CAPABILITIES OF OPEN SOURCE SOFTWARE FOR DEVELOPMENT IVR APPLICATIONS

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ABSTRACT

In recent times, the purpose of telecommunication exchanges was to interconnect each call. At present, the job has changed and main task is to handle it. Modern telecommunication exchanges have many possibilities to accomplish the task successfully. Main part of the work is secured by Interactive Voice Applications, IVR. This paper is focused to analyze possibilities of open source software in this part of telecommunication services.

1. PRESENT STATE

Classic access to PBX problematic concentrates all services in PBX (Private Branch Exchange). The telecommunication terminals couldn't provide any services, they used only the PBX. In present, many services are provided by end user terminals. This trend was driven mainly with internet telephony, voice over IP. Voice over IP exchanges are not capable only to switch calls, but they are also designed to handle the calls, even the callee is not accepting the call. For this reason, the voicemail applications are used for example. If the callee does not answer the call, caller is appealed to leave a message.

The next logical step is to ensure, before the call is connected, that it is really necessary for caller to deal with caller. When not, the call can be handled by PBX itself, with no need to contact live person. There is the main sphere of competence of Interactive Voice Response type of applications.

2. IVR APPLICATION

Interactive Voice Response applications are designed to handle the call without direct connecting caller with live person. High number of calls is unneedful or the callers are demanding same information. When are those calls handled by IVR system instead of called organization employee, it often means major reducing of expenses for the called party. Because of this, we are encountered with IVR of applications with increasing frequency.

2.1. PURPOSE OF WORK

The main purpose of work is to analyze open source solutions for IVR applications and to prove, that this type of applications can be successfully used in real world.

2.2. IP PBX ASTERISK

The most present used freeware IP telephony solution is Asterisk® - the Open Source PBX! Asterisk project is presented as complete, software PBX, supporting wide range of Voice over IP platforms including SIP. It is available for Linux, BSD and Mac OS X operating systems. [1] [2]

Basic VoIP Asterisk PBX can be used on common Linux, BSD or MAC OS X platform without any additional hardware. However, the hardware can be used for install analog or ISDN gateways, GSM gateways and so on [3]. Asterisk PBX was successfully installed and tested on two servers with working trunk interconnection.

Open source private branch exchange Asterisk is also capable in developing IVR applications. This can be assured by dialplan configuration function file, named extensions.conf. This file is programmed with Asterisk-specific programming language.

2.3. PROPOSED IVR APPLICATION

The proposed IVR application should reflect typical usage in small and medium organization. Its scheme is presented on figure 1. The scheme was, in means of lucidity, simplified. For example, the first timeout, executed from main menu, waits two times for the basic information message is played and than it connects the caller with operator. This approach is necessary for calls performed by old analog phones with pulse dialing. There is no need for this type of timeout at other levels of IVR application, because the fact, that caller is able to access it proves, that his phone is capable to use tone (DTMF) dialing.

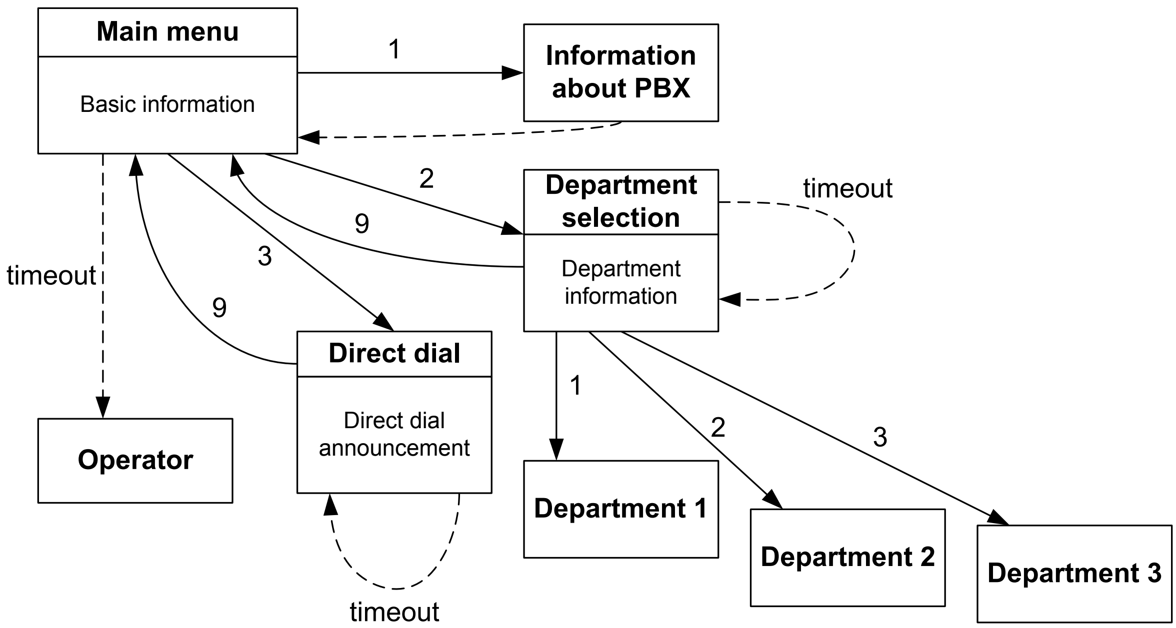

Figure 1: IVR application scheme

2.4. CONFIGURATION, TEST RESULTS

Whole configuration is made in *extensions.conf* dialplan configuration file. For recording voice messages, the script shown in Figure 2 should be used.

```
1: exten => 0,1,Answer()
2: exten => 0,2,Wait(1)
3: exten => 0,3,Record(filename:wav)
4: exten => 0,4,Playback(filename)
5: exten => 0,5,Hangup()
```

Figure 2: IVR application scheme

The sound file is recorded by pressing 0 key after beep signal. For end of recording, the # has to be pressed and sound is played back to user. If the result is satisfying, the recording should end. In other case, user has to repeat the action until is satisfied.

The tests proved that the Asterisk PBX software is capable for designing IVR applications. The IVR system worked without problems and the configuration scripts are powerful enough for developing any type of structure and can support this type of applications.

3. ADAPTATION ON SMALL, EMBEDED DEVICE

After successful tests, open source PBX was adapted to small, embedded platform. Toradex Colibri PXA290 was used. The previous experience was used during installation [2]. The next logical step after successful adaptation is to take over the load tests.

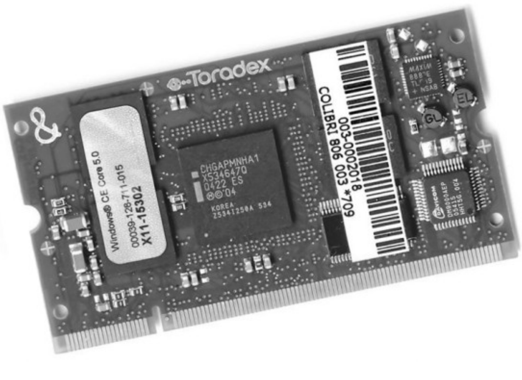

Figure 3: Toradex Colibri PXA290

3.1. TEST SETUP

SIPP open source software in the client mode was used for load generation. Client software was connecting to Asterisk PBX and “listened” for voice menu messages. Number of client instances was explicitly set to 0, 9, 18, 27 and “full saturation”. The test call was generated by standard IP phone and received by softphone installed on software terminal. Call quality was measured by Surveyor network analyzer on same terminal. The test setup scheme is shown on figure 4.

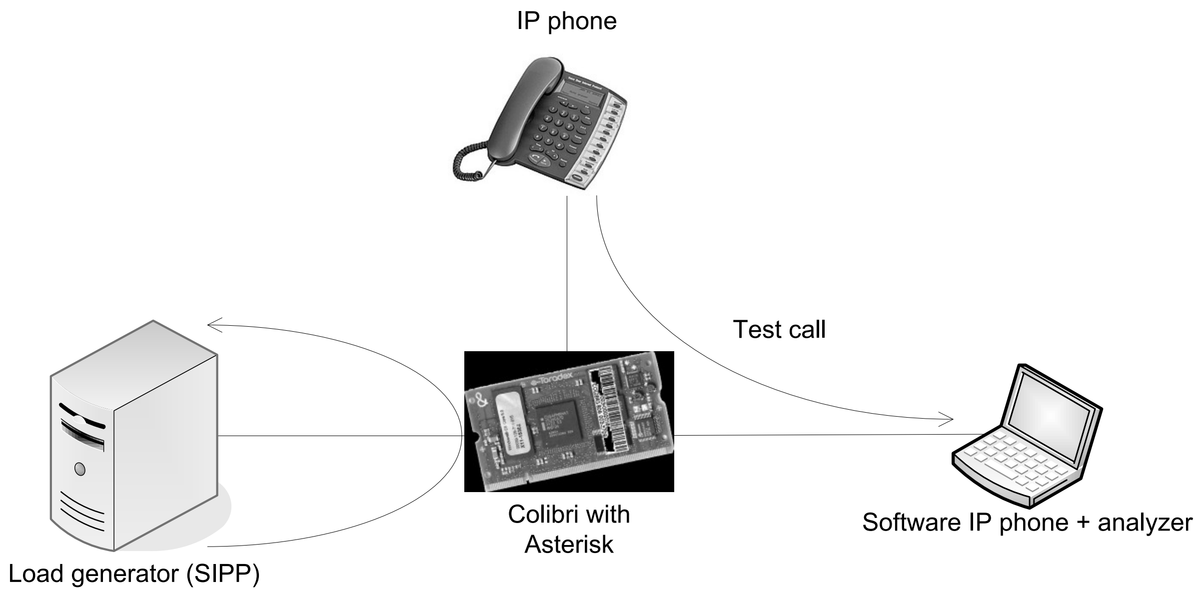

Figure 4: Test system scheme

3.2. TEST RESULTS

10 test iterations for each number of calls were passed. Mean results are shown in Figure 4. Network R Factor and MOS factors are whithin the “Excellent” and “Good” levels [4] at all times, even in saturation mode. Those results are unexpected for full saturation, but the reason is quite simple. When the PBX is fully loaded, it prioritizes actual calls before the new ones, which are rejected or cancelled on client side in terms of timeout. The reason of lower jitter and normalized delay levels in saturation state is not as obvious and further research will be focused in this direction. Results are shown on figures 5 and 6.

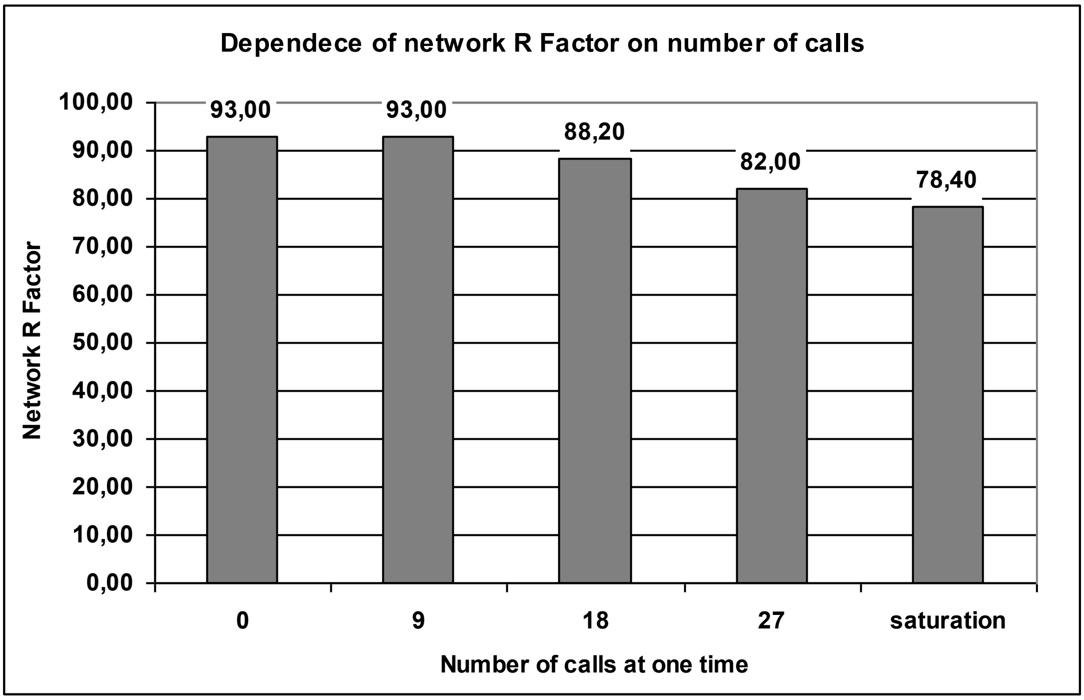

Figure 5: Network R Factor

Number of calls	0	9	18	27	saturation
Network R Factor	93,00	93,00	88,20	82,00	78,40
MOS -LQ	4,20	4,20	4,07	3,91	3,75
MOS -PQ	4,27	4,27	4,16	4,01	3,89
Load [kB/s]	0,00	59,60	119,00	180,00	200,00
Normalized delay [ms]	0,47	1,05	24,12	72,25	47,11
Jitter [ms]	0,20	0,54	2,25	8,795	7,926
Packet loss	0	0	59,00	90,00	205,00

Figure 6: Test results

4. CONCLUSION

At present, the IVR applications are used more often than before. For IVR applications, commercial products were often used. Our tests with Asterisk PBX proved, that is possible to develop IVR application using free open source software only which can compete with commercial products.

It is expected that in future, the IVR applications will be used more and more often with more and more complex logic. The advanced techniques, as voice recognition or voice synthetization will be used.

Asterisk PBX can be used in embedded machines with lower amount of memory and CPU power and it is able to ensure good quality for calls in progress even in saturation mode, in which the new ones are rejected or canceled by caller in terms of timeout.

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