Network Analysis using the FEC Network Monitor

1. Program Task

This program simulates voice communication via IP networks and provides useful information for evaluating network quality. To accomplish this, data packets are transmitted between two computers, representing existing telephone connections, and the time required for these packets to be sent/received registered. Funkwerk Communication Systems GmbH recommends testing of the network quality using this application as a preparatory stage prior to VoIP installation, or when problems arise with echoes/voice quality in an existing VoIP installation.

2. Prerequisites

This program must be started on at least two (2) computers, with one computer simulating the server, and the other(s) the clients. The program runs under all common Microsoft operating systems. If an existing and up-and-running FECP SIP-Serv installation is used as a server case analysis can also be conducted for Linux systems at the server end.
Important: The program can be started on a maximum of three (3) computers simulating clients. If more than three clients are simulated simultaneously there is a risk that the results of the analysis will be invalidated.

3. Background Information

The data packets that are transferred for voice communication are composed of a particular number of bytes (for example, 240 bytes for FEC SIP-Serv). A certain amount of time is required for transforming voice into IP packets. This time is required, for example, to convert the buffer from the soundcard to standard voice data using a codec, to compress it, where required, and transform it into IP packets via the IP stack. The time required for transmission of the data packets between the computers is an indication of the quality of the network. The FEC Network Monitor analyzes the influence of the IP stack on the network.

4. Procedure

The FEC Network Monitor continuously sends out data packets from the client to the server. The server then sends the data packets back directly and the time period from the point when the data packet is sent to the time when that same packet arrives back at the client is measured as "delay". This delay time may fluctuate, depending on the quality of the network and the current network traffic volume. Typical values are 150 ms for 240-byte packets. The influence of the codec, or the soundcard is not taken into consideration in this procedure.

If a time difference occurs between the sending of two data records and the receiving of two data records, a so-called "gap" is created. This gap can result in perceptible interruptions in voice transmission (brief crackle). Fluctuation of these gaps is referred to as "jitter".

5. Installation and Use

a) Server mode

If you already using an FEC SIP-Serv program and this program is active, you can use that program for network analysis at the server end instead of the FEC Network Monitor. It does not matter whether FEC SIP-Serv is installed under Windows or Linux operating systems. Continue as described under Client Mode b).
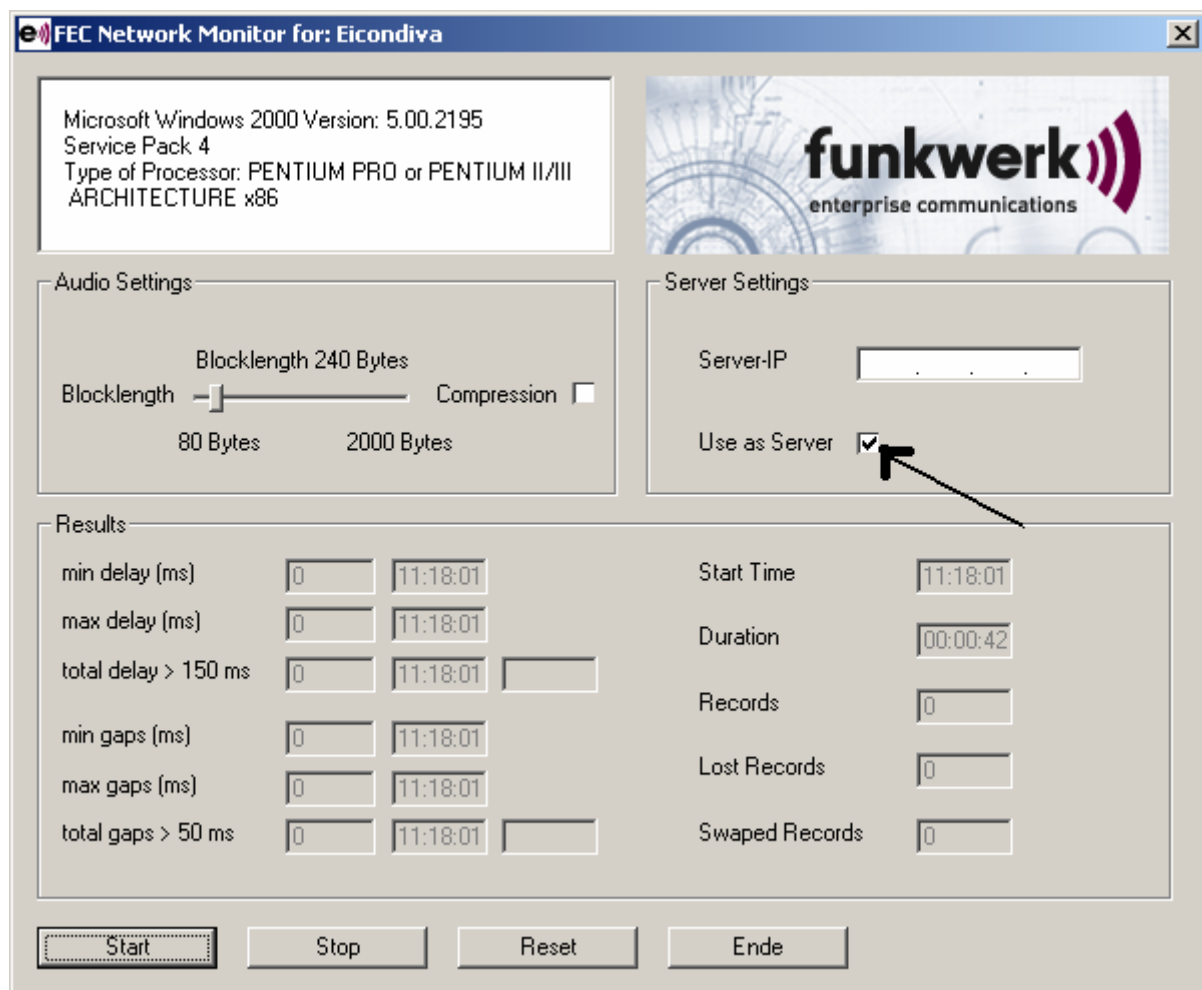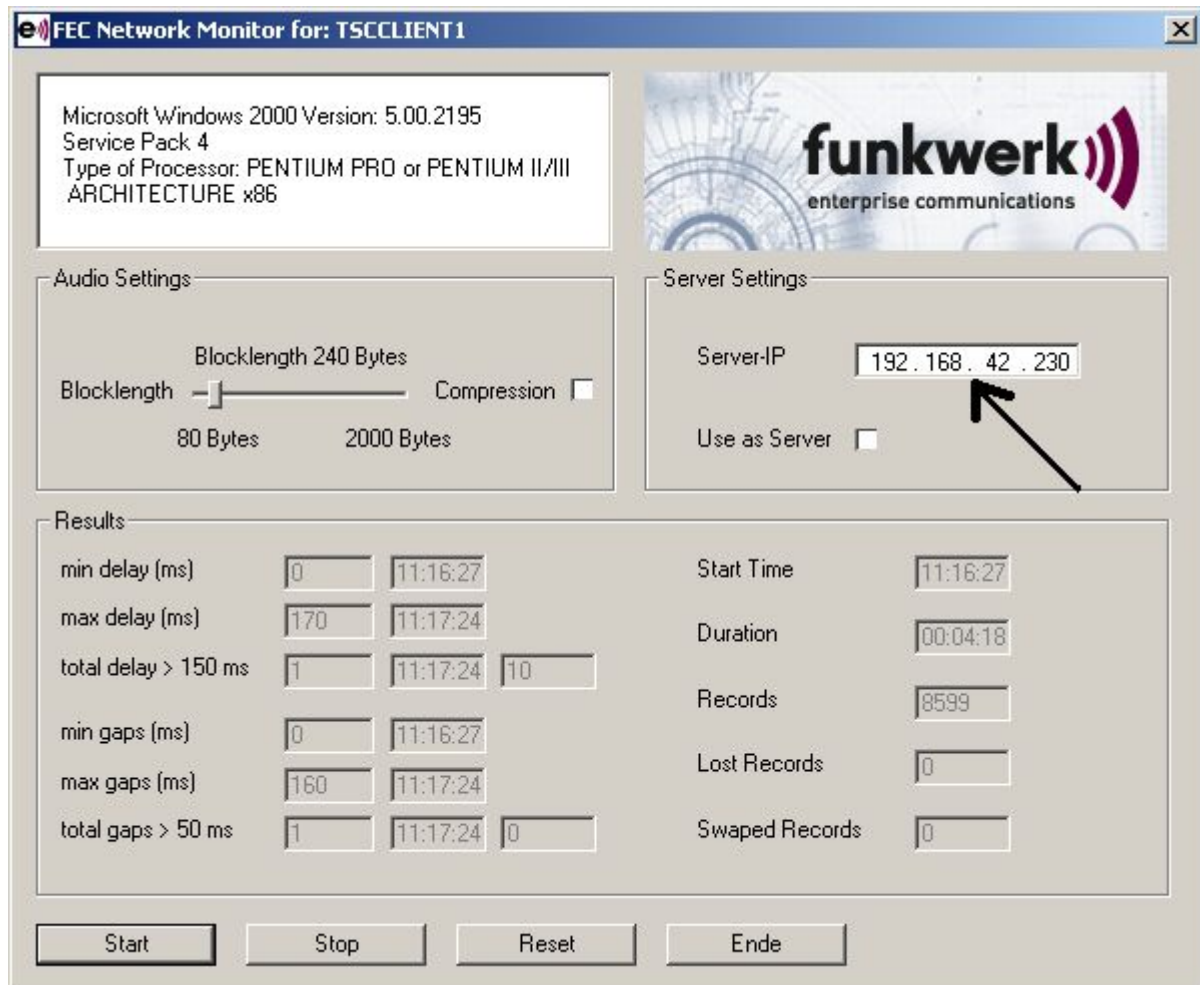

Start the FEC Network Monitor at any computer by executing the program file fec-monitor.exe. In the Network Monitor check the box "Use as server", see figure on next page.

Click the button "Start". You can change the block length using the mouse to move the slider and activate compression by checking the box. Under normal circumstances you should leave these settings in their default state however (block length 240 bytes, compression not checked (de-activated)).

b) Client mode

We recommend that you first create a directory on your computer, copy the file monitor.exe into this directory and then execute the program file from that location. This has the advantage that file generated by the FEC Network Monitor ("Monitor.txt") with the registered delay times will also be written to this directory.

Start the FEC Network Monitor by executing the program file fec-monitor.exe. In the field "Server IP" enter the IP address for the computer on which the FEC Network Monitor is to be started in the Server mode, or on which an FEC SIP-Serv has been started, see figure below.

Click the button "Start". The button "Stop" halts registration of the analysis; use "Reset" to return "Duration" to zero and click the button "End" to quit the FEC Network Monitor.

Let the simulation test run in parallel to actual production operation for at least 30 minutes. The longer a simulation runs, the more closely representative the analysis for the network. The FEC Network Monitor

displays the analysis results directly on screen and also generates a log file ("Monitor.txt") in ASCII format in the directory cited above.

6. Evaluation/Interpretation of Data

a) Application display

The display shown below appears in the client monitor when you start the FEC Network Monitor in the server and client mode. The displayed values are yielded on the basis of your current network conditions and are described in detail below. The values given in red are important for evaluating the performance and the quality of the network with regard to voice communication via IP protocol (Voice over IP – VoIP).

1. Start time: Time at which the (client) monitor was started.

2. Duration: Current duration of analysis, i.e. the time during which packets have been transferred back and forth across the network.

3. Records: Number of records transmitted up to now, from the time when the analysis was started (FEC SIP-Serv data packets)

4. Lost records: Number of records lost since the analysis was started.
Limit: 1% of all records *(3.)*

5. Swapped records: Number of records swapped (exchanged) since the analysis was started.
Limit: 1% of all records *(3.)*

6. Min delay – in ms.: Lowest delay, in ms, that has occurred since the start of the analysis.

7. Time of min delay: Time at which minimum delay occurs

8. Max delay – in ms.: Greatest delay, in ms, for received records that has occurred since the start of the analysis. *Limit: 150*

9. Time of max delay: Time at which maximum delay occurs

10. Total delay > 150 ms.: Total number of delays that have been received with a delay greater than 150 milliseconds. *Limit: 5% of all records* *(3.)*

11. Time of total delay: If 10. = 0, this time = time (7.).
If 10. <> 0, this time = time (8.)

12. Average delay – in ms.: Average delay value, yielded from all sent and received records. *Limit: 80*

13. Min Gaps – in ms.: Lowest gap value, in milliseconds, that has occurred since the start of the analysis

14. Time of min gaps: Time at which the minimum gap occurs

15. Max gaps – in ms.: Maximum gap value, in milliseconds, that has occurred since the start of the analysis. *Limit: 50*

16. Time of max gaps: Time at which the maximum gap occurs

17. Total gaps > 50 ms.: Number of all received records for which a gap greater than 50 milliseconds has occurred. *Limit: 2% of all records* **(3.)**

18. Time of total gaps: If 17. = 0, this time = time (14.). If 10. <> 0, this time = time (16.)

19. Average gap – in ms.: Average gap for all sent and received records. *Limit: 20*

Synopsis:

The following values are of great significance for quickly and reliably determining your network quality:

- Lost and swapped records (**4**/**5**). Neither of these values should exceed 1%, relative to the total number of records.

- Delay: The maximum delay (**8**) should be less than 150 ms (otherwise a situation could occur in which the calling parties begin interrupting each other, as voice transmission begins to be delayed). The number of delays greater than 150 ms (**10**) should not exceed 5%, relative to the total number of records. Also a critical criterion here is whether the delays are distributed over time (non-critical), or whether they occur closely together, successively (critical). You can extract this information from the log file that is generated (see below). The average delay for all records (**12**) should be less than 80 ms.

- Gap: The maximum gap (**15**) should be less than 50 ms (otherwise the interruptions in voice transfer become perceptible). The number of records with gaps greater than 50 ms (**17**) should not exceed 2% of the total number of records. Also a critical criterion here is whether the gaps are distributed over time (non-critical), or whether they occur closely together, successively (critical). You can extract this information from the log file that is generated (see below). The average gap value for the total records (**19**) should be less than 20 ms.

b) Values given in the log file

In addition to the presentation of the values directly in the application, the log file "Monitor.txt" is also generated which reflects the chronological sequence of the recorded values. The first line of the log file gives the designations for the recorded values. The following values are also given in the log file:

a) Time

b) Total, for all registered records

c) Lost records

d) Swapped records

e) Minimum delay f) Maximum delay

g) Total delays > 150 msh) minimum gap

i) Maximum gap

j) Total gaps > 50 ms

A new entry is created in the log file (new line added) when there is a change in the values (c-j). When this occurs, the time and the maximum value registered for all transferred records is additionally logged at the beginning of the new line. Clicking "Reset" clears all the values for the current analysis, and the values are restored to their original amounts (zero).

Example 1 (High network quality with regard to VoIP):

This example illustrates an analysis (up to the second reset) lasting about 30 minutes, with approximately 55,000 records transferred and with very good delay/gap values. As can be seen from this example, there are relatively few entries (lines) for this large number of records, and the increase in delay from 30 to 40 ms (arrow) occurs only after 20 minutes, or after around 32,000 records.

Example 2 (Low network quality with regard to VoIP):

This example illustrates an analysis lasting about 2.5 minutes, with around 4,800 transferred records and poor delay/gap values. As can be seen from this example, there is a relatively large number of entries (lines) for this small number of transferred records, indicating that this network does not exhibit stable transfer criteria. The number of maximum delays/gaps increases for nearly every record that is transferred (between 4808 and 4830). The number of delays greater than 150 ms also increases steadily. The arrow marks the first point where a problem arose in the network. The network produced normal values in the first two minutes (up to the entry marked by the arrow).