OPERA APPLICATION NOTES (1)

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1 INTRODUCTION

1.1 What is Covered by this Paper

This paper will present some typical applications of OPERA™, the premier voice quality analyzer from OPTICOM. Thereby the main focus will be on the "physical layer". We will concentrate on how to connect OPERA™ to various devices/networks under test and how to configure OPERA™ for proper data acquisition, especially in non-trivial situations. There is also a short introduction to OPERA™, but it will give a general overview only and it is not intended to replace the OPERA™ user's manual. Also, the evaluation of the acquired data will be covered only very briefly.

Although the emphasis of this paper is clearly on the telecommunications applications, users of the broadcast version of OPERA™ will benefit from reading this paper too. Some applications are identical, although this is not always obvious (e.g. measurement of terminal equipment or multimedia terminals uses the same setup as measuring ISDN codecs).

1.2 Recommended Further Readings

For those readers that are not yet familiar with OPERA™ or are interested in the details of the psychoacoustical analysis, we recommend having a look at the OPERA™ user's manual. The basics of psychoacoustics are generally covered very well by the ITU recommendations P.861 and especially BS.1387. These two documents are enclosed in all OPERA™ manuals. For a more practical oriented introduction to perceptual measurements, especially with regard to VoIP we recommend our white paper "State of the Art Voice Quality Testing".

1.3 Short Introduction to OPERA™

OPTICOM's new generation of voice quality analyzers, called OPERA™ – short for „Objective Perceptual Analyzer“ – represent the latest development to objectively evaluate and assure the quality of compressed speech and wide-band audio signals by modeling the human ear. With OPERA you can achieve a comprehensive analysis of the end-to-end quality, from the studio source to the receiver, or from the caller to

the callee. OPERA's flexible scalability may range from a single stand alone tester up to powerful network-wide setups with distributed systems sharing information over IP.

The open framework concept of OPERA™ allows to add advanced measurement algorithms as plug-ins in the future as soon as they will become available. In addition, user defined measurement algorithms may be integrated upon request. OPERA™ is available as a software suite, a completely pre-installed workstation system and a completely pre-installed portable system. The operating system currently used is Microsoft Windows NT 4.0.

In the following, we want to give a short overview of the functionality and the operation of OPERA™. Basically, a measurement of end-to-end quality comprises two steps, first signal acquisition and second the analysis of the test signal. Within OPERA these two steps are always performed by two separate programs. **OptiCall**™ is used for the data acquisition, and **OPERA**™ for the evaluation of the speech/audio quality. Since software version 3.0, OptiCall™ can also be used for wide band audio measurements. The only exception from this separation of data acquisition and evaluation are online measurements. This type of measurement is characterized by a streaming input signal. Streaming input signals are acquired by OPERA™ directly, and evaluated in realtime.

The advantages of keeping the data acquisition separate from the evaluation are:

- OptiCall™ and OPERA™ are not overloaded with unnecessary functionality. Both serve completely different purposes and both are optimized for easiest operation and best performance. Although the user requires two programs, each of them is much easier to use than one thick client.

- Since the actual evaluation of the speech/audio quality is performed file based, measurements can be repeated several times with the same test data but different algorithms. In the case of VoIP networks, for example, this means that both, the echo as well as the degraded speech signal are acquired simultaneously, under one and the same condition of the network, and evaluated afterwards. This means that both results, the echo measured at the near end as well as the quality of the transmitted speech measured at the far end represent the same conversation. Other systems acquire the data in two different runs. One run for the speech data and another run for the echo data. In such a case both runs may describe different situations in terms of packet loss etc.

- Since OPERA™ as well as OptiCall™ can be run fully automated and unattended, as well as remotely controlled, there is the possibility that various users on a network use some fully equipped OPERA™ systems for the data acquisition only, while the local PC will run the OPERA™ Software Suite for the evaluation of the quality.

1.3.1 Signal Acquisition Using OptiCall™

The principle of OptiCall™ is easy. First a connection is established between any two available interfaces and then the speech/audio data are transferred through the selected interfaces. Simultaneously, OptiCall™ records all the incoming data on full duplex connections. Although OptiCall™ was originally developed for telecom applications and telephony interfaces only, it can make use of audio interfaces, too. If either for the originating or the terminating side (or even both sides) an audio interface is selected, the call control is simply skipped for this particular interface, and the audio interface is immediately set to the connected state. Using the audio interfaces, OptiCall™ can also be used for the assessment of wide band audio codecs (like e.g. MPEG or AAC codecs), multimedia terminals or any other terminal equipment.

One of the key features of OptiCall is that you may run the user interface on a local machine, but the call may be executed on a remote machine. This is accomplished by simply telling OptiCall the name of the remote machine.

OptiCall offers the following four operating modes: Loop, Origin, Termination and Terminate All.

- In the **Loop** mode, one instance of OptiCall controls both, the originating as well as the terminating side of a connection. Both sides must be connected to the same OPERA system.

- If set to the **Origin** mode, OPERA is connected to the originating side of a connection. OptiCall will initiate a phone call on the specified interface and start transmission as well as recording of the audio data as soon as the call is answered by the terminating side (which may be another OPERA system, or just another instance of OptiCall running on the same machine).

- In **Termination** mode, OPERA is connected to the terminating side of a connection. OptiCall will wait for a incoming call on the specified interface and start transmission as well as recording of the audio data immediately. The origin of the call may be another OPERA system, or again just another instance of OptiCall running on the same machine.

- The **Terminate All** mode does exactly what the Termination mode does. The only difference is, that the system will wait for incoming phone calls on any of the line interfaces, and after one call is completed, the system will automatically wait for the next call.

1.3.2 Dynamic Driver Latency Compensation (DDLC)

Since OPERA is based on Windows NT, per definition a non-realtime operating system, some special means are required in order to obtain reliable latency measurement results. OPTICOM developed the DDLC method which compensates for the non-deterministic delays of the operating system and driver. Most of the time DDLC is invisible to the user, it will simply be applied to the recorded data only. Only under some circumstances DDLC must modify some timing parameters and repeat the test call. In general, this happens transparently to the user. Since DDLC requires access to the originating, as well as to the terminating side, it is active in the Loop mode only. Using DDLC, the resulting accuracy of the delay measurements is in the range of approximately $\pm$5ms for 95% of the test calls. Without DDLC it drops down to $\pm$50ms. Please note that this is the accuracy due to the data acquisition only. OPERA itself measures the delay between two file based signals within the accuracy of one sample.

1.3.3 Evaluation of the Recorded Data Using OPERA

Since the intention of this white paper is to report typical applications and to clarify the hard- and software settings required for the data acquisition part, the evaluation of the data is covered only very briefly. An in depth explanation which measurement algorithm should be applied under which circumstances and how to accomplish this is contained in the OPERA manual. Here we only want to mention the principle of all perceptual measurements in so far as this has an effect on the data acquisition.

In general the evaluation of the data follows the same steps for all measurement algorithms (except for Echo Return Loss measurement). There are always **two input signals** which are compared to each other.

One of the signals is the unimpaired reference signal (the signal which was sent to the device under test). The other signal is the degraded, or test signal. The measurement algorithm will now identify all audible

differences between those two signals. In a first step, the delay and level differences between the reference and the test signal must be compensated for. This process is also called the time alignment and scaling. OPERA is capable of handling up to ±10s of delay between the two input signals and up to 60dB attenuation. The next step is to calculate what sound a human being would perceive from the reference signal and what one would perceive from the test signal. This is effectively modeling the peripheral ear. Any differences between these two results could now theoretically be perceived by the listener. In a last step these differences are fed into a cognitive model which simulates effects like disturbance and forgetting.

2 EXAMPLE SETUPS

In this section of the paper we will describe several measurement setups as they are typically used. The list of examples is by far not complete, and the possibilities offered by OPERA's flexibility are numerous. We tried to choose the most frequent applications and those that can be easily transferred to other, similar measurement problems. The list is dynamic and may be updated in the future.

All descriptions of the setups follow the same structure: There is a short description and a sketch of the setup, followed by the required prerequisites and a detailed description.

2.1 General Remark

This chapter covers some considerations which are common to all of the following measurement setups. This section is structured like a tutorial and it is recommended that you go through all of the examples, even those that do not match your application. Especially understanding the first two setups is fundamental for understanding all the others.

2.1.1 Selection of the Reference file

As a rule of thumb the reference file should be a signal that comes as close as possible to the kind of signal which shall be applied to the device under test in real life. If e.g. you design a special headset for female call center agents, you should use a test stimulus which contains mostly female speech. If the device should be used by male and female users as well as children, you should perform separate tests with typical stimuli for each of these cases. For the assessment of MPEG audio codecs which are used for the transmission of high quality music between broadcast studios, you should test your device with real music. Especially with wide band music codecs you should select a variety of at least six to ten different test samples, since the performance of audio codecs differs widely depending on the test material.

The duration of the test sequence should be within the range of approximately four to eight seconds. Longer tests will lead to averaging effects (short distortions may be averaged down by a long but almost perfect transmission) and shorter sequences may not be long enough to contain representative parts of the signal.

The sample rate of the reference file is frequently already defined by the algorithm that shall be used for the evaluation of the recorded data. PEAQ according to ITU-R BS.1387 for example requires 48kHz sample rate, although the implementation in OPERA will deliver reliable results at 44.1kHz, too. Most speech quality measures are defined for 8 and 16kHz sample rate only.

The selection of the sample format should be mainly driven by considering the capabilities of the underlying hardware. While using the audio interfaces provided by OPERA, it makes sense to select the 16bit linear. Since currently all measures use 16bit linear internally, any higher resolution, although supported by the

hardware will not result in more accurate measurements. When performing test calls with the voice board, the sample format should be 8bit mu-law or 8bit A-law (G.711). Otherwise the measurement will include at least one more step of encoding, since the DSP on the voice board will convert all input data back to G.711.

A set of typical wide band audio examples is mentioned in the ITU-R rec. BS.1387. Speech samples are also provided by the ITU-T, in the Series P Supplement 23.

2.1.2 Synchronization

Synchronization of A/D converters and sample rate converters is an important issue when using two audio interface boards. If each interface board is running on it's own internal clock, one board will acquire some samples more than the other one since the two oscillators will never run at exactly the same frequency. Consequently, the evaluation of the data will result in a varying delay between the reference and the test signal. To overcome this problem, OPERA provides various synchronization inputs. A detailed description on how these are applied is contained in the OPERA manual.

2.1.3 Important Settings of the Audio Interface Options

There are some points which have to be obeyed if a setup requires usage of the audio interface option. The detailed settings required are mentioned in the OPERA manual. In this place we only want to pin point some frequent sources of errors in setting up these interfaces.

The first point is that all boards used in OPERA provide some means of using the outputs of the audio interface to monitor any of the input sources. This works fine as long as the output is not used to play another signal, or even worse, if the output is fed back to the input. In both cases the "real" signal will be mixed with the monitoring signal. The monitoring feature may be switched off using the mixer control for the appropriate board from the Windows systray (the lower right corner of the screen, where you find several small icons). The detailed settings are mentioned in the OPERA manual.

The second source for misleading measurement results are misadjusted level controls of the audio interfaces. Again these settings are performed using the mixer control. Please note that the level, the balance and the headroom/trim must all be adjusted. We recommend using the highest possible settings of headroom/trim which is still compatible with your device under test. It is also highly advisable to leave the faders of the volume control in the zero position. The volume controls will modify the signals in the digital domain and thus increase the signal to noise ration significantly.

2.2 PSTN, Loop Measurement

A loop call is the simplest setup for PSTN measurements. OPERA is connected to two telephone lines and the test call is established between these two interfaces. Figure 2.1 shows the typical setup. Of course, this setup works for all kinds of interfaces offered by OPERA, for telephony as well as audio interfaces.

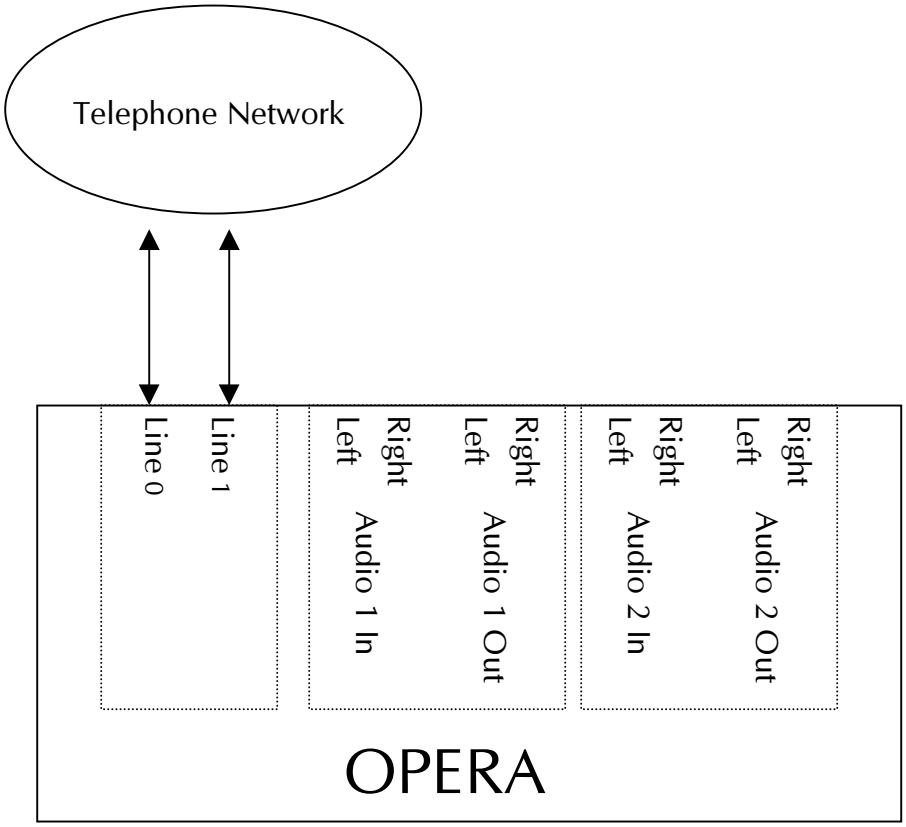

Figure 2.1: Setup for loop measurement

2.2.1 Prerequisites

1 OPERA base unit

1 Four port POTS interface option

1 Telecom measurement algorithm, e.g. PSQM+, PESQ and Echo.

2.2.2 Description

Two line interfaces of OPERA are connected to two analog telephone lines. OPERA will then take line 0 off hook and dial the number which is associated to line 1. As soon as line 1 rings, OPERA will take that line off hook, too, and start transmitting the selected speech reference file from line 1 to line 0 (please note the

direction). Simultaneously, it will record all signals which are received on both lines. The typical settings for OptiCall are shown in Figure 2.2. The recorded files for the example setup will be:

d:\test-line0.wav Degraded speech signal

d:\test-line1.wav Echo signal.

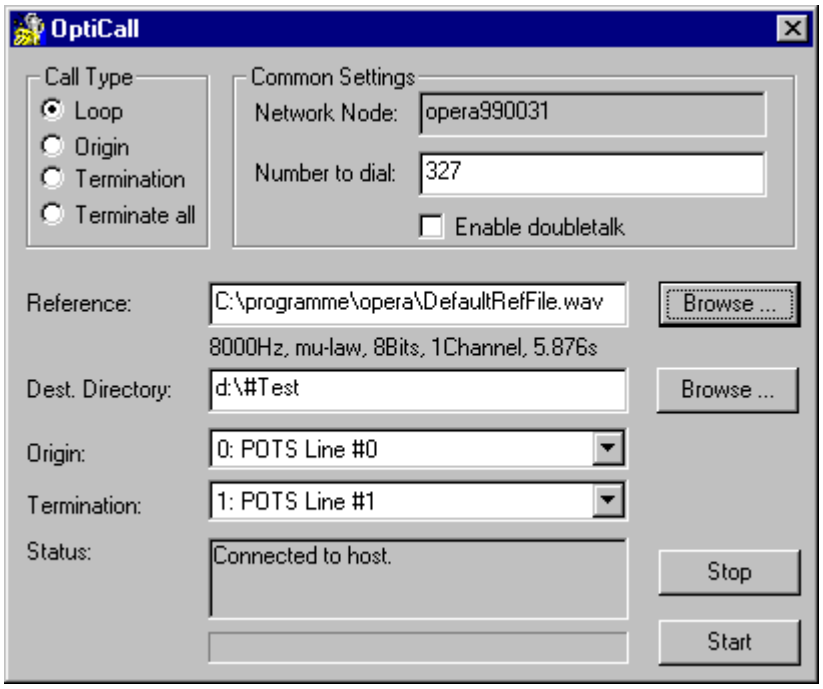

Figure 2.2: OptiCall Settings for PSTN loop call

2.3 PSTN, Distributed Call

Distributed calls allow for measurements between physically distant locations. Two OPERA systems are required then, and both are connected to at least one telephone line. An additional IP connection allows to remotely control both systems from one machine and to access the recorded files on the remote machine. Figure 2.3 shows the typical setup. Of course, this setup works for all kinds of interfaces offered by OPERA, for telephony as well as audio interfaces.

2.3.1 Prerequisites

2 OPERA base units

2 four port POTS interface option

1 Measurement algorithm, e.g. PSQM+, PESQ and Echo.

2.3.2 Description

One of the line interfaces of each of the OPERA systems is connected to e.g. the PSTN. In addition to that there should be a data network between the two OPERA systems. Assuming you are in Paris and somebody switched on the system in New York, you may want to start two instances of OptiCall on the machine located in Paris. Figure 2.4 shows the settings chosen in the two instances of OptiCall for this type of measurement. The left dialog is controlling the "local" system in Paris, while the right one is controlling the remote machine in New York.

The fields **Network Node** must be set to the network names of the OPERA machines involved. The system in Paris is configured to be the originator of the call, while the system in New York will terminate the call. The speech file is then transmitted from New York to Paris, and both systems will record what ever they receive. The resulting recorded files for the shown settings will be:

\\OperaParis\d\test-line0.wav	Degraded speech signal
\\OperaParis\d\testNY-line0.wav	Echo signal

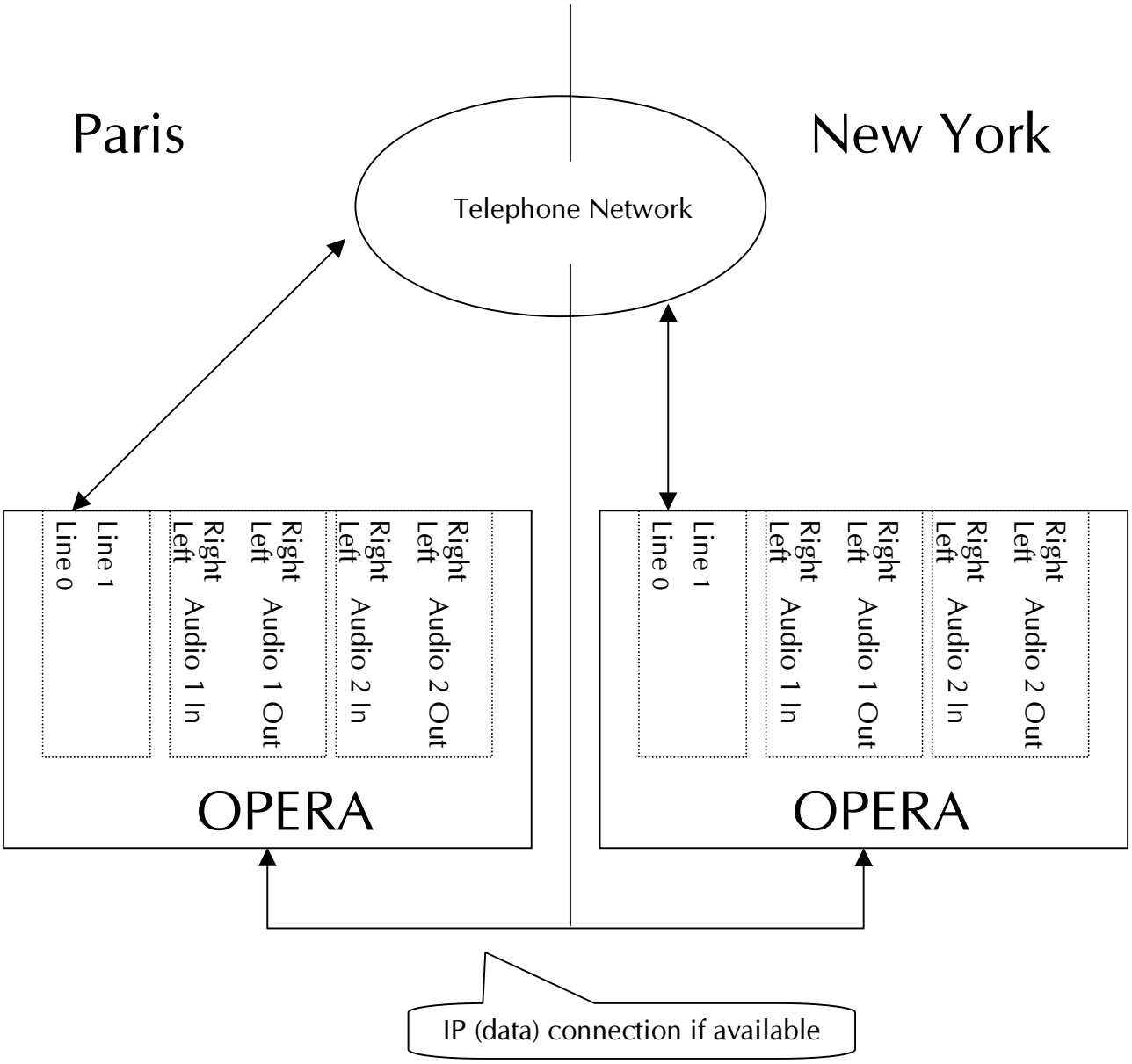

Figure 2.3: Measurement across geographical boundaries.

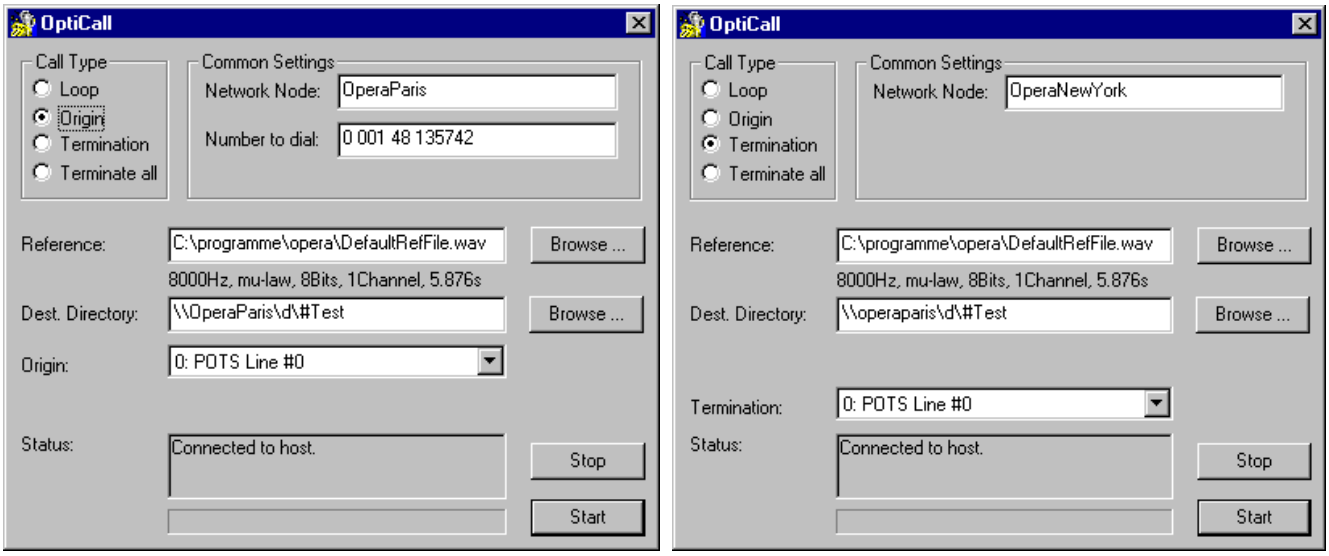

Figure 2.4: Two instances of OptiCall, controlling the system in Paris as well as the system in New York. Both instances are running on the machine in Paris.

Please note that a UNC path name is used to identify the recorded files. We highly recommend this in order to avoid confusion if the same path exists on both systems. All path names are relative to the machine on which the call is actually executed. If in the above example we had used D:\ instead of \\operaxxxx\d\... the recorded files would have been the following:

d:\test-line0.wav	Degraded speech signal (d drive on OperaParis)
d:\testNY-line0.wav	Echo signal (d drive on OperaNewYork)

Also note, that if both systems are told to store their recorded files on the same disk, you must either use different line interfaces or – much better – use a different root file name in order to avoid conflicts with the same file name used by both processes. In our example we used the root file name "Test" for Paris and "TestNY" for New York. In any case, the recording process will append the identifier of the line interface used for the recording to the file name.

If you have more than two OPERA systems and you would want to monitor the quality of your network permanently, the optional OPERA Control Center may be the best solution for you. The OPERA Control Center allows you to create a schedule of test calls which are executed automatically and fully unattended. Speech quality, Echo Return Loss (with real speech) as well as call timing information are automatically calculated. and reported in a table which may be viewed directly. In addition, several filters may be applied to the table of results, or the table can be exported to MS Excel for further statistical analysis.

2.4 VoIP, Measuring Multimedia Terminals

Since the applicability of OptiCall and OPERA is not limited to telephony interfaces, you may also apply it to directly measure the quality of terminal type equipment that may be attached to the audio interfaces. The corresponding setup and usage is very straight forward if your system is equipped with two audio interface boards. If you ordered the system with only one audio interface board, you can still use it for this application, just the setup is a bit tricky. In the following we will explain both setups.

The first setup, using two audio interfaces is also the setup which is recommended for the assessment of wide band audio codecs like MPEG Layer 3 (MP3) or AAC in a broadcast environment. Both setups have in common, that the call control must be performed manually, since OPERA does not know how to operate your equipment, of course.

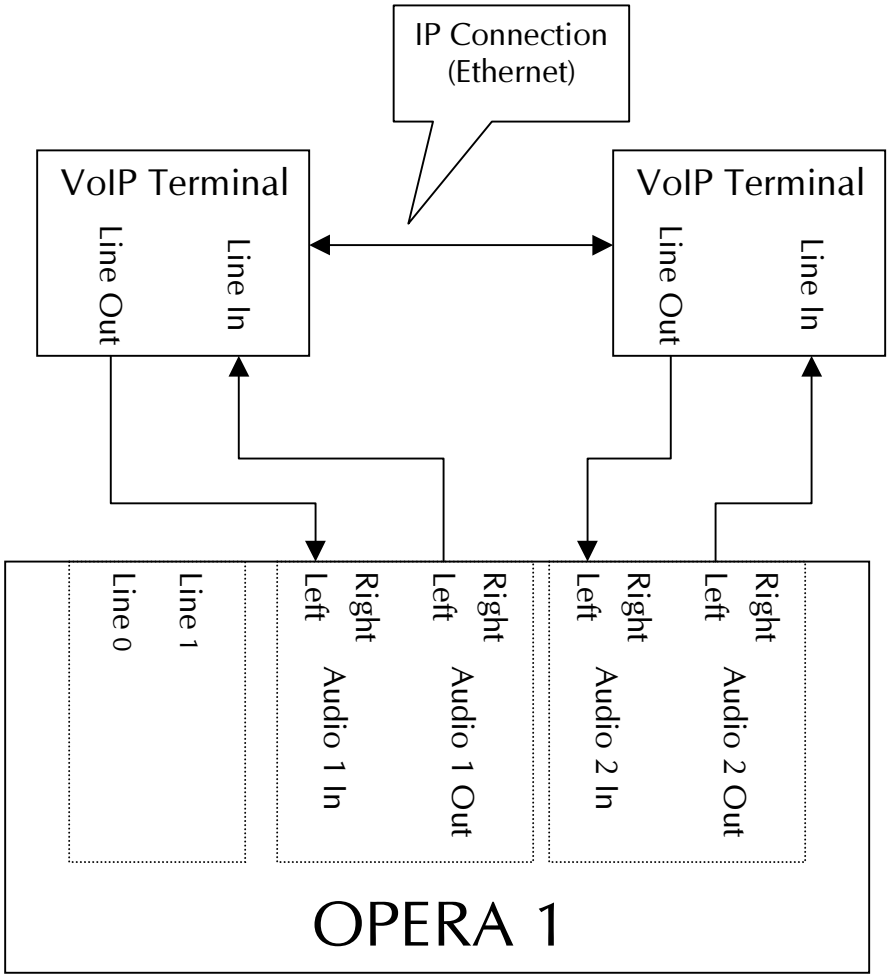

Figure 2.5: Measurement of VoIP / multimedia terminals with two sound cards.

2.4.1 If you have two Audio Interfaces

If your system is equipped with two interfaces, the setup is simple: Just connect the first terminal to interface 1 and the second to interface 2. You may then use OptiCall in all modes, exactly as in Figure 2.2 for the PSTN calls.

2.4.1.1 Prerequisites

1 OPERA base unit

2 Audio interface board options

1 Optional OPERA XLR adapter kit

1 Telecom measurement algorithm, e.g. PSQM+, PESQ and Echo.

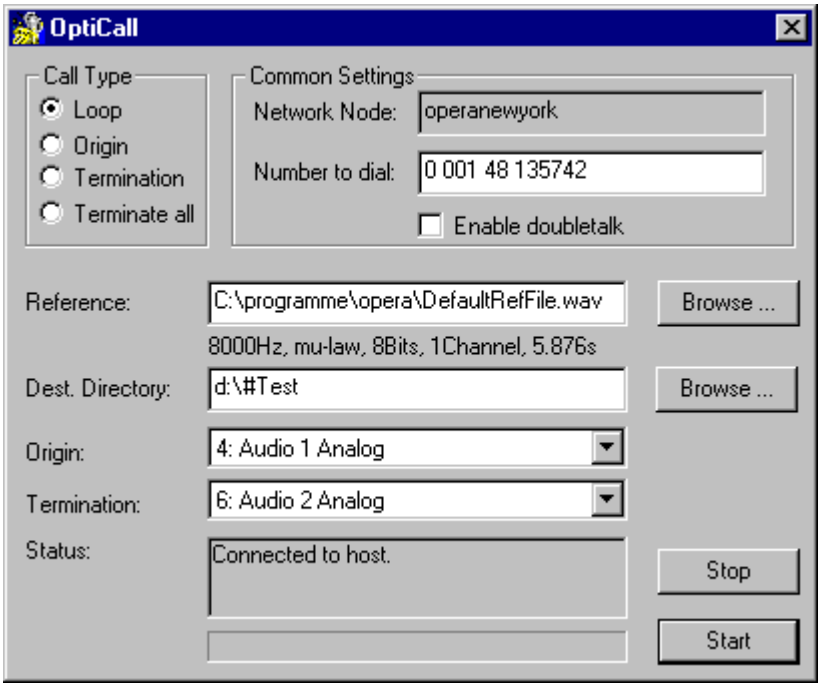

Figure 2.6: OptiCall settings for audio interfaces on the originating as well as the terminating side.

2.4.1.2 Description

This setup may require the optional OPERA adapter kit since most IP terminals and PC equipment in general use 3.5mm jack consumer type audio connectors. OPERA instead comes with professional, symmetrical audio interfaces with XLR connectors. Since OPERA's audio interfaces are potential free, no additional transformers etc. are required in order to avoid problems with hum etc. After connecting the system according to Figure 2.5, you are ready to run OptiCall. Figure 2.6 shows the correct settings of OptiCall for this mode. Since both, the originating, as well as the terminating side are bound to audio interfaces, the telephone number will be ignored. After pressing start, the system will immediately refer to the connected mode: It starts playing the speech file on Audio 2 while recording on both sides, Audio 1 as well as Audio 2.

The resulting recorded speech files will be:

d:\test-line4.wav	Degraded speech signal
d:\test-line6.wav	Echo signal

Please note that the index of the line used to form the resulting file name is the same as the number preceding the name of the interface in the drop down list box of OptiCall.

2.4.2 If you have only one Audio interface board

This is the typical configuration which most customers use that bought OPERA for telecommunication purposes. It is more economic than using two audio interface options, but the setup as in Figure 2.7 is slightly more complicated.

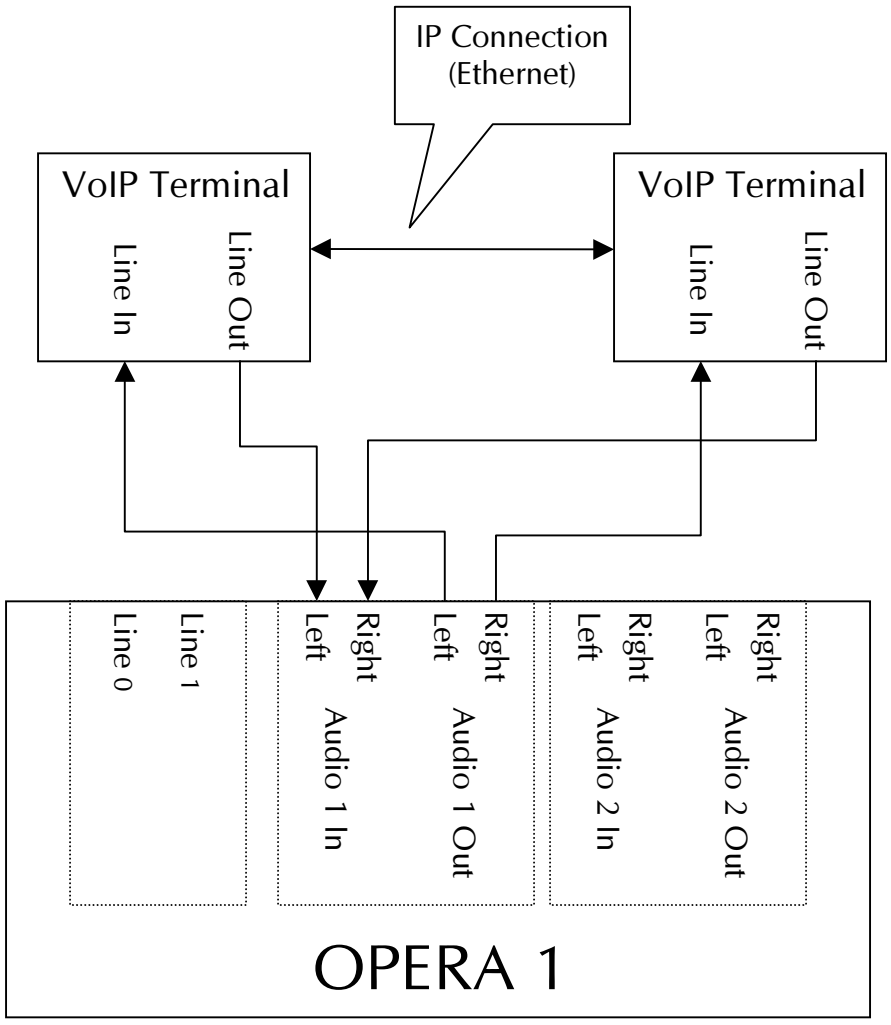

Figure 2.7: Measurement of VoIP / multimedia terminals with one sound card.

2.4.2.1 Prerequisites

1 OPERA base unit

1 Audio interface board option

1 Optional OPERA XLR adapter kit

1 Telecom measurement algorithm, e.g. PSQM+, PESQ and Echo.

2.4.2.2 Description

The setup according to Figure 2.7 shows that essentially terminal 1 is connected to the left channels of the audio interface and terminal 2 to the right channels. This setup requires a special stereo reference file that contains the speech sample in only one of the two channels and silence in the other. In this example we assume that the right channel is used. OptiCall should now be configured as in Figure 2.8. Please note that the Termination mode is required here. After pressing start, OptiCall will be connected immediately, and will start playing the reference file on Audio 1 while recording whatever it receives on Audio 1 simultaneously. Since our speech signal is carried in the right channel of the reference file, the signal will be played from terminal 2 to terminal 1. The output signal of terminal 1 will be recorded to the left channel of the received signal and the echo signal from terminal 2 will be recorded to the right channel of the same signal. The resulting file will be called:

d:\TestStereo-line4.wav

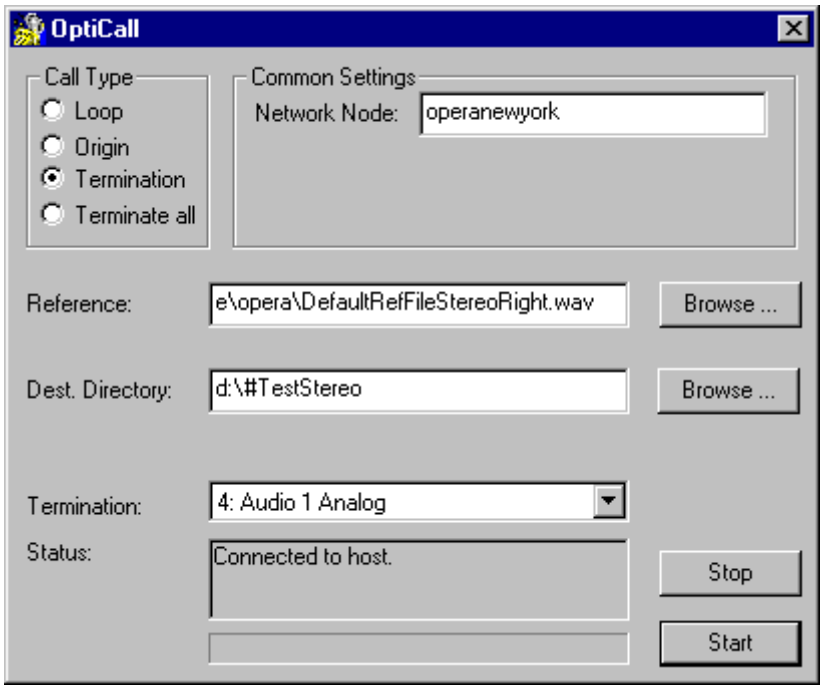

Figure 2.8: OptiCall settings for VoIP terminal assessment with only one audio interface option installed.

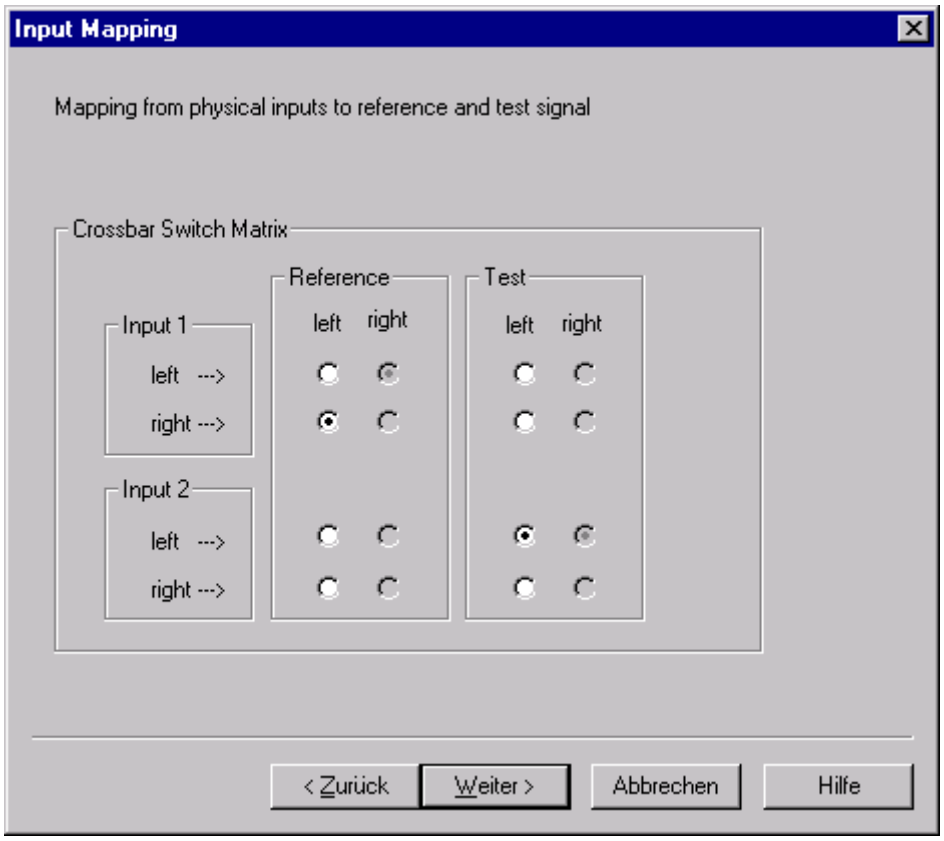

Figure 2.9: Settings of the crossbar switch matrix for speech quality measurement

This setup also requires some special settings within the OPERA software. Since the input and output files are stereo, you will have to adjust OPERA's cross bar switching matrix in the input mapping dialog according to Figure 2.9 for speech quality evaluation and according to Figure 2.10 for echo measurements. Both settings assume that the reference file was chosen as the input signal 1 and the recorded file was chosen as input signal 2.

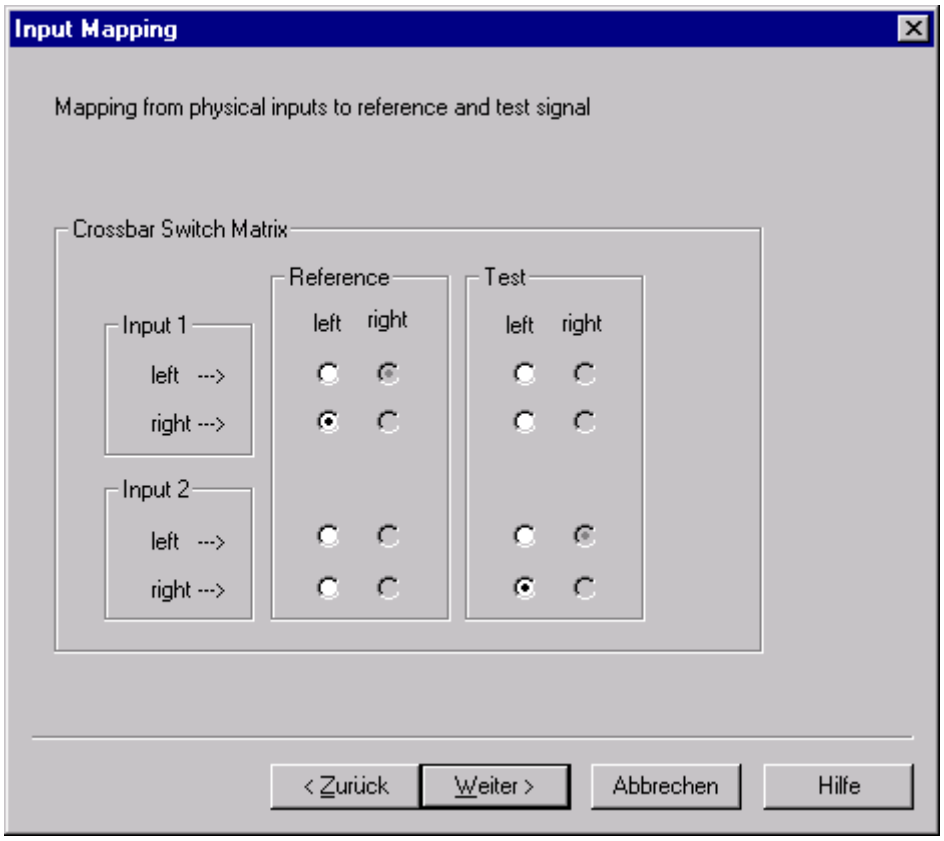

Figure 2.10: Setting of the crossbar switch matrix for echo measurement

Please note, that this setup does not allow for really accurately latency measurements at the moment, since DDLC is active in the loop mode only. Future versions of OptiCall however, may provide a switch to allow for using DDLC in the Termination and the Origin mode as well.

2.5 VoIP, Measuring PSTN to Multimedia Terminal calls

The setup below is essentially a mixture of the setups from 2.2 and 2.4. It requires that one telephone line interface of OPERA is connected to a VoIP gateway or terminal adapter, and the VoIP terminal equipment is connected to an audio interface. Setting up the call or answering the call can be handled automatically by OPERA on the telephony interface. On the terminal side, however, call control must be handled by the user.

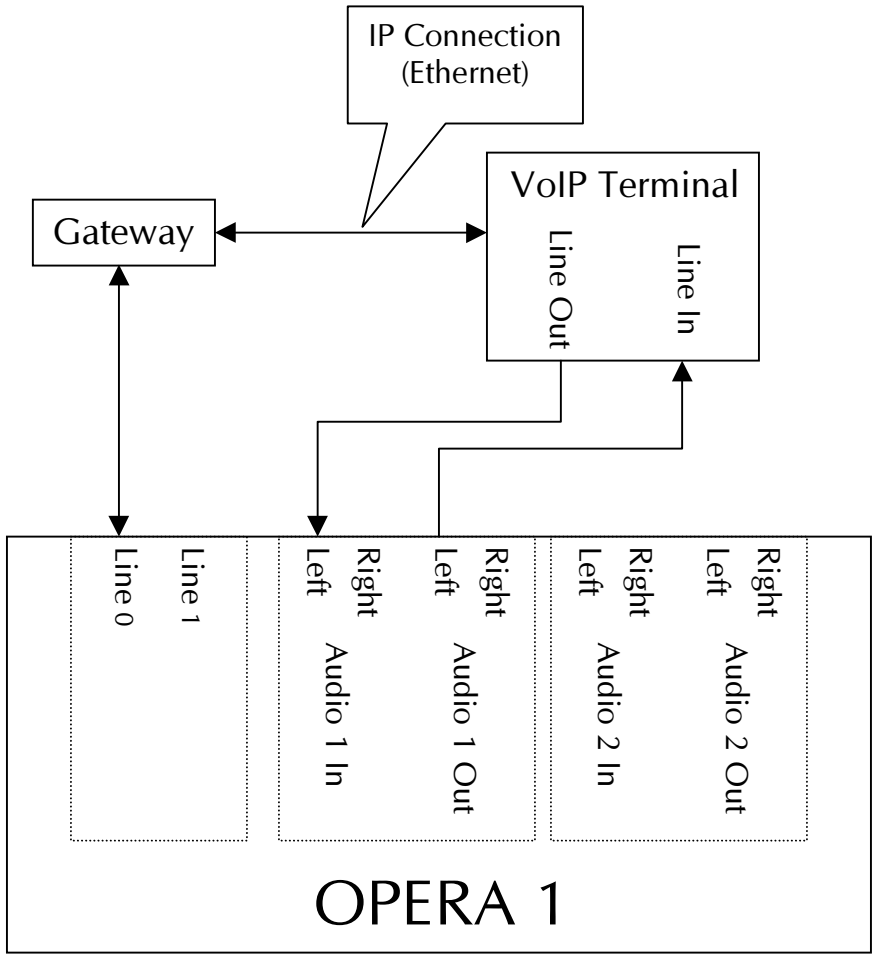

Figure 2.11: Measurement of VoIP terminals through a gateway.

2.5.1 Prerequisites

1 OPERA base unit

1 Four port POTS interface option

1 Audio interface board option

1 Optional H.323 terminal adapter (gateway)

1 Optional OPERA XLR adapter kit

1 Telecom measurement algorithm, e.g. PSQM+, PESQ and Echo.

2.5.2 Description

The setup according to Figure 2.11 may require the optional adapter kit since most IP terminals and PC equipment in general use 3.5mm jack consumer type audio connectors. OPERA instead comes with professional, symmetrical audio interfaces with XLR connectors. Since OPERA's audio interfaces are potential free, no additional transformers etc. are required in order to avoid problems with hum etc. With the shown configuration, OPERA can initiate calls from the gateway to the terminal as well as answer calls from the gateway to the terminal.

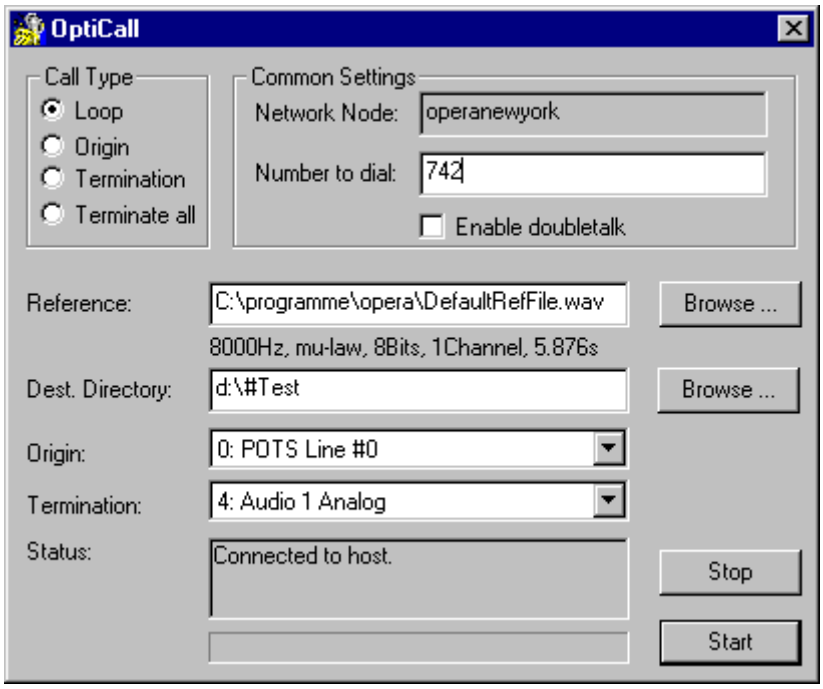

Figure 2.12: OptiCall settings for calls from the POTS interface to the audio interface

In order to initiate the call, set OptiCall to the Loop Mode, select "POTS Line #0" as the originating interface and "Audio 1 analog" as the terminating interface (Figure 2.12). OptiCall will then dial the given telephone number and switch to the connected state, as soon as the terminal accepts the call. The speech

file will then be transmitted from the audio interface through the terminal and the gateway to the POTS interface. The resulting recorded file names will be:

d:\test-line0.wav Degraded speech signal

d:\test-line4.wav Echo signal.

If you want to reverse the direction of the call, you must choose "Audio 1 analog" as the originating interface and "POTS Line #0" as the terminating interface (Figure 2.13). If you now click on start, OptiCall is waiting for an incoming call on the POTS interface. The call must be initiated by the user at the terminal. After the connection is established, the speech file is transmitted from the POTS interface to the audio interface. The recorded files in this case will be:

d:\test-line4.wav Degraded speech signal

d:\test-line0.wav Echo signal.

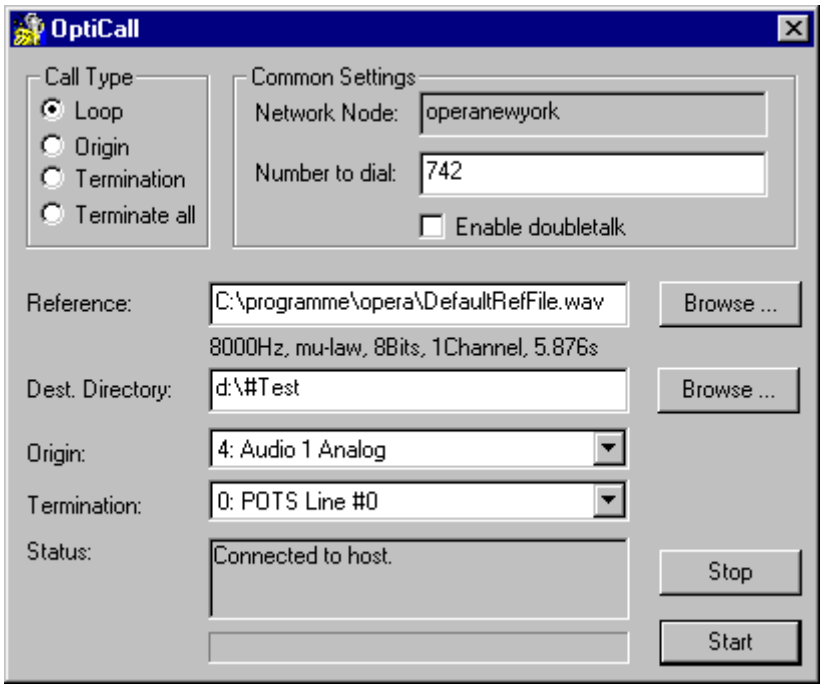

Figure 2.13: OptiCall settings for calls from the audio interface to the POTS interface

2.6 Measurements Including the Acoustical path

Since OPERA provides analog as well as digital audio interfaces, it can also be attached to almost any head and torso simulator (HATS). Essentially the setup shown in Figure 2.14 is similar to the one in chapter 2.5. The only difference is the type of equipment being tested. The following description therefore focuses on the peculiarities of this setup and a proper understanding of chapter 2.5 is assumed.

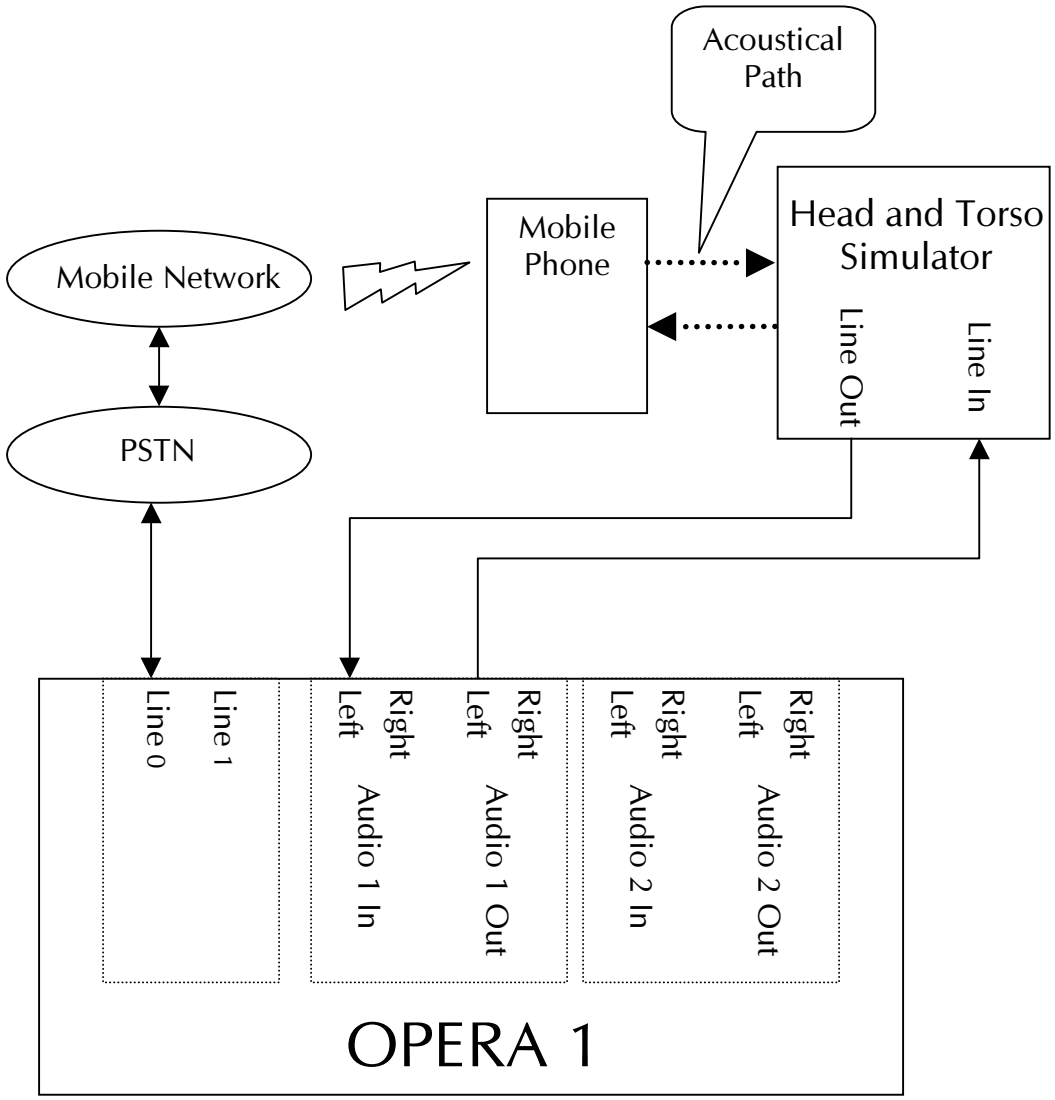

Figure 2.14: Measurement terminal equipment including the acoustical path.

2.6.1 Prerequisites

1 OPERA base unit

1 Four port POTS interface option

1 Audio interface option

1 Head and Torso Simulator (HATS)

1 Telecom measurement algorithm, e.g. PSQM+, PESQ and Echo.

2.6.2 Description

Using this setup, a test call is issued from the fixed PSTN to the mobile network. The mobile phone will accept the call and the speech file will be exchanged between the audio interface, the HATS, the mobile phone and back through the PSTN to the telephony interface of OPERA. Of course, as in 2.5, the direction of the call may be reversed just by changing the settings of OptiCall.

This measurement includes both, the acoustical path as well as the RF part of the mobile transmission. Including the RF part makes it very difficult to reproduce the measurement results. Normally several measurements have to be performed in this case in order to obtain statistically relevant results. Just imagine that one call may show perfect quality, while another is significantly distorted by RFI. If only one measurement is performed it is impossible to say whether any impairments are caused by the mobile phone, or by the mobile network itself. For e.g. production testing etc. it is frequently a better solution to use special BTS simulators and connect them to the mobile phone using coax cables. This will widely reduce the risk of unwanted distortions in the measurement and lead to very reliable and reproducible results. Including the acoustical path in turn requires a sound lab or at least a controlled acoustical environment. Otherwise the impairments reported by OPERA will be more due to the ambient noise than due to the device under test.

Please note, that some HATS contain filters that model the outer ear. Depending on the measurement algorithm used for the evaluation, this may lead to wrong results, if the psychoacoustic measure applies such a filter as well. The only measure implemented in OPERA today that takes into account the outer ear filtering is PEAQ (ITU-R BS.1387). This means if you use a HATS with filters and if you want to apply PEAQ, you need some means to switch off the HATS filtering or at least to compensate for it. We would not recommend switching off the filtering in the measurement algorithm, since it is part of the ITU standard.

Extensions of the measurement algorithm to better cope with acoustical measurements are currently under study and further information on such applications may be available soon.

2.7 Testlab

Figure 2.15 shows a scenario where several developers are sharing a pool of OPERA systems that is located in the test lab. The developers can fully remote control the data acquisition from their office and work in a comfortable environment, while the OPERA systems are placed in the noisy test lab. The evaluation of the data may be performed on the OPERA systems themselves, using the OPERA Control Center, or on the local office PC running the OPERA Software Suite. The latter solution allows for a more detailed graphical analysis of the results on the developer's PC.

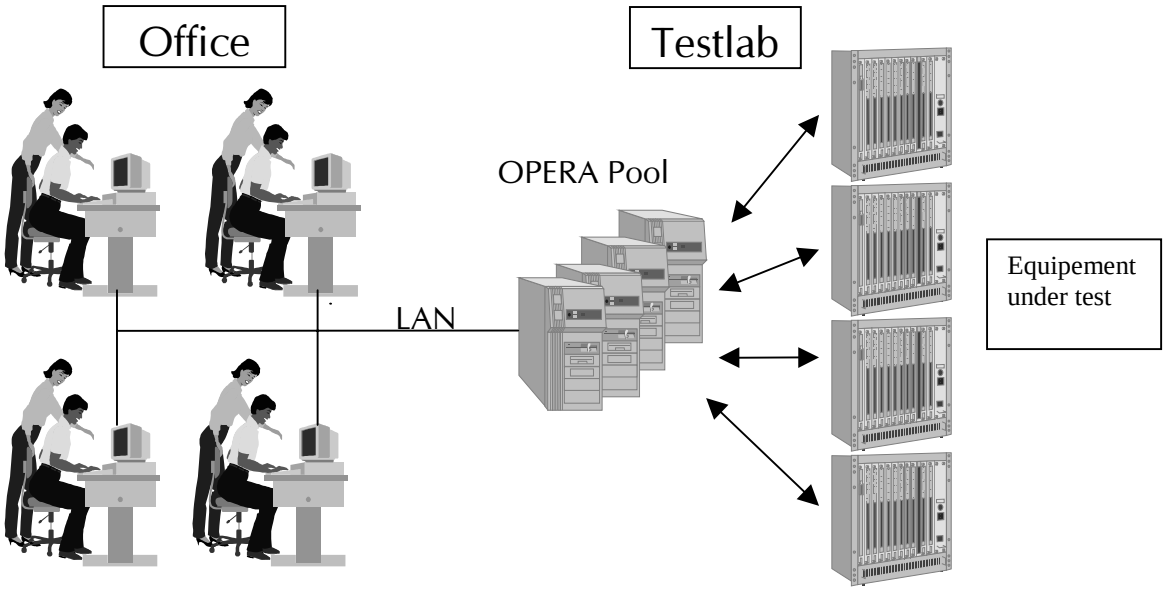

Figure 2.15: A pool of OPERA systems in the test lab is shared by a group of developers controlling the systems from their office.

2.7.1 Prerequisites

One or more OPERA base units

Each OPERA base unit is equipped with e.g. 2 four port POTS interface options

One OPERA Software Suite for each office PC

2.7.2 Description

The details of this setup depend on the customers individual requirements.

3 SUMMARY

These application notes outlined the major applications of the OPERA analyzer only, especially those related to VoIP and PSTN speech quality measurements. Due to the flexibility of OPERA, numerous other setups can be deployed in R&D as well as operational departments. We will continue to add scenarios to this application guide.

If you have any questions related to one of the setups mentioned here or on OPERA in general, please don't hesitate to contact our support department. Also, if you are looking for a customized setup which you don't see realized yet using the standard OPERA system, please feel free to contact us. We will be happy to offer you our services and customized solutions.