

MULTIAPPLICATION PLATFORM BASED ON TECHNOLOGY FOR MOBILE TELEPHONE NETWORK SERVICES

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ABSTRACT

This paper describes a new platform developed at Telefónica I+D, based on Speech Technology, for being integrated in the Mobile Telephone Network, and specially suitable for quick incorporation of new customer demanded services. This multiapplication platform has been conceived as "call-basis" driven, that means, that the number called by the user will determine which application must be run at each time in order to serve him. The system permits dynamically redistribute the lines assigned to each application following traffic criteria. In the other hand, the use of dynamic libraries allow a quick procedure to update, incorporate or eliminate applications.

The integration of computers with the telephone network into commercial equipments has become feasible due to the proliferation of inexpensive Personal Computers and advances in speech processing.

The availability of the Speech Technology Products of Telefónica of Spain [1] [2] [3] [4] has made possible to develop speech based servers for new telephone services in a rapid and reasonable way.

1. INTRODUCTION

Mobile Telephony represents a dynamic, changing and competitive market which demands new services focused in customer needs, ready to be integrated in the network and suitable for changes or readaptations in a short period of time.

The use of the voice, as the most natural way of human communication, and in this case between humans and machines, joined with the great advantage of using high quality Speech Processing Technologies make the development of this platform an interesting and profitable project.

This platform is the first attempt to incorporate Speech Servers to the Spanish Mobile Telephone Networks of Telefónica: MoviLine™ (TAC) and MoviStar™ (GSM), in order to provide different voice services, with the goal of make it more attractive and feasible for the customers. The use of Speech Technology incorporates flexibility in

order to develop, incorporate or substitute new services depending on the needs, success or demand variations in a short period of time.

MoviLine™ and MoviStar™ are trademarks of Telefónica of Spain.

Speech Servers are presented as an alternative to SMSC (Short Message Service Centre) in the GSM network due to the habits of most customers of not using them, because of forgetfulness or lack of interest. At the other side, for the TAC network, which does not have the flexibility of services provided by the SMSC, Speech Servers appear as the way of develop and offer to the customer the same advanced services than in the GSM network.

The capability of attending incoming calls and generating outgoing calls simultaneously, joined to the policy of recalling, in the case of non-successfully finished calls, represents one of the most robust and flexible aspects of this system.

In chapter 2, a description of the selected hardware components is given. Chapter 3 is focused in the software architecture. Finally, chapter 4 gives a general view of some dialogue and application aspects, including an application example.

2. HARDWARE DESCRIPTION

The hardware elements of this platform are located within a PC, which holds all the boards needed for the line signalling handling, computational capabilities (for the Speech Processing Algorithms) and the communication interfaces. The speech related boards have an SCBus for data communication between them. A very simple configuration of boards permits to attend a great number of customer and to develop a great variety of services.

This structure can be divided in three parts:

2.1. Signalling Handling

Two different signalling protocols have been considered. In a first approach the platform was designed to control

an ISDN-E1 link. The chosen board for this task was the D300SC-E1 of Dialogic for calling

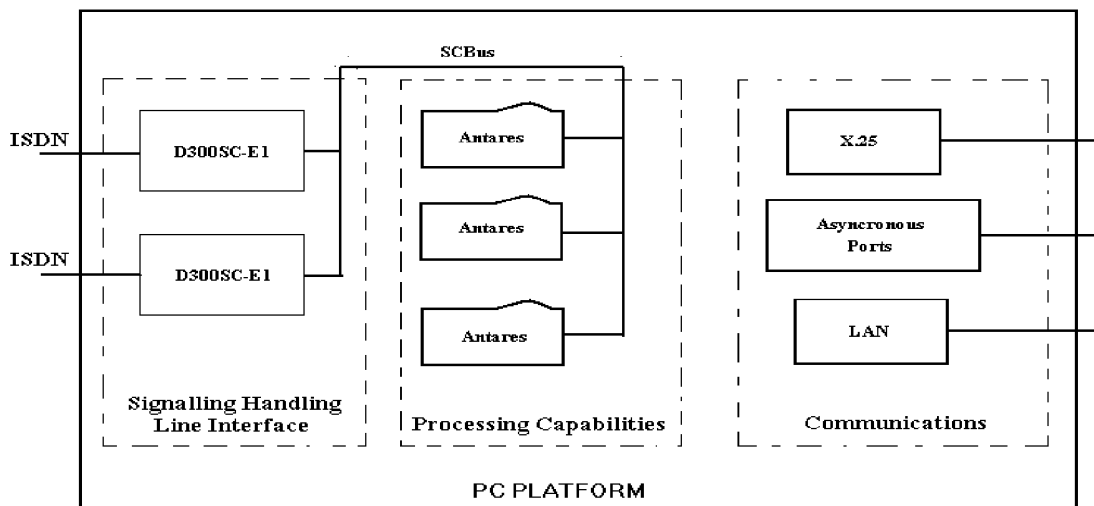


Figure 1. PC Platform Scheme of Speech Server with ISDN-E1 Connection.

and network signalling handling. Each of these boards can handle an E1-ISDN link and can provide as well some other services like "touch tones" detection and generation, and play and recording speech capabilities.

The ISDN link to the Mobile Switching Centre (MSC) permits incoming and outgoing calls.

The second attempt was driven to control ISUP signalling. For this task we chose the PCCS6 signalling card of DataKinetics.

2.2. Speech Processing

Antares Boards, with 4 DSPs each, provide the computational power for the signal processing algorithms. The algorithms licenses in the Antares boards used are those of Telefónica of Spain for Speech Recognition and for Text-to-Speech Synthesis.

2.3. Communication Handling

The essence of all the services given by voice servers are the information requested by the user. In order to retrieve all the informations that derived from the services or applications planned to work in the multiapplication platform, it was necessary to design and install several interfaces which will permit connections with databases and servers distributed in the net in order to obtain the requested informations.

The main communication interfaces incorporated in our system are the following:

- X25. A great variety of protocols can be encapsulated and transmitted via X.25.
- LAN interface for TCP-IP
- asynchronous ports for RS-232 and X.28 protocols

The general scheme of the hardware platform with ISDN Interface is shown in figure 1.

3. SOFTWARE ARCHITECTURE

The system is running under UNIX SCO. The key of this development has been the use of the dynamic libraries provided by this operating system.

Two classes of processes run in this platform. The first is composed by a main process, the "Process Manager" (PM), which runs all the control needed to start, stop and attend the different applications.

The second class of processes are the line processes (LP), all of them exact copies of a general "line process" capable of execute any of the applications included in the "Multiapplication Platform". Each of the LPs will handle the signalling interfaces and the whole telephone call of one telephone line.

The operation interface of the system offers three possible actions:

- to start the system: starting PM and LP with a known application configuration and a line distribution per application given.

- to stop the system: that is, the PM and all the LPs. The stop of the system is needed when the

incorporation, elimination or update of applications are desired. An immediate re-start of the system can be

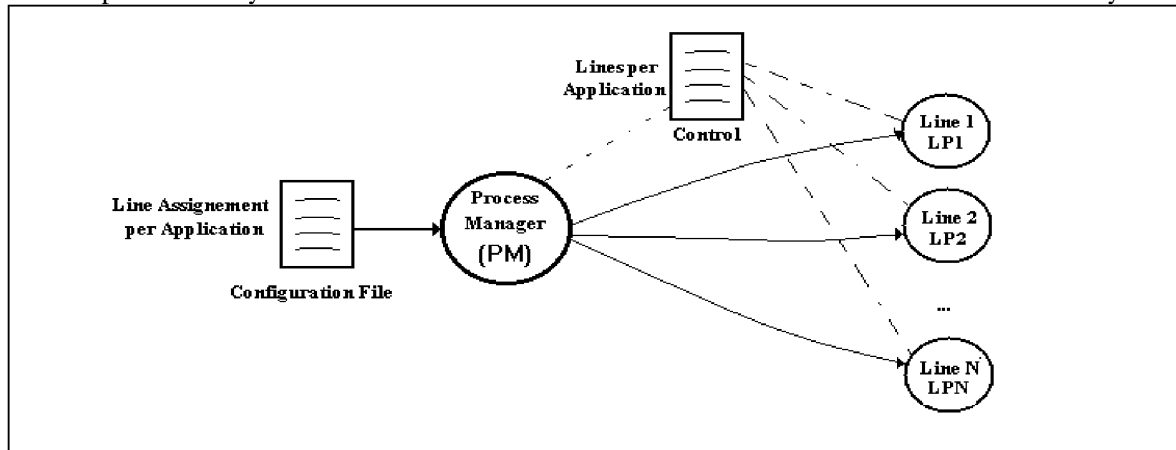


Figure 2. Software Architecture.

done in order to continue the system operation with the new configuration, thanks to the dynamic libraries.

- to introduce changes in the configuration of the line distribution per application. This action does not force the stop of the system due, to the flexibility provided by the dynamic libraries.

Any of these operations can be done dynamically, at any moment, and only by accessing to one Configuration File which contains the "Line Assignment per Application".

This configuration file can be managed by any kind of editor, from the simplest, the "vi" editor of UNIX, to CGI or Java editors.

Figure 2 shows a scheme of the process architecture. When starting the system, and during normal operation time, the "Process Manager" consults periodically the "Line Assignment Configuration File" in order to update and control the number of lines attending each application, or in its case, to stop the system.

This control is done by using "Interprocess Communication" tools provided by the UNIX SCO operating system. The PM creates a memory image containing the line configuration per application specified and the number of LPs executing each application at any time.

This memory image is consulted by both, LPs and MP. The latter will update this information any time the operator introduces a change in the configuration.

4. APPLICATIONS AND DIALOGUES

The present platform has been conceived as a "Call-Basis" multiapplication platform. The goal is to have a Voice Server which permits simultaneously to provide several services depending on customer demand levels, and optimizing the resources used.

For an incoming call two data are expected to be retrieved:

- the "called number" or DNIS (Dialled Number Identification Service), which is used to select the application that must be run in the LP to attend the call.
- the "calling number" or ANI (Automatic Number Identification) for customer identification.

The latter, joined with the DNIS, starts the request information processes needed for the application, to which the call belongs. These information requests would be directed to other servers distributed in the net and connected through any of the communication interfaces available.

These parameters can be obtained in an automatic way, that is, ANI and DNIS can be obtained thanks to the ISDN or ISUP links, who make it possible in both the Moviline™ and Movistar™ networks. For systems in which this kind of interfaces are not available, a first stage of dialogue must be done, in order to capture and verify the identification related information of the customer calling and the service requested.

Once the demanded information is received the "voice server" starts the dialogue with the user, in which the information is given and new options are presented.

The Speech Technology Resources used in this dialogue can be any combination of the following:

- Automatic Speech Recognition (ASR), including "Natural Numbers Recognition" [1] [2] and "Flexible Vocabulary Recognition"[3]. The latter permits the application developer to create and adapt the vocabulary to his needs. Nowadays, there are automatic speech recognition resources from Telefónica of Spain in several languages of the Spanish state, like the Castilian Spanish, the Catalanian and the Galician.

- Text-to-Speech Synthesis (TTS) [4] from Telefónica of Spain also in several languages: Castilian Spanish, Catalanian and Galician.

- TTS and ASR in different languages, like British English or French, from several providers.

- Playing and Recording capabilities, which permit the inclusion in the platform of messaging services.

- Fax transmission and reception, which means a complementary extension of the services for long size information requests.

- DTMF detection and generation.

All applications have a common line interface handling program for the ISDN or ISUP interface. Both incoming and outgoing calls are attended simultaneously.

For incoming calls, depending on the "called number" an application is chosen in order to run the service demanded by the customer.

For the outgoing calls an exhaustive and information expiry dependent policy of re-calling is followed.

Any platform installed can hold and provide any of the services developed over it.

The criteria followed to group and select which services should be given in any platform have to be chosen following market and exploitation aspects. The main criteria are those related with traffic distribution during the day and more demanded services, in order to obtain an equal distribution of use of the speech and line interface resources.

4.1. Application Example

As an example of application we can consider a "Prepaid Card Information Service" (PCIS) for Mobile Telephony. This service have a complex application involving incoming and outgoing calls. In this example we consider the case of handling an ISDN interface.

A brief scheme of the dialogues and behaviour of the application in both cases is as follows:

4.1.1. Incoming Calls

In this case, the initiative comes from the customer. The steps followed are:

1. The "Set-Up" call information is received.
2. Extract DNIS and ANI from the "Set-Up".
DNIS: determines that the PCIS application must be run.
ANI: credit information request for the ANI user.
3. Credit information request through any of the communication interfaces available for the credit

information server.

4. a. Successful reception of the credit.
The information is read to the customer using Text-to-Speech synthesis.
4. b. No credit information is received:
 - Finish the call
 - Include customer in re-calling policy.

4.1.2. Outgoing Calls

The initiative of the service to be started comes from a external server, connected with our system, who sends an information message. In the PCIS case, an important variation of the customer credit, or a limit value of it can provoke this initiative of informing the customer.

1. Message received from information server.
2. The system extracts from the message what application must be run.
3. The system starts a call via ISDN.
4. a. Successful call.
 - The information of the credit is read using Text-to-Speech synthesis.
4. b. Unsuccessful call (called number busy or nobody answers).
 - Include customer in re-calling policy.

CONCLUSIONS

We have described the hardware and software architecture of a Multiapplication Platform, conceived as "call-basis" driven. The flexibility of the system permits dynamically redistribute the lines assigned to each application and a quick procedure to update, incorporate or eliminate applications. The use of speech related technologies provide a customer friendly and flexible architecture to develop market adapted services.

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