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COMPUTER FACILITIES

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The main computer facilities at the Institute of Phonetic Sciences are in the middle of a transition between operating systems. In 1992 we, as did most of the Phonetic Institutes in the Netherlands, made the decision to base our future audio signal processing facilities on Unix platforms. Until that time all Phonetic Institutes had standardized their facilities on Digital Equipment Corporation's VAXes with the VMS operating system. However, the price and performance of Unix workstations compared significantly favourable with VMS. Especially the appearance of Silicon Graphics' Iris Indigo workstations, with fully integrated digital audio, triggered this transition. By the end of 1994 we intend to have based on Unix the main part of the speech signal analysis software and hardware.

Presently our signal processing configuration consists of a mixed VMS and Unix network as is laid out in figure 1. On the one hand we have a VMS-cluster which consists of a microVAX II with two VAXstations 3100 and graphics terminals. Audio recording and playback, based on 12 bits of precision, is centralized and user processes have to wait in queue to use these facilities. All printing and email facilities are presently handled by the VMS platform. By the time this cluster will become obsolete (at the end of 1994, after 6 years of service) these facilities will have been transported to the Unix platform. On the other hand 5 Silicon Graphics machines (2 Iris Indigos R4000 and 3 Indy's) are presently available for audio signal processing. Each machine has 48MB memory and 512MB hard disk capacity. The audio facilities of each machine include the following:

- Analog to Digital conversion with variable sampling frequencies. Available sampling frequencies are 48 kHz (the digital audio tape norm), 44.1 kHz (the CD-norm) and integer divisions of these numbers. Software for conversions between these sampling frequencies is available. The precision of the sampling can be chosen to be either 8, 16 or 24 bits. All input and output levels are under software control.
- Filtering is done by linear phase filters. These filters do not introduce phase distortions of the signal.
- Digital I/O interface is present. The recording from DATs and CDs is digital which means that no information is lost. The recordings to DAT are also digital.
- Line level inputs and outputs are available for connections to analog equipment such as cassette decks and speakers.
- All audio functions are available via an application programmers interface (API).
- Another API controls the handling of the CD and DAT.
- All audio generation is handled by a built-in digital signal processor.

An RS/6000 model 530H, equipped with 32 MB memory, 3 GB external storage and 3 X-terminals, serves as a file server for the Unix environment. This configuration was kindly provided by IBM under the terms of the ACIS programme.

A large part of the software that was used on the VMS platform has not been ported to the new platform but will be designed anew. Knowledge of object oriented software design methods and the Motif widget set enabled the creation of a very powerful, easy extendible, and user friendly, computer program for signal manipulation, analysis, editing and visualisation. This program, *praat* (Boersma, 1994), will be extended shortly with labelling facilities and synthesis options. It can be used interactively and from scripts. Besides the *praat*-program another program is present with similar analysis features. This program called SPEXlab has been developed by SPEX (1994).

Apart from the above mentioned workstations, networked PC's and Macintoshes coexist for doing stand-alone experiments and text editing. Telnet and ftp services are available on all platforms. A router enables our contact with the external world through email and internet services.

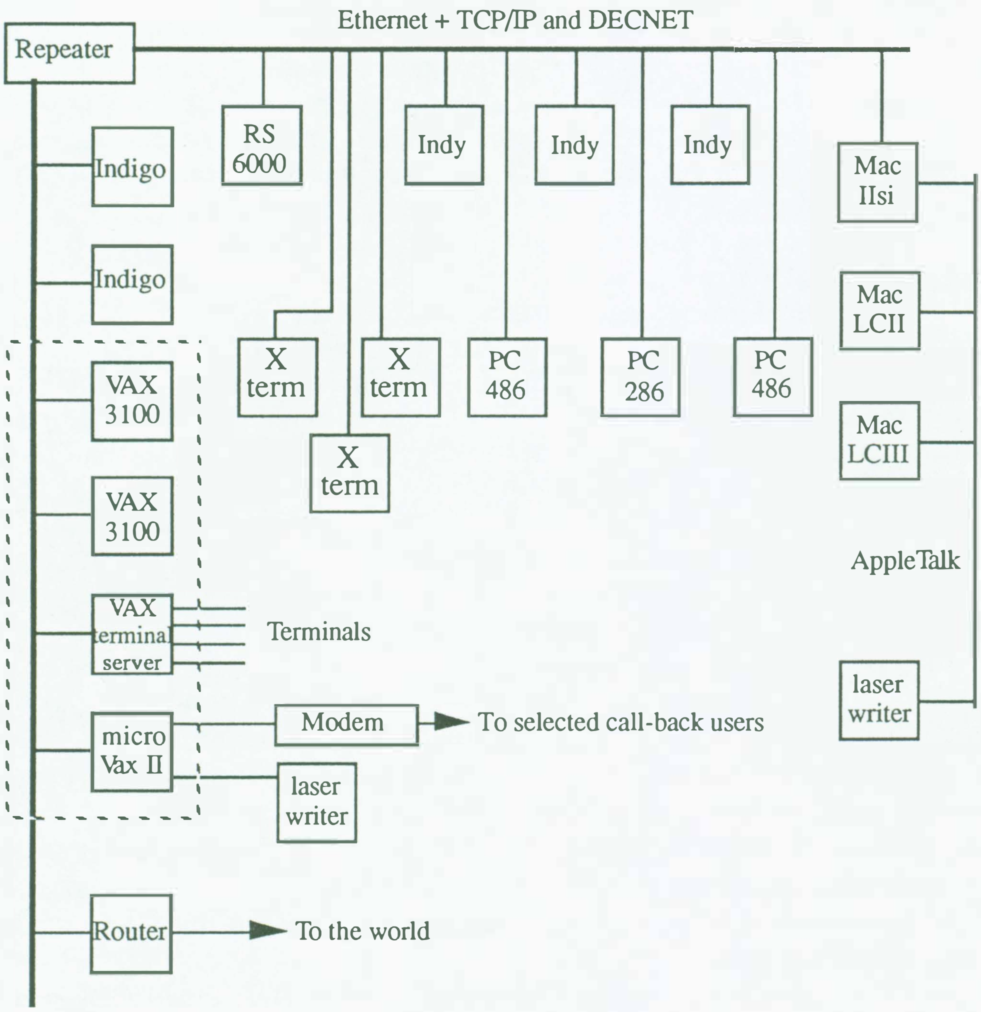

Fig. 1. Layout of the computer configuration at the Institute of Phonetic Sciences. The machines within the dashed rectangle will become obsolete by the end of 1994.

References

Boersma, P. (1994), The *praat* speech analysis and synthesis program.

SPEXlab user manual (1994).