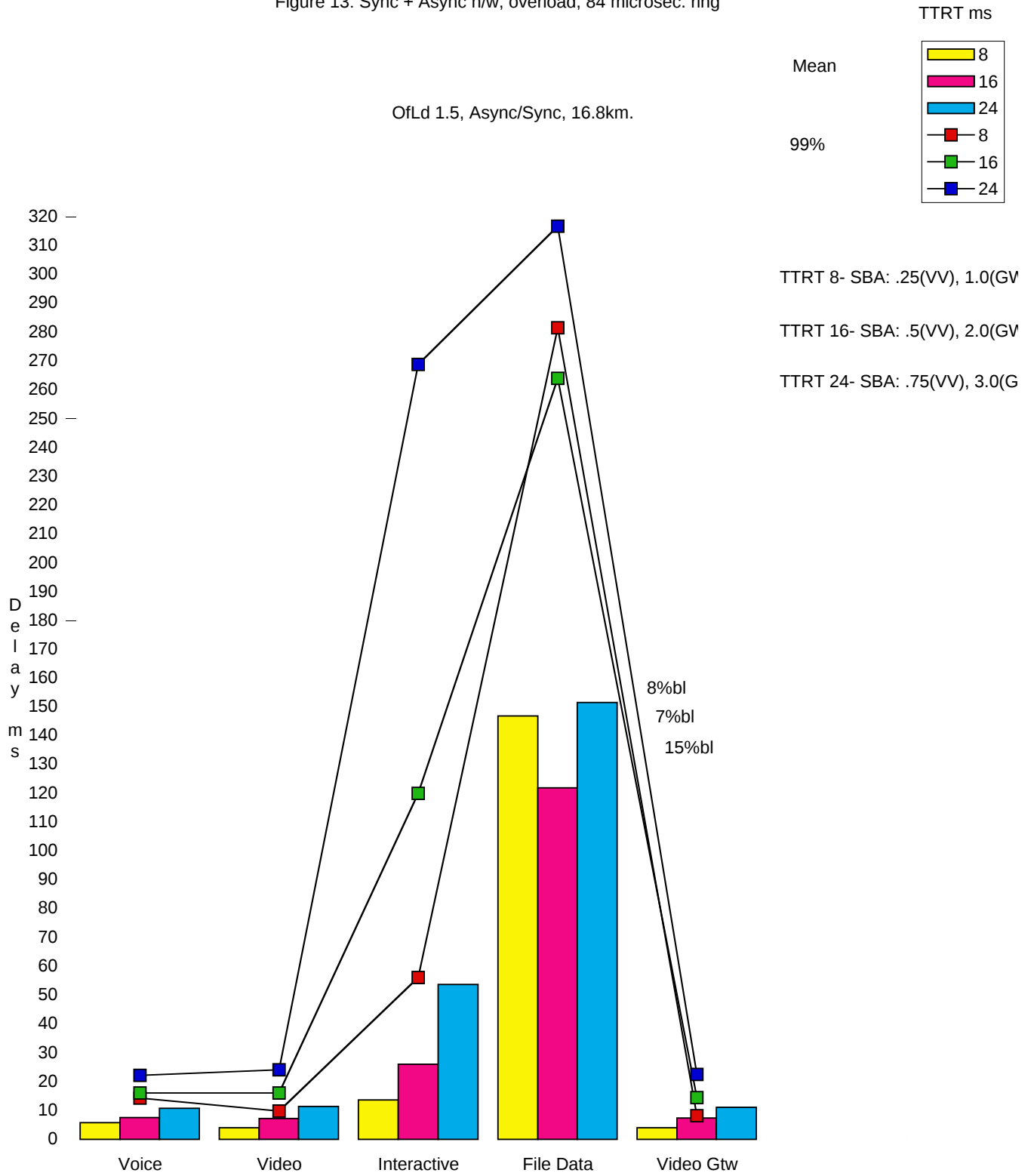


Figure 14: Async. only n/w, overload, 84 microsec. ring with Imaging WS

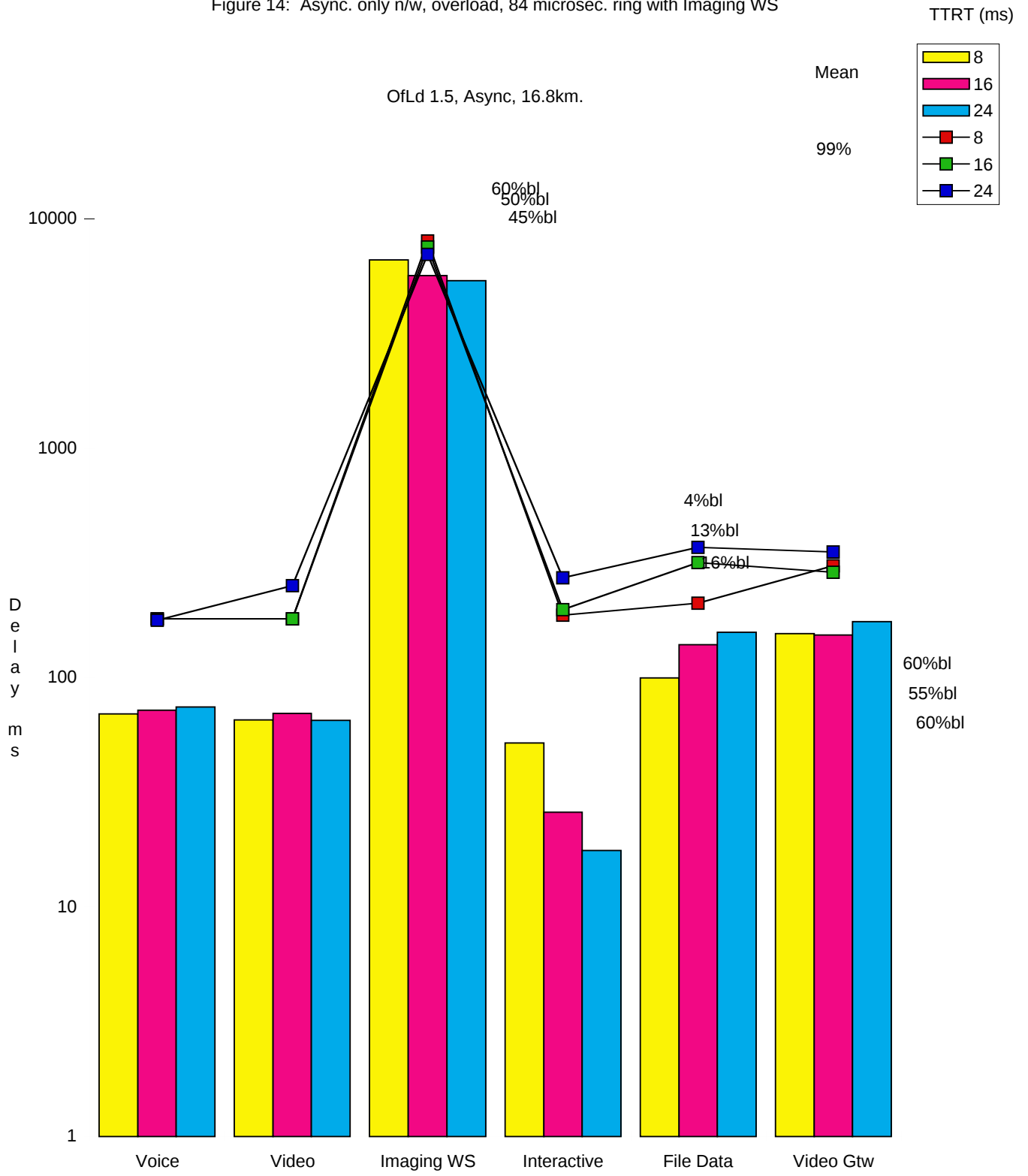


Figure 15: Sync. + Async n/w, overload, 84 microsec. ring

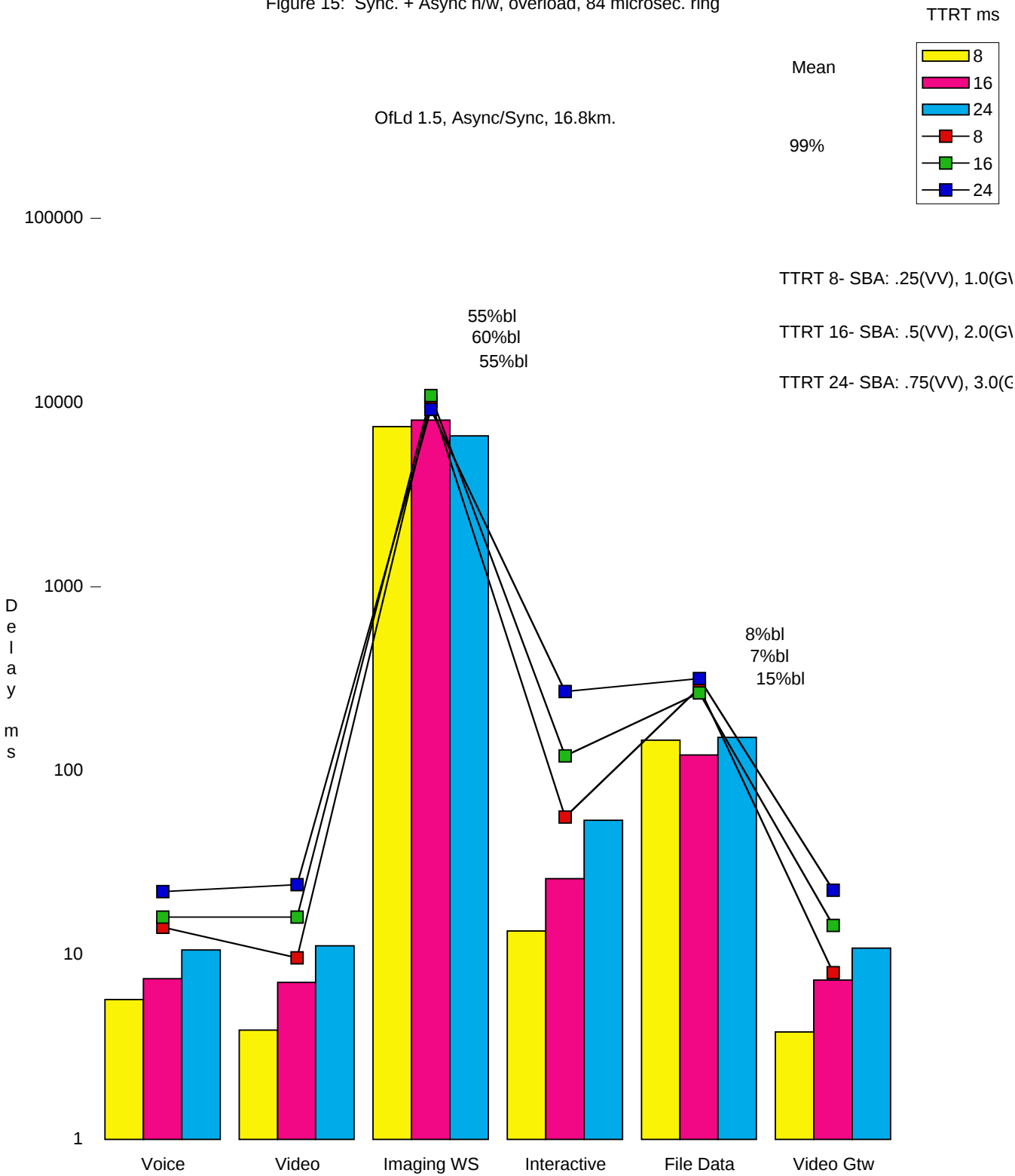


Figure 16: Async only n/w, overload, 1000 microsec. ring with Imaging WS

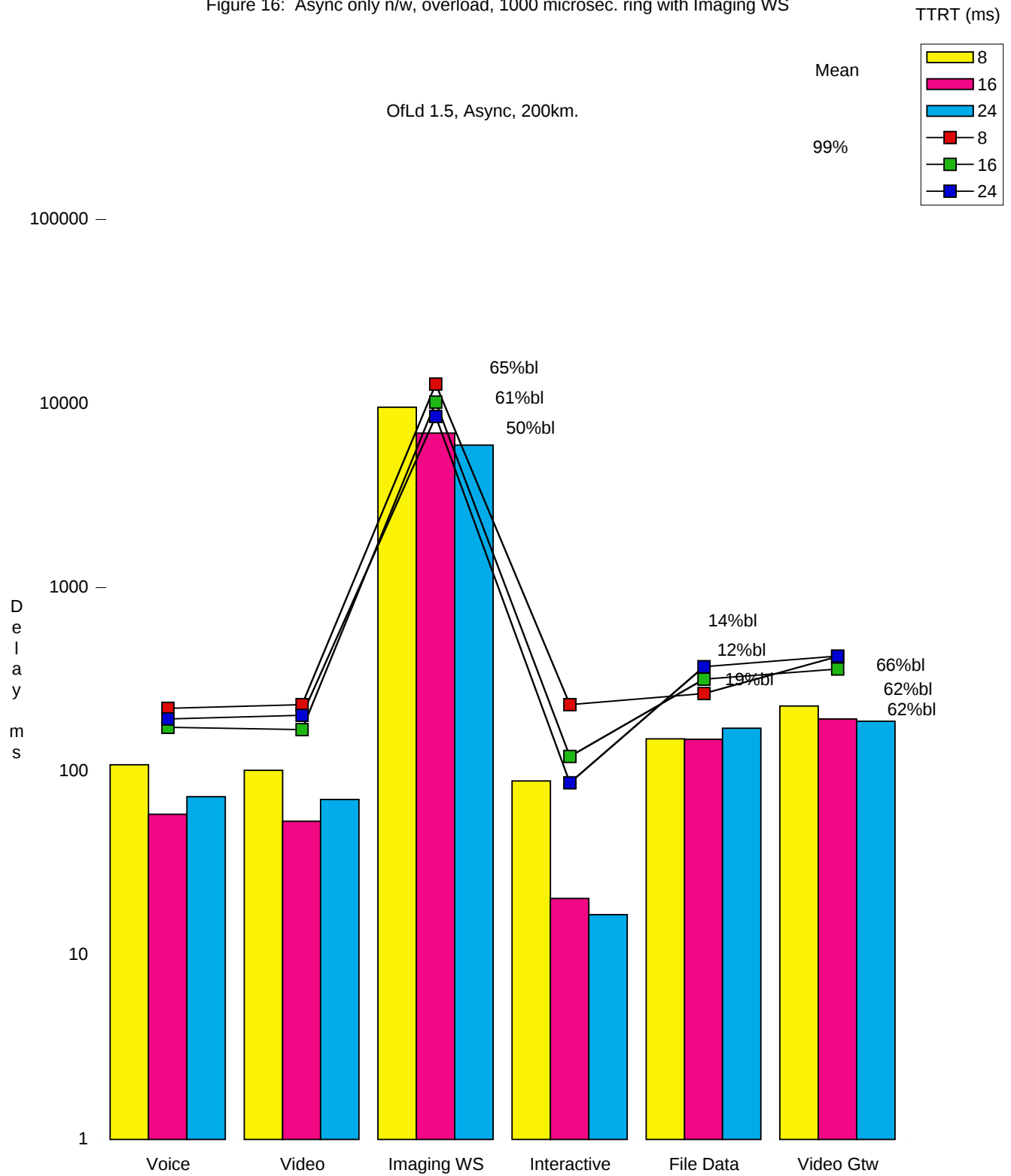


Figure 17: Sync + Async n/w, overload, 1000 microsec. ring with Imaging WS

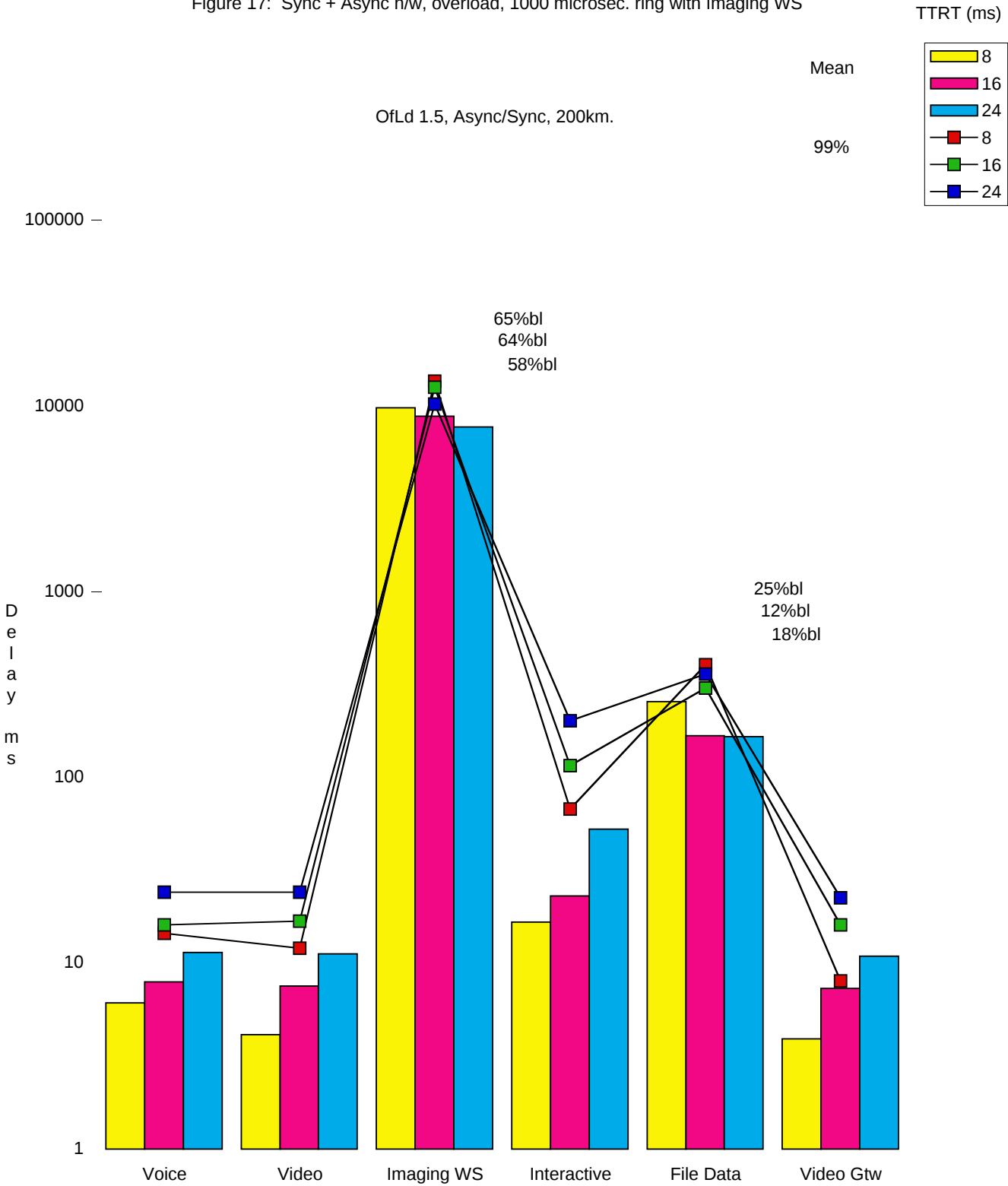


Figure 18: Async only n/w, normal load, 1000 microsec. ring

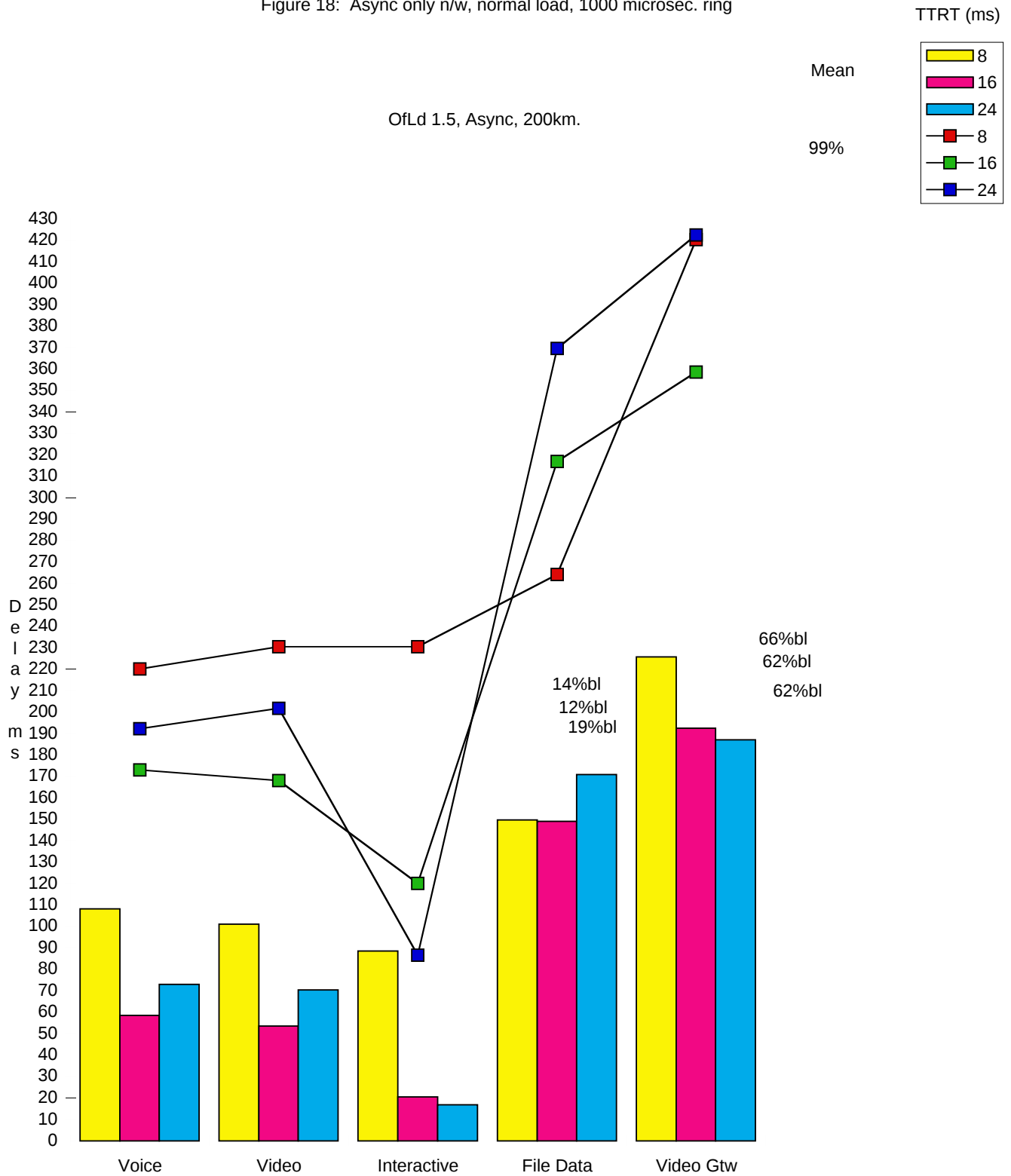


Figure 19: Sync only n/w, normal load, 122 microsec ring

