SOME ASPECTS OF IMPLEMENTING MULTIMEDIA SERVICES ON NETWORKS

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ABSTRACT

Some aspects of implementing multimedia services using LAN futures are studied with focus on video/audio information transmission. Consideration on compression methods and comparative results are shown. The data flow parameters are estimated for different casses. Using of video-teleconferencing systems for different purposes were took into account.

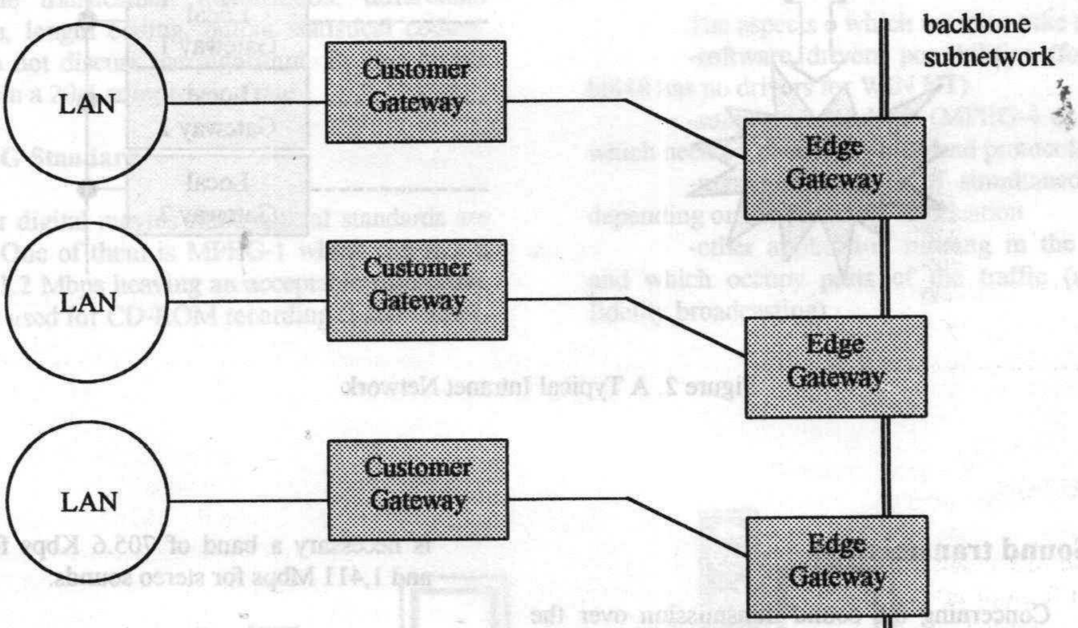

Figure 1. LAN Interconnection

1. INTRODUCTION

Using the Intranet futures it is possible to implement some interesting application based on well known devices and software. The authors experiments on LAN support for implementing a multimedia learning center, some particular problems and solution are take into account.

Some special services were tested in order to find some LAN limitations.

2. THE NETWORK STRUCTURE

The network structure is organized around a main backbone having a 10/100 Mb/s speed (see figure 1). The network element terminating the Customer Access Network (CAN) is the Edge Gateway (EGW); the network element connecting the Customer application to the access line is the Customer Gateway (CGW). To connect more networks this interconnection is accomplished by providing a LAN bridge for similar interconnection or a LAN gateway for dissimilar interconnection.

3. MULTIMEDIA ON NETWORK

A standard Intranet network has specialized servers doing different kind of access: web, ftp, mail, news, proxy.

Particularly studied applications are sound and video distribution using Intranet support: video/audio teleconferencing, video/audio broadcasting, video/audio supervising (figure 2).

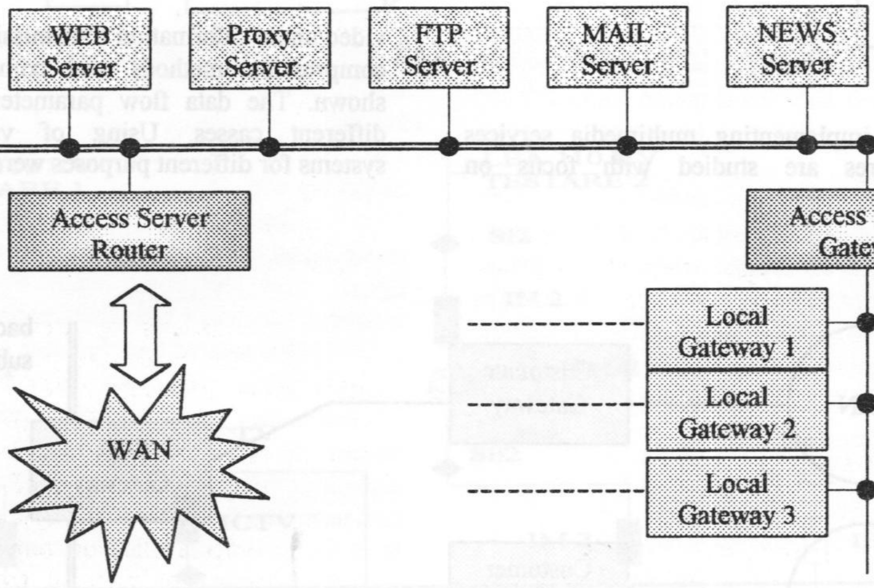

Figure 2. A Typical Intranet Network

3.1. Sound transmission

Concerning the sound transmission over the network a simply calculus show that for a limited band voice transmission, like phone vocal band, A PCM (pulse coded modulation) is used with 7 bits (in America and Japan) or 8 bits (in Europe) A/D samples conversion. Transmitting 8000 samples per second, a simply calculus gives a necessary data transmission rate of 56.000 bps, 64.000 respectively. The frequencies greater than 4 KHz frequencies are rejected.

For high fidelity sound (CD quality) we have to cover a frequency band of 22.050 Hz associated with 44.100 samples per second. Using a 16 bit conversion, we can have 65.536 amplitude levels (more than the human ear possibilities). Finally, there is necessary a band of 705.6 Kbps for mono sounds and 1,411 Mbps for stereo sounds.

As we can see, for high quality voice transmission over a LAN network without compression algorithm it is possible to transmit no more than 6 high fidelity voice channels simultaneously.

3.2. Video Transmission

For video transmission is necessary to send minimum 25 frames per second (to have continuity). Each frame usually has 640x480 (VGA), 800x600 (SVGA) or 1024x768 (XGA) pixels dimensions. For each pixels are used 8 bits for each color. So, we need (1024 x 768) pixels x 3 colors x 8bits = 472 Mbps binary speed.

Because the twisted pair network allow 10 Mbps, for transmitting video information it is necessary to make some compromises: reducing the number of colors (till 256 or 16 gray level or 256 color instead), reducing the number of pixels (the dimension of the image).

In order to increase the video delivery performance on the networks compression algorithms are used. Generally to coding methods are used: Entropy Encoding and Source Encoding. Entropy Encoding treats data bits without considering their particular signification. It has no loses and it is completely reversible. Source Encoding take into account the associated information for each bit or group of bits. Some information may be rejected with no important quality loses. The used encoding methods are: differential encoding, transformations (Fourier Transformation, Discrete Cosine Transformation) and vector quantization.

3.2.1. JPEG Standard

For static images (photos) JPEG standard is used. It supposes few steps: block preparation, discrete cosinusoidale transformer, quantization, differential quantization, length coding, output statistical coding. Here we do not discuss the algorithm. By this means we can obtain a 20:1 compression rate.

3.2.2. MPEG Standard

For digital movies some special standards are developed. One of them is MPEG-1 which can rich a bit rate of 1.2 Mbps heaving an acceptable resolution. It is meanly used for CD-ROM recording (I and video). The next step is MPEG-2 initialy developed for high quality video broadcasting. Now it includes some MPEG-3 unfinished standard futures concerning high-resolution video transmission. It needs 4 to 6 Mbps bits rate and is basicaly used for high distance video transmissions (TV broadcasting).

Compression rate is generally from 3 Mbps to 100 Mbps for HDTV, normally 3 to 4 Mbps.
In the same family we have MPEG-4 standard used for medium resolution video-conferencing at 10 frames per second with 64 kbps bandwidth.

So, a 10 Mbps network allow more than 100 simultaneous video/audio conferences if using the MPEG-4 standard.

Our user application consists in using video conferencing kit support like supervising system on the local network (figure 3). We try different video conference capture cards like BT848 from Brooktree and CL 5480 from Cirrus Logic. Commercial software was used like VDOPhone 3.0, Creative Video WEB Phone 3.0 , VocalTec Internet Phone 4.0.
The technical performances are quite similar in similar condition.

The aspects o which we must take care are:
-software drivers possibilities (for example, bt848 has no drivers for WIN NT)
-supported standards (MPEG-4 or compatible which need 64 kbps data flow) and protocols
-maximum number of simultaneous session, depending on the particular application
-other application running in the same time and which occupy parts of the traffic (audio high-fidelity broadcasting).

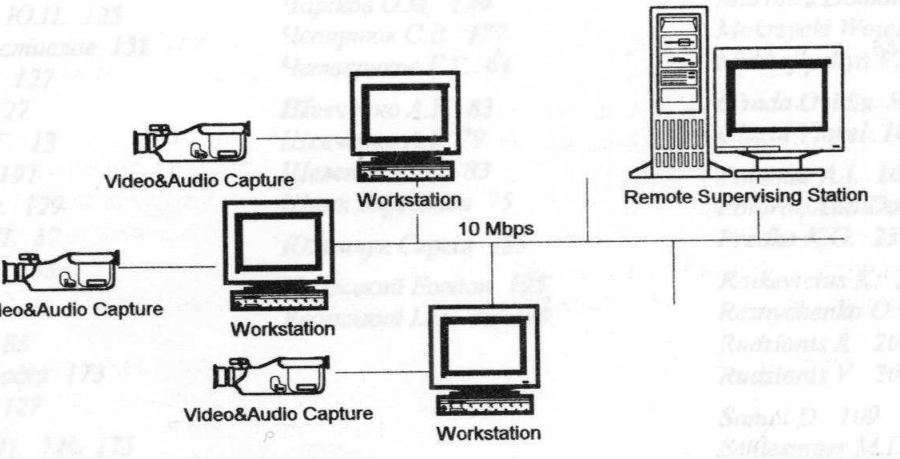

Figure 3. Video Supervising System on LAN

CONCLUSION

In order to have optimum results and avoid traffic congestion it is necessary to have in attention the network possibilities.

Depends on the particular application, some traffic calculus have to be made. As it was showen, for simultaneous teleconferencing or simultaneous video supervising a simply calculus (worst case) could confirm or not the local network state

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