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A comprehensive IVR (Interactive Voice Response) analysis model using online analytical processing (OLAP) on a multidimensional data cube

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Abstract

Private Branch eXchange (PBX) is a tool indispensable in the business world. The telephone exchanges allow employees to perform internal connections between telephones, or make calls to the external network also known as Public Switched Telephone Network (PSTN). With increasing Internet usage, there is interest in understanding what services are offered. Enterprise Courier is a commercial Internet Protocol Private Branch eXchange (IP PBX) based on open source Asterisk web-based PBX software for Linux, which supports multiple protocols and services, like Interactive Voice Response (IVR). Cisco Unified Communications Manager (CUCM) or CallManager, is a software based call-processing system (IP PBX) developed by Cisco Systems. CUCM tracks all active Voice over IP (VoIP) network components; including phones, gateways, conference bridges, among others. IVR is part of the Academic Services costumer contact and ticketing of University of Beira Interior (UBI). IVR monitoring and analysis are essential for effective operation and resource management, in particular, multidimensional analysis for long-term data is necessary for comprehensive understanding of the trend, the quality of customer service and costumer experience. In this paper, we propose a new IVR analysis model for large volumes of IVR data accumulated over a long period of time. The IVRCube proposed is an analysis model using online analytical processing (OLAP) on a multidimensional data cube that provides an easy and fast way to construct a multidimensional IVR analysis system for comprehensive and detailed evaluation of long-term data. The feasibility and applicability are validated, as the proposed IVRCube analysis model is implemented and applied to Academic Services costumer contact and ticketing IVR data.

Resumo

A *Private Branch eXchange* (PBX) é uma ferramenta indispensável no mundo dos negócios. As centrais telefónicas permitem que os funcionários realizem chamadas internas entre telefones, ou façam chamadas para a rede externa, também conhecida como *Public Switched Telephone Network* (PSTN). Com o aumento sistemático da utilização da Internet, há um interesse acrescido em entender quais os serviços que são oferecidos nas redes baseadas em *Internet Protocol* (IP). Um destes serviços é o *Voice over IP* (VoIP).

O *Enterprise Courier* é um *software* IP PBX comercial para VoIP baseado na aplicação de código aberto Asterisk, que opera sobre Linux. O IP PBX *Enterprise Courier* suporta vários protocolos e serviços, por exemplo o *Interactive Voice Response* (IVR). O *Cisco Unified Communications Manager* (CUCM) também chamado de *CallManager*, é um sistema de processamento de chamadas IP, ou IP PBX, desenvolvido pela Cisco Systems. O CUCM permite fazer a gestão e operação de todos os componentes ativos de voz, incluindo telefones, gateways, equipamentos de conferência entre outros.

Estes sistemas coexistem na rede de gestão de comunicações de voz da Universidade da Beira Interior (UBI), sendo que o sistema automatizado utilizado para o encaminhamento de chamadas dos Serviços Académicos na UBI utiliza a tecnologia IVR. Este serviço da UBI é uma das formas que os clientes da Universidade (alunos e não alunos) têm para obter informações e resolver questões de forma rápida e simples usando o telefone.

Por ser um importante ponto de interface entre a universidade e a comunidade, a monitorização e análise de desempenho do IVR são essenciais para o funcionamento eficaz e gestão de recursos humanos atribuídos a este serviço, o que torna a tarefa de extrair os dados do sistema de VoIP e apresentá-los de forma a poder extrair deles informação útil à gestão, o centro deste trabalho de investigação. Para a análise dos dados, foi usada uma técnica de análise multidimensional de dados a longo prazo, necessária para uma compreensão abrangente da evolução e qualidade de serviço prestada ao cliente tendo como objetivo a melhor experiência possível por parte do cliente.

Neste trabalho, propomos um novo modelo de análise de IVR para grandes volumes de dados acumulados ao longo de um extenso período de tempo. O IVRCube é um modelo de análise utilizando *online analytical processing* (OLAP) num cubo de dados multidimensional que fornece uma forma fácil e rápida de construir um sistema de análise multidimensional para avaliação exaustiva e pormenorizada dos dados ao longo do tempo. A viabilidade e aplicabilidade deste modelo são validadas, uma vez que o modelo de análise IVRCube proposto é implementado e aplicado ao serviço de contacto telefónico (IVR) dos Serviços Académicos da UBI.

Keywords

Call Center, Contact Center, Customer Contact Transformation, Customer Self-Service, Customer Service, IVR, IVR Design, Customer Care, Business Intelligence, OLAP, Multidimensional information systems, Data Warehousing, Decision Support Systems, Business Analytics, Knowledge Management

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Acronyms

AM	Amplitude Modulation
CDR	Call Detail Record
Codec	Coder-decoder
CUCM	Cisco Unified Communications Manager
CUCS	Cisco Unified Computing System
DDI	Direct Dial-In
DID	Direct Inward Dialing
FCCN	Fundação para a Computação Científica Nacional
FM	Frequency Modulation
FR	Functional Requirements
HCS	Cisco Hosted Collaboration Solution
IP	Internet Protocol
IP PBX	Internet Protocol Private Branch eXchange
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
IVR	Interactive Voice Response
MOS	Mean opinion Score
ms	Milliseconds
NFR	Non-Functional Requirements
NREN	National Research and Education Network
OLAP	Online Analytical Processing
OSI	Open Systems Interconnection
PBX	Private Branch Exchange
POP	Point of Presence
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
SBC	Session Border Controller
TCP	Transmission Control Protocol
TV	Television
UBI	University of Beira Interior

UC	Unified Communications
UCaaS	Unified Communications as a Service
UDP	User Datagram Protocol
UHF	Ultra High Frequency
VHF	Very High Frequency
VoIP	Voice over IP

Extended Abstract in Portuguese

Introdução

Esta secção apresenta, em português, o resumo alargado da dissertação de mestrado intitulada "*A comprehensive IVR (Interactive Voice Response) analysis model using online analytical processing (OLAP) on a multidimensional data cube*" ("Modelo de análise abrangente de sistemas de resposta de voz interativa usando processamento analítico em tempo real num sistema de cubo de dados multidimensional").

O seu propósito é o de enunciar o problema a ser tratado, os principais objetivos e as suas contribuições para o avanço da ciência.

Este resumo alargado tem a seguinte organização: em primeiro lugar, são apresentados uma pequena introdução tecnológica, o problema e objetivo desta dissertação. Seguidamente a solução abordada e na secção resultados são apresentados alguns resultados relevantes. A finalizar são descritas as conclusões deste trabalho e propostas linhas de investigação para trabalho futuro.

Introdução Tecnológica e objetivos

Durante o ano de 2005 a Universidade da Beira Interior (UBI) iniciou o seu processo de convergência da rede telefónica e da rede de dados, após ter adquirido uma solução *Internet Protocol Private Branch Exchange* (IP PBX) da *Cisco Systems*. Esta solução foi interligada com o PBX existente e através de um *media gateway* foi também interligado com a rede *Public switched* *telephone network* (PSTN). Após dois anos, em 2007, a UBI integrou um projeto à escala nacional denominado VOIP@RCTS, que foi liderado pela Fundação para a Computação Científica Nacional (FCCN). Este projeto consistiu em interligar todos os PBX existentes através de *media gateways* e encaminhar o tráfego sempre que possível por IP. O *software* escolhido é denominado de *Enterprise Courier* e é baseado na plataforma *Asterisk*.

Com o novo IP PBX *Enterprise Courier* novas funcionalidades tornaram-se disponíveis, nomeadamente a possibilidade de utilizar *Interactive Voice Response* (IVR). Esta funcionalidade permite que as chamadas sejam atendidas automaticamente pelo sistema e após a seleção de uma opção do menu, a chamada seja encaminhada para um agente ou operador que vai atender o assunto em causa. Esta seleção permite que o agente que atenda a chamada seja o mais capacitado para o assunto em causa.

Problema e motivação

A UBI procura melhorar de forma continuada a sua estrutura organizacional, os seus processos e seus métodos de controlo, com o objetivo de satisfazer e antecipar as necessidades dos seus clientes e outras partes interessadas.

O novo serviço IVR do IP PBX *Enterprise Courier* não tem nenhuma forma de análise estatística dos dados que gera e portanto não existem métricas que permitam a UBI avaliar e planear melhorias na área do atendimento telefónico automático.

Em 2013, o Sr. Vice-Reitor responsável pelos Serviços Académicos requereu que fosse implementada monitorização neste sistema para uma correta gestão e supervisão do mesmo. Tendo isto em consideração, este estudo descreve a investigação levada a cabo que permitirá no futuro ter estatísticas do serviço IVR e assim contribuir para a tomada de decisões de gestão melhor informadas.

IVRCube: um modelo de análise IVR usando online analytical processing (OLAP) num cubo de dados multidimensional

Baseado num conjunto de entrevistas efetuadas, foi criado um conjunto de requisitos funcionais e não funcionais. Estes requisitos foram fundamentais para o desenvolvimento deste trabalho, pois indicam sob um aspeto geral as funcionalidades mais relevantes do sistema.

O sistema *Enterprise Courier* guarda os registos *Call Detail Record* (CDR) de todas as chamadas, incluindo as chamadas que entram no sistema IVR, em base de dados *MySQL*. A primeira fase do trabalho foi identificar quais os dados relevantes constantes dos CDR e importar esses dados para o *data warehouse* institucional. Com esses dados foi criada uma estrutura multidimensional, normalmente denominada de cubo ou hipercubo, que contém medidas ou factos e inclui dimensões ou perspetivas. O hipercubo construído tem seis dimensões e duas medidas.

Resultados e discussão

Os decisores normalmente pretendem ter acesso à informação de forma simples, generalizada e rápida. Nesta seção são apresentados vários resultados em forma de gráfico e tentou-se de uma forma geral que esses mesmos resultados fossem expostos a partir das várias perspetivas do hipercubo construído.

A partir da informação recolhida foi possível identificar imediatamente duas situações no IVR que necessitam ser modificadas:

1. A ordem do menu não está de acordo com o que vários estudos indicam, ou seja, no menu a opção mais solicitada deve ser colocada em primeiro lugar, a segunda opção mais solicitada em segundo lugar e assim sucessivamente.

2. O período matinal de atendimento automático termina às 13h, isto permite que chamadas que entrem muito próximo do fecho sejam atendidas durante o intervalo de almoço dos funcionários deste serviço. Tendo em conta tempo médio que um cliente está disposto a esperar para ser atendido e o tempo médio mais longo de conversação das várias opções IVR, identificou-se que o sistema deve deixar de aceitar novas chamadas antes das 12 horas 52 minutos e 25 segundos.

Conclusões e trabalho futuro

Os estudos indicam que grande maioria das pessoas ainda prefere utilizar o telefone em detrimento das novas tecnologias. Os sistemas de atendimento telefónico IVR são desta forma um dos principais e normalmente o primeiro ponto de contato e pretende-se que a experiência do utilizador seja a melhor possível.

Este estudo permitiu a resposta a várias questões normalmente colocadas pelos decisores e poderá ser usado para ajustar a mão-de-obra disponível de acordo com os meses de maior afluxo de chamadas, as horas a que estas normalmente ocorrem, materializar estímulos positivos aos agentes que interagem com atendimento automático e de uma forma recursiva melhorar o desempenho global.

Trabalho futuro

Como trabalho a curto prazo será a definição de um *dashboard* para que o Sr. Vice-Reitor possa ter acesso a informação de gestão de uma forma rápida e efetiva. Esta informação poderá então ser usada para fazer decisões baseadas no conhecimento adquirido.

Existe ainda a oportunidade de utilizar esta tecnologia e esta abordagem em vários sistemas informáticos, por exemplo, ligados às redes de computadores, que tipicamente geram um grande volume de dados e que raramente são analisados, não contribuindo assim para a extração de novo conhecimento.

Chapter 1

Introduction

1.1. Technological background

University of Beira Interior (UBI) [1] started the convergence of the phone system and the data network in 2005, after acquiring a commercial Internet Protocol (IP) Private Branch Exchange (PBX) software from Cisco called CallManager [2]. The long term objective to UBI, was to have 100% Voice over IP (VoIP) network until 2020. The Internet Protocol Private Branch eXchange (IP PBX) was integrated with the analog PBX's and with the Public switched telephone network (PSTN). Figure 1.1 presents a scheme depicting CallManager integration in UBI voice architecture. In Figure 1.1, E-carrier (E1) represents a Primary Rate Interface (PRI). PRI is a standardized telecommunications service level within the Integrated Services Digital Network (ISDN). E1 provides 30 communication channels and 2 signaling channels, this is also referred to as 30B channels plus 2D channels. In this case Media Gateway has 60 voice channels to interconnect VoIP and legacy telephony.

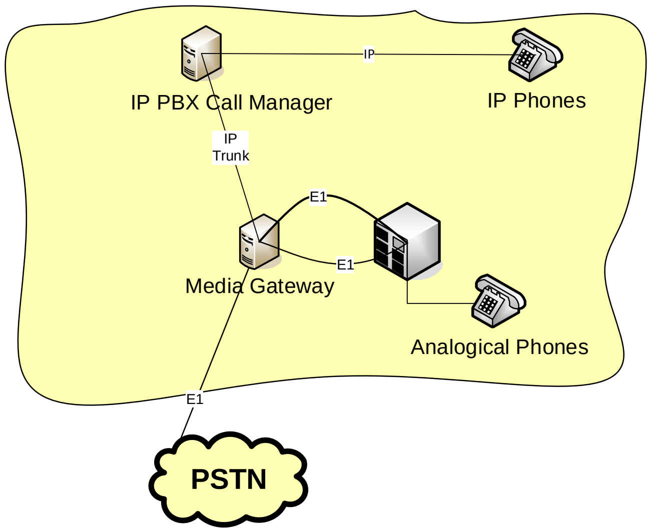


Figure 1.1 - Scheme depicting CallManager Integration.

In 2007, Portuguese Universities started changing their phone systems into a converged system using the data network to route phone calls. This project, named VoIP@RCTS [3] was launched by the *Fundação para a Computação Científica Nacional (FCCN)*, the Portuguese National Research and Education Network (NREN). The chosen solution was a software platform based on FreePBX [4] which uses the Asterisk core [5]. This software, although based on FreePBX, it has been redesigned to suite Universities' specifications. This new software is called Enterprise Courier from itCenter [6]. UBI integrated that same project and the system was incorporated in its telephone network. Figure 1.2 illustrates UBI integration of VoIP@RCTS in the existing infrastructure of UBI. In Figure 1.2 we can also see the Session Border Controller (SBC) that acts as a firewall for the VoIP world.

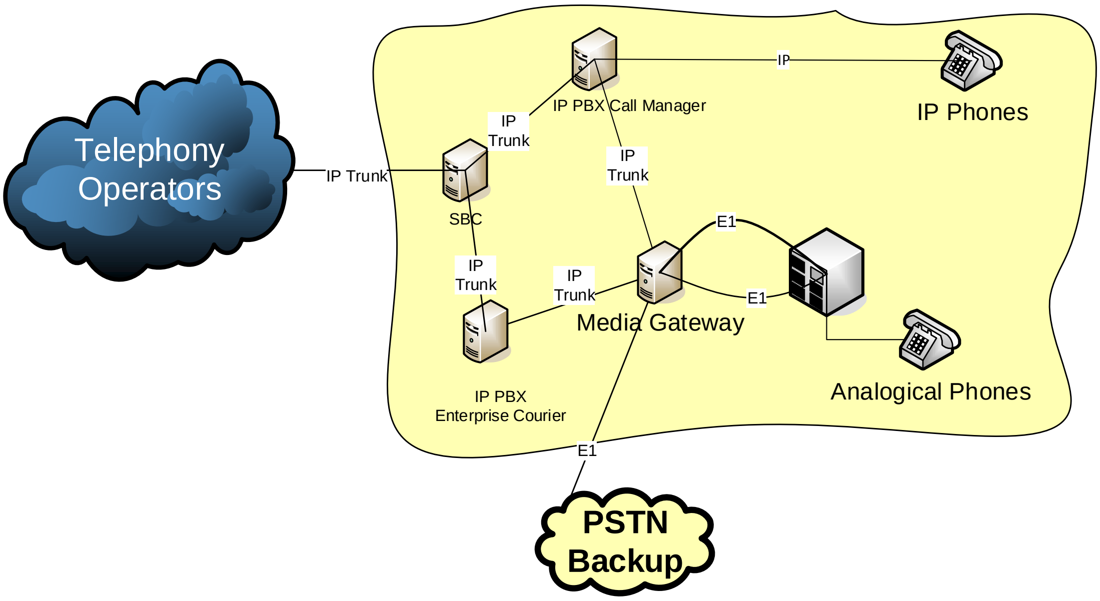


Figure 1.2 – Integration of VoIP@RCTS in UBI infrastructure.

Although the provided solution was based on open source software, it was customized for the universities specific needs permitting further use of the new features offered by Enterprise Courier. One of these possibilities is the use of Interactive Voice Response (IVR), which suited some of the specific services, allowing for example, the calling user select the most adequate operator for his/her particular call motivation.

1.2. IVR a new offered service

Interactive Voice Response play an important and increasingly role in today's business world and global economy. Indeed, they serve as the primary customer-contact channel, for companies in many different industries. IVR systems, if properly designed, can increase customer satisfaction and loyalty, cut staffing costs and increase revenue by extending business hours and market reach [7].

Poorly designed IVR systems, on the other hand, will cause the opposite effect and lead to dissatisfied customers, increased call volume and even increased agent turnover, as customers take out their frustrations on the agents. A study from Purdue University, revealed that 92% of US consumers evaluate a company based on their experience using the company's call center. More surprisingly, the study found that 63% of consumers stop using a company's products based on a negative call center experience. That number could raise to 100% for consumers with ages between 18 and 25 years old [8].

There are two options to create an IVR: deep divided design or broad design. Deep divided design is used in order to limit the number of options in every menu, but it can have many submenus. This can increase the complexity and create confusion because it might not fit the user's mental model [9]. Experimental results showed that users who used broad IVR design performed tasks faster and with greater satisfaction than users who used deep IVR design [10]. In this study it was preferred a broad design, as it is showed in Figure 1.3.

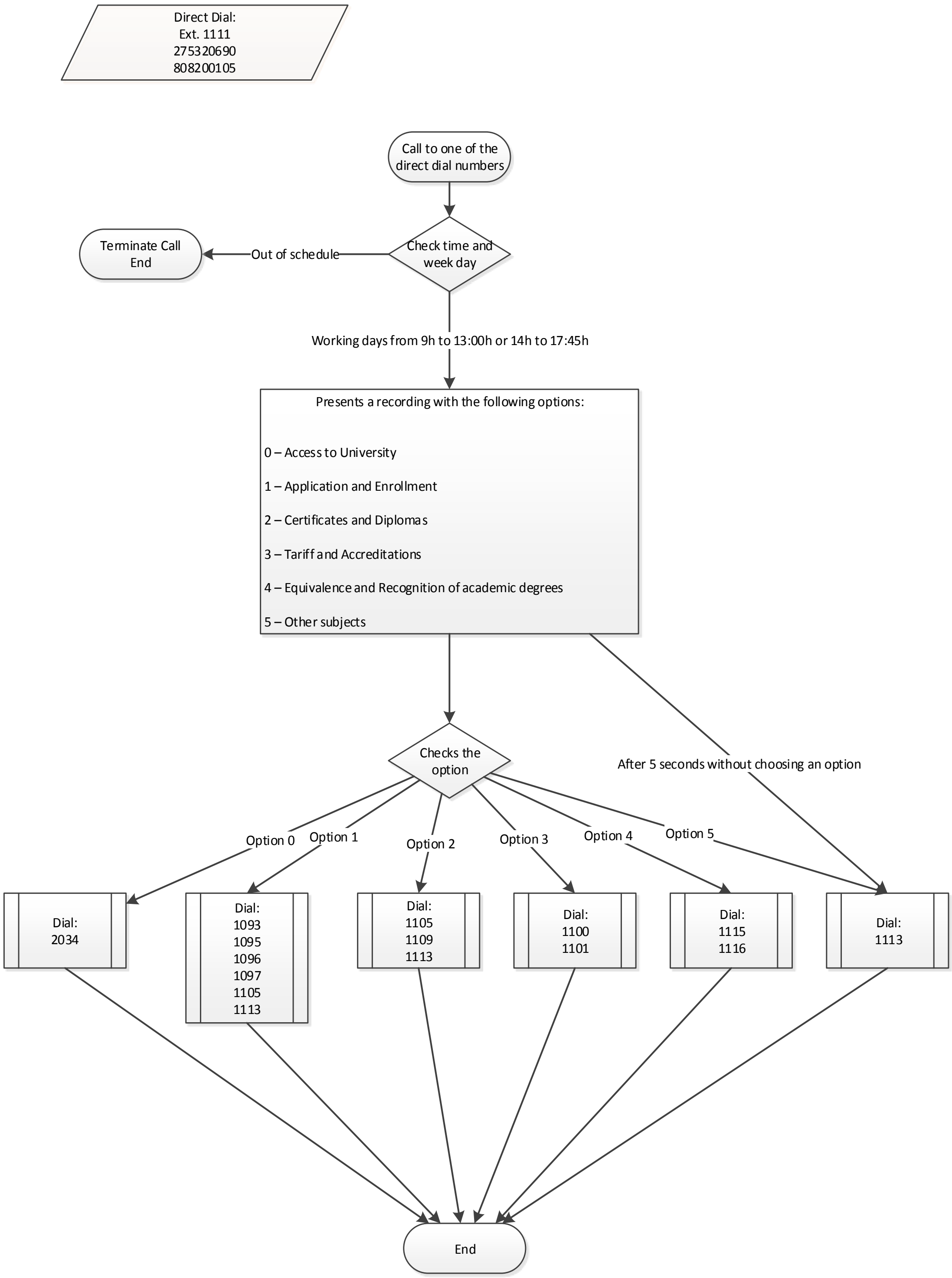


Figure 1.3 - Flowchart for the IVR Academic Services costumer contact and ticketing.

According to Figure 1.3, there are two direct inward dialing (DID), also called direct dial-in (DDI) and one internal extension to reach Academic Services costumer contact and ticketing. The date and time are only checked at the beginning.

To increase customer satisfaction, there were great concerns ordering the six menu options. According to the latest studies, the most frequent options must be placed in first place [11]. In this way, users calling the service can identify their option and don't have to wait until the recording. This ordering was done empirically without previous knowledge and will be discussed later in the results section.

There was also a great concern about the recordings. The first challenge had to do with the tone, pitch and accent. The chosen voice had to transmit a feeling of sympathy, had to be sharp and without any accent.

The second concern was related to what to present first, the textual description or the option number itself. The goal-action sequence (*e.g.*, "To do x, press y") was preferred, rather than the action-goal sequence (*e.g.*, "Press y to do x"), for options. Most researchers and standards agree with the goal-action sequence because it seems more consistent with the cognitive makeup of the task [12]. The goal-action sequence seems to encourage dial-through and was shown slightly faster response times than the action-goal sequence [13].

After the initial greeting, it's usual to prompt the user to indicate whether touch-tone service is being used. Many IVR systems instruct users to press 1 as evidence of a Dual-Tone Multi-Frequency (DTMF). Instead of asking users to press a key for evidence of DTMF, it's used the timeout value for key insertion. According to Figure 1.3, after five seconds the system will timeout and automatically choose option 5 [14].

After the option is chosen, the call is inserted in a first-come first-served (FCFS) queue. The ring strategy used is ring all, *i.e.*, the call that is leaving the queue is placed simultaneously in all phones identified for that option, *i.e.*, a call living option's 3 queue will dial extensions “1100” and “1101”. The first operator to pick up the handset gets the call.

1.3. Motivation and problem

UBI seeks to continuously improve its organizational structure, its processes and its control methods, aiming to satisfy and anticipate the requirements of its customers and other stakeholders.

Yet, the Enterprise Courier IP PBX does not provide IVR statistics, and therefore, there are no metrics that allow UBI to evaluate and plan improvements in the area of automated telephone response.

Therefore, in 2013, the Vice-Rector for Academic Affairs, recognized as the main stakeholder, required that monitoring and supervision would need to be implemented in the Academic Services costumer contact and ticketing IVR system.

Having this in consideration, this paper describes the carried out research that will allow the future creation of a solution that allows monitoring and supervision, for example, get a relationship between how many calls an option receives and which operator takes it, or how many minutes does a call last, or how many seconds is a client on hold until he starts to talk with the operator, on an Asterisk based software.

1.4. Organization of the Thesis

The remainder of this paper is organized as follows: this paragraph concludes section 1, the Introduction, where the problem, motivations and objectives of the research were outlined; section 2 follows, describing the state of the art; section 3 presents the architecture and the solution; section 4 describes and discusses the results obtained; section 5 concludes de the paper presenting conclusions and future work; and finally in the appendices section a paper with the title “A comprehensive IVR (Interactive Voice Response) analysis model using online analytical processing (OLAP) on a multidimensional data cube” fruit of this work.

Chapter 2

State of the art

2.1. Telephony Evolution

The first voice transmission was accomplished on 10 March 1876 through a ring down circuit, in which two devices were connected by wire, but there was no dialing numbers and it just worked one-way. Inventor Alexander Graham Bell called out to his assistant Thomas Watson, “Mr. Watson, come here! I want to see you.” This transmission took place in their attic laboratory. Over time, one-way voice transmission transformed to bi-directional voice transmission using wired networks. Before adopting the circuit switched technology has the basis of the telecommunication services there where human operators making the connections. Until several decades ago, the communication services were based on analog infrastructure, but the analog communication is neither robust nor efficient at recovering from the line noise (Figure 2.1). In digital communication, the line noise is less of an issue because digital amplifiers also known as repeaters not only amplify the signal, but also clean it to its original condition (Figure 2.1).

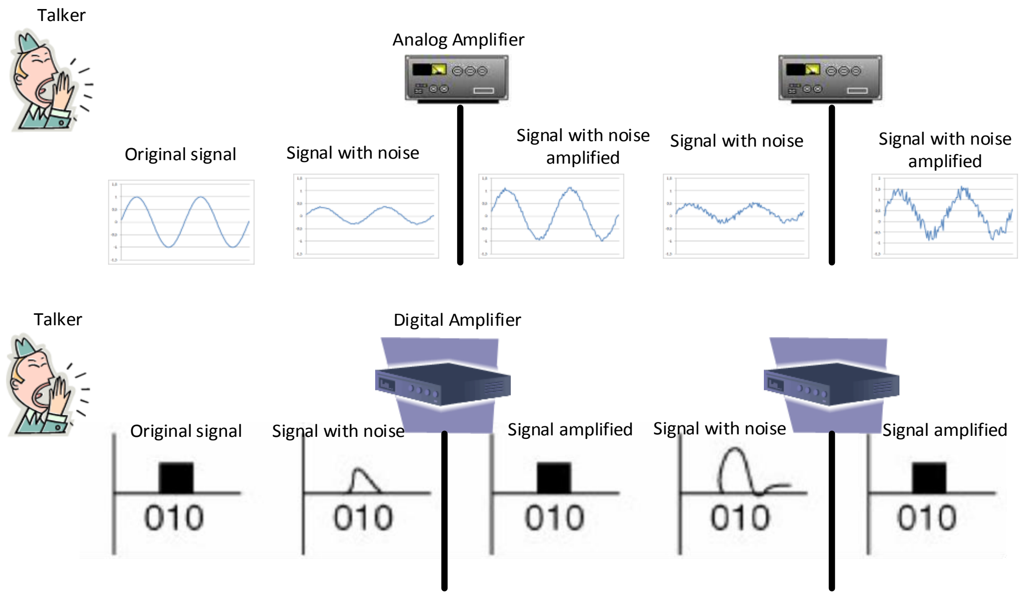


Figure 2.1 - Digital amplification vs Analog amplification.

Public switched telephone network (PSTN) system sampled the voice stream at 8 kHz and transmits the digitized voice over the circuit switched network with bit rate of 64 kb/s [15].

At the end of the 20th century, analog radio, television (TV) and telephony services stood out in particular. Most of the audio distribution in home was centered in radio, and for this purpose different modulation standards were used, as is the case of frequency modulation (FM) and amplitude modulation (AM). Regarding the TV, different signal transmission technologies were used. One example of this was the use of analog TV broadcasting on the very high frequency (VHF) and ultra-high frequency (UHF) wavebands. After that, the satellite broadcasting technology made possible the transmission of audiovisual content over very long distances, thus encouraging cultural diversity and enabling new business models to be set up. Traditionally, telephony services have been provided by telecom operators through the PSTN infrastructure.

High-quality voice services had been provided by the traditional PSTN system, since this system was built for switching voice calls over the network. The connection between the end points is established before starting the voice communications and that connection remains in use until the end of communication. This makes the bandwidth unavailable for the other services over this established circuit. The architecture built for voice is not flexible to converge data, video and voice traffic at the same time.

Interestingly, it was the PSTN that made possible many of the IP data transmissions. In the 90s it was common to have a home computer and a dialup modem connected to the nearest point of presence (POP), as showed in Figure 2.2.

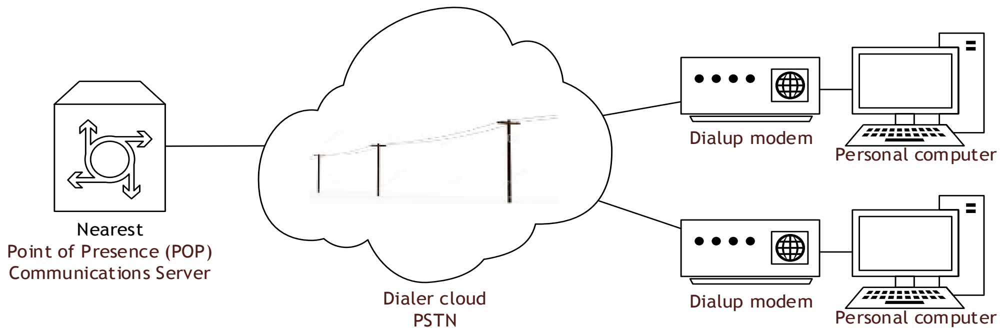


Figure 2.2 – Scheme depicting dialup connection to nearest POP.

Modern networks are constantly evolving to meet user demands. Initial data networks were limited to exchanging character-based information between connected computer systems. Traditional telephone, radio, and television networks were maintained separately from data networks. In the past, every one of these services required a dedicated network, with different communication channels and different technologies to carry a particular communication signal. Each service had its own set of rules and standards to ensure successful communication.

The Open Systems Interconnection (OSI) transport layer protocols, transmission control protocol (TCP) and user datagram protocol (UDP) are being used to overcome the drawbacks. Nowadays, VoIP is one of the prominent and fastest growing telecommunication service based on an Internet protocol suite. VoIP enables the users to use the Internet as the transmission medium for voice communication. The general architecture of the VoIP system presented in Figure 1.2 demonstrates some of the basic requirements for implementing a VoIP system.

Advances in technology are enabling us to consolidate these different kinds of networks onto one platform referred to as the “converged network”.

Unlike dedicated networks, converged networks are capable of delivering voice, video streams, text, and graphics between many different types of devices over the same communication channel and network structure. Previously separate and distinct communication forms have converged into a common platform [16].

In traditional systems, increasing the connectivity, means more ports to connect in telephones which leads to an increase in cost. But in the VoIP system, no extra cost would be paid to increase the connectivity. In VoIP, increase in cost relay on the functionality improvement. VoIP is software based, so any type of addition can be done just by upgrading the software. The hardware cost is also very low because it runs on computer platforms [17].

The VoIP system has a lot of advanced features, which makes the VoIP system the best alternative to the traditional circuit network. These features attract the communication industries and business community towards the new telephony.

Low cost is the main advantage of the VoIP system. The long distance calls can be made at very low cost through the VoIP system, since it can access the public Internet and the low cost broadband connection, increases the adaptation of VoIP system in the small business world, too. It uses the existing infrastructure such as computer system and Internet to provide the communication without any addition cabling cost [15].

Data networks are based on packet switching and can be architected to be reliable and resilient, providing alternative paths to data in case of failure. VoIP systems benefit from that design and are less susceptible to total failure. Because the sound must be digitally encoded to be packetized, the resulting sound is more robust compared to the traditional analog systems, as showed in Figure 2.1. Although digitally encoded speech has many advantages over its analog counterpart, it nevertheless requires extra bandwidth for transmission, if it is applied without compression. To counteract this, VoIP systems use coder-decoder (codec) through software and hardware to compress and decompress sound. The selection of a suitable codec for the VoIP system is of much importance, since this selection may affect the system performance. Table 2.1 describes some codecs and various characteristics [18].

Table 2.1 - Comparison of the most well-known voice codecs.

Codec	*Year*	*Bit rate (kb/s)*	*Algorithmic delay (ms)*	*Mean Opinion Score (MOS)*
G.711	1972	64	0.125	4.1
GSM-FR	1987	13	20	3.6
G.722	1988	48, 56, 64	1.5	4.1
G.726	1990	24	0.125	3.5
G.726	1990	32	0.125	4.1
GSM-HR	1994	5.6	24.4	3.5
G.723.1	1995	6.3	30	3.8
G.723.1	1995	5.3	30	3.6
G.729	1996	8	15	3.92
G.729A	1996	8	15	3.7
GSM-EFR	1998	12.2	20	4.1
AMR-NB	1999	4.75 - 12.2	25	3.5 - 4.1
G.722.1	1999	24, 32	40	4.0
Speex (NB)	2002	2.15 - 24.6	30	2.8 - 4.2
iLBC	2004	13.33	40	3.8
iLBC	2004	15.2	25	3.9
BV32	2005	32	5	4.1

The most obvious way to measure speech quality is to go right to the source and use human subjects to rate the quality of the telephone calls. This is a subjective method, described in two ITU recommendations [19] [20]. The primary results of the subjective test are mean opinion score (MOS), which rate the quality of a telephone call from an end-user perspective. The MOS score is a subjective score of voice quality as perceived by a large number of people listening to speech over a communication system.

To determine the MOS for a particular phone connection, a statistically valid group of males and females rate the quality of special test sentences read aloud over the connection. This recommendation uses the scale from 1 to 5 and the MOS of the voice transmission is the average estimate of voice quality rates assigned by this statistical group as presented in Table 2.2.

Table 2.2 – Subjective speech quality measurement.

Speech quality	*MOS*
Excellent	5
Good	4
Fair	3
Poor	2
Bad	1

An MOS of 4.0, is considered toll quality (PSTN quality) within the telephone industry and anything below MOS of 4.0 would then be below toll quality level. Subjective testing includes both listening opinion and conversation opinion tests.

Delay is the time taken by the voice to reach from the talker's mouth to the listener's ear. Round trip delay is the sum of two one-way delays that occur in the user's call. In a VoIP system, beside the encoding delay and decoding delay, the propagation delay is also affected by three additional delays, packeting delay, network delay and playback delay. These delays may be variable during the transmission [21]. Table 2.3 resumes the general five components that contribute to the overall delay.

Table 2.3 – Five components that contribute to the mouth-to-ear delay

Type of delay	*Description*
Encoding delay	The amount of time needed to encode the voice signal, which depends on the voice codec used
Packetization delay	The amount of time required to packetize the encoded voice stream
Network delay	The amount of time to transmit the packets to the receiver’s side
Decoding delay	The time interval needed to reconstruct the voice signal
Playback delay	The delay induced by the playback buffer that resides at the receiver’s side. This buffer is needed to smooth delay between consecutive packets (jitter)

One-way delay, *i.e*, from mouth-to-ear, should be less than 150 milliseconds, latency greater than 400 milliseconds is unacceptable [22]. Table 2.4 describes the delay according to the user's satisfaction.

Table 2.4 – Delay limits for one-way transmission.

One-way delay milliseconds (ms)	*Description*
0-150	Acceptable for most users
150-400	Acceptable but with impact
Above 400	Unacceptable

Compared to traditional telephony (circuit switching), VoIP is now more scalable, robust, reliable, allows the integration of new features and the possibility of reducing costs. From the perspective of data networks, VoIP has transformed the voice into just another application [5].

2.2. Unified Communications

VoIP as evolved on the last years and today offers a complete set of tools called unified communications (UC). UC integrates real-time communication services such as instant presence information, messaging (chat), desktop sharing, telephony, video conferencing, data sharing, call control and speech recognition. UC isn't necessarily a single product, instead, it's a set of products that provides a consistent unified user-interface and user-experience across multiple devices and media-types.

UC products facilitate the use of multiple enterprise communications methods to improve user productivity and to enhance business processes. This can include the management, control and integration. UC products integrate communications channels, networks and systems, as well as IT business applications, consumer applications and devices. Offers the ability to significantly improve how individuals, groups and companies interact and perform. The UC products may be composed of a single vendor suite, or may span multiple platforms and vendors. These products are used by individuals to facilitate personal communications, and by enterprises to support collaborative communications and business workflows. Some products may even extend UC to the cloud.

Table 2.5 – UC communications areas.

Telephony	This area includes mobile, fixed and soft telephony.
Conferencing	This area includes videoconferencing, voice conferencing and Web conferencing. Includes application sharing and document sharing.
Messaging	This area includes voice mail and email.
Presence and IM	Presence allows individuals to see the status of other people and resources. Instant messaging allows individuals or groups to send and receive text information in real time.
Clients	Unified clients enable access from various platforms from a consistent interface. May include thin browser clients, thick desktop clients, and mobile clients in smartphones and tablets.
Communications-enabled applications	Communications-enabled applications include key application in collaboration area. This area includes contact center, notification, and consolidated administration, reporting and analytics tools.

Many authors find it useful to divide UC into six broad communication areas, according to Table 2.5.

Gartner is the world's leading information technology research and advisory company, delivering the technology-related insight necessary to make the right decisions. Figure 2.3 presents the major players for UC [23].

Figure 2.3 - Gartner magic quadrant for unified communications.

The leader's quadrant will be used to focus the best solutions available in the market [23]. Asterisk UC will also be included in this study for its popularity in the open source community.

All UC solutions in study allow IVR configuration. IVR allows call handling, in order to give some sort of response to them without human interaction. Being the caller responsible to choose one of the options that are presented. IVR is usually presented in a form of a menu that can have many layers or submenus and are composed by audio files played to the caller, witch, through touch keys on the phone can interact with the system, for example, allowing the creation of a voting system via phone, money transfer from a bank account or to choose the more skilled operator in certain subject [24].

2.3. Microsoft UC solution

Microsoft Lync offers UC functionality. Microsoft continues to improve UC functionality with each release and integrates with Active Directory, Office applications and Skype. Microsoft has a large set of business applications that will increasingly be leveraged, including Cortana (a digital assistant) and Office Graph (which uses machine learning to define the context and connect users with relevant documents, conversations and people).

Microsoft Lync federation capabilities are an effective way for groups to collaborate. Microsoft Lync Online is part of Office 365, however, Lync Online only offers a subset of the on-premises solution, with limited PSTN connectivity.

According to its manufacturer, organizations with a significant number of employees that can benefit from Lync's collaboration model, should consider the Lync solution. Organizations that use advanced telephony feature requirements should also ensure that the needed functions are supported.

Also according to its manufacturer Microsoft Lync continues to make significant gains in the market and is attractive to a broad range of enterprises. In many cases, it is initially deployed for its IM, presence and Web conferencing functionalities, with gradual incremental deployments of telephony and video added as follow-on phased deployments for specifically targeted groups or regions.

Microsoft has added video capabilities. Support for Lync room-based video systems and interoperability of standards-based video endpoints were included.

Customers report that Lync functions can be readily integrated into business processes and applications, providing new, different and effective ways to perform tasks. Often, these new functions are achieved by deploying Lync enhancements from a growing list of ecosystem partners.

Lync implementation might not eliminate legacy PBXs. Usually, Lync Web conferencing and IM/presence are deployed across the organization, while telephony is deployed only for a subset of employees. Telephony implementation will include third party’s software and hardware. Deploying Lync might be a challenge, multiple partners may be required to obtain a complete deployment, for example, different partners for telephones, gateways, servers, remote support and network monitoring. This can lead to incompatibilities on software from any of these partners.

According to the documentation it's possible to do IVR reporting. Documentation indicates the need to export the data and then run the analyses using PowerShell. A PowerShell example would be [25]:

```
$calls = Import-Csv -Path
"C:\Response_Group_Call_List_Report_export.csv"

$calls | Group-Object Workflow | Select-Object Count, Name
| Sort-Object Count –Descending
```

The resulting information would be similar to Figure 2.4 [25].

```
Count  Name
-----  ----
  190  Help Desk
   30  Sales
   17  Customer Support
   12  Employment Opportunities
```

Figure 2.4 - Example of IVR Lync reporting using PowerShell.

2.4. Cisco UC solution

Cisco offers a full UC suite, and as part of it a broad range of communications functions. Key parts of the UC suite include Cisco Jabber, CUCM, Cisco Unity Connection and Cisco WebEx. The entire portfolio is available on VMware, operating on Cisco Unified Computing System (CUCS) servers or other qualified servers. Cisco leverages its UC software into a cloud portfolio branded Cisco Hosted Collaboration Solution (Cisco HCS), which allows to build Unified Communications as a Service (UCaaS) or hybrid options on premises.

According to the vendor, Cisco UC has strong voice and video capabilities and solutions from midsize to multinational sites. It's very attractive for enterprises that wish to leverage Cisco's networking infrastructure or that require full UC client integration on leading mobile platforms. A full UC suite is offered with IM/presence, video, multiple conferencing options and telephony, also available on leading mobile platforms.

Through Internet service providers (ISP), Cisco is advancing hybrid on-premises and cloud options. Cisco HCS is based on the on-premises software and both support the same Jabber client, there are also the possibility to transfer licenses from on-premises to hosted environments.

Cisco UC solution is based on a set of distinct products and acquisitions over the years, as a result some administrator and user experiences are fragmented.

IVR possibility comes in two flavors Unified Contact Center Express, for small implementations and Unified Contact Center Enterprise for larger deployments [2].

The ability to create IVR reports is given by one add-on tool called Cisco Unified IP IVR. This tool can create reports for application performance analysis report, detailed call by call report and traffic analysis report for answered and unanswered calls [2].

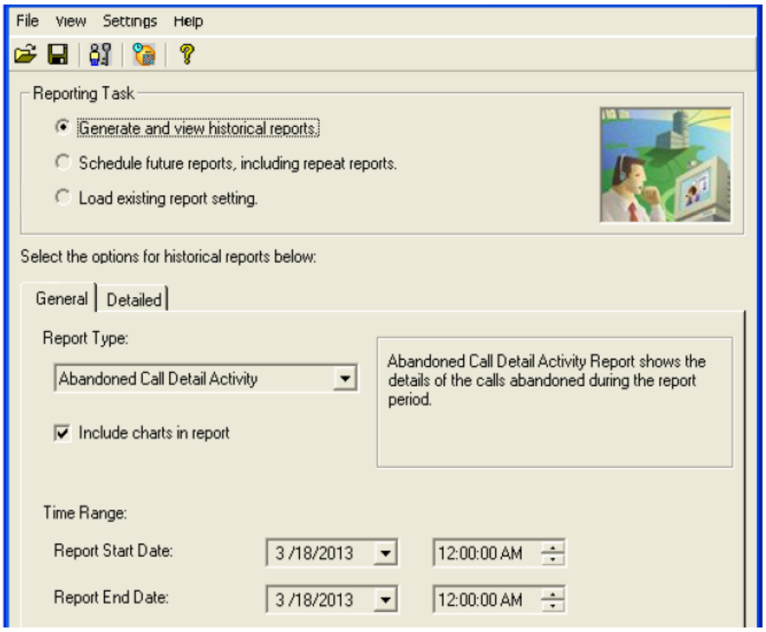


Figure 2.5 – Report generation for abandoned calls.

Figure 2.5 show's how to create a report for Cisco Contact Center Express. There's also the possibility to schedule reports and load existing reports. For Enterprise version the Unified Intelligence Center is a more complete tool. It's possible to create Dashboards, and reporting is more flexible. Figure 2.6 show's a simple example.

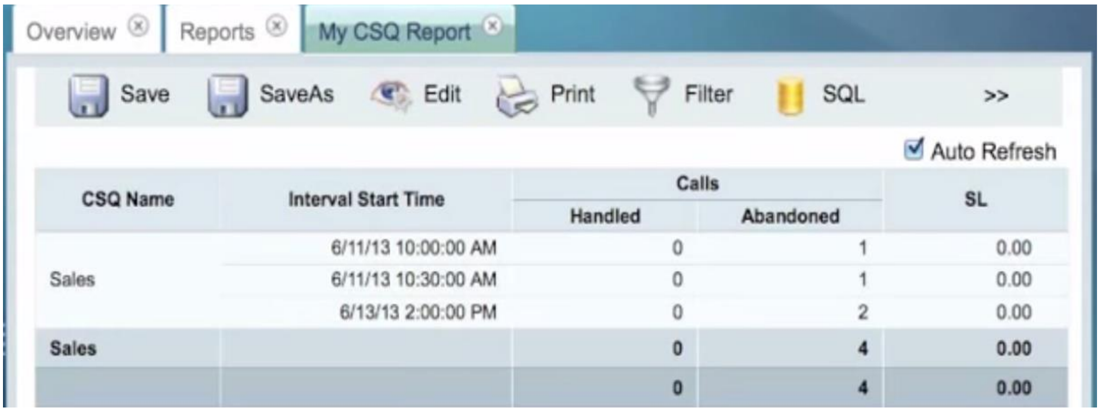


Figure 2.6 – Report generation using Unified Intelligence Center.

In Figure 2.6 the report was manipulated by the administrator just to show handled calls, abandoned calls and service level for the analyzed period. Reports still have to pass through a configuration menu indicating start and end date, similar to Figure 2.5. For advanced users there's also the possibility to run SQL queries.

2.5. Avaya UC solution

Avaya is one of the leaders in Gartner review and has a wide set of communications products. The UC product is Avaya Aura Platform. Other elements in the UC portfolio are: Avaya Aura Conferencing, Avaya Aura Messaging, Scopia Video Conferencing and the Avaya Aura Collaboration Environment. There are also available UC applications for desktop, mobile, phone and video. Avaya Aura is considered a good solution in the presence of heterogeneous environments (systems, services and devices). The vendor's strength and brand recognition in telephony and contact center are central key elements for Avaya's products. The UC solution is advancing to the cloud and is based on the same solution as the on-premises version.

Avaya is expanding its portfolio in Avaya Aura Collaboration, expanding middleware integration options there for enabling a broader ecosystem of partner applications. The expansion on the integration capabilities, allows a stronger multivendor UC integration, for example, integration with Microsoft Lync. The multivendor integration and the portfolio redundant capabilities (Scopia, Aura Conferencing and Avaya one-X Communicator) might lead to client confusion in the sales process and the Avaya solutions themselves appears individual point solutions, rather than as a comprehensive UC suite.

Avaya's contact center solutions are: Aura Contact Center and Aura Call Center Elite, both capable of IVR. Reports are also available in this solution but have to pass through a few configuration menus indicating the kind of report and the span date.

Sample Agent Performance

Report Interval: 4/ 1/2013 12:00:00 AM - 4/10/2013 11:59:59 PM (GMT)
Site Name: CHRYAACC
Table Name: dAgentPerformanceStat

	Calls Offered	Calls Answered	Avg Total Talk Time	Avg Wait Time	Not Ready Time	DN In Calls	DN In Talk Time	DN Out Calls	DN Out Talk Time	Logged In Time	%Work
Supervisor Name - ID:	**Default Supervisor - 0**										
Agent Name - ID:	**2008 2008 - 2008**										
4/10/2013	0	0	00:00:00	02:45:17	00:00:11	5	00:00:39	1	00:00:20	02:45:28	0.11
Agent	0	0	00:00:00	02:45:17	00:00:11	5	00:00:39	1	00:00:20	02:45:28	0.11
Agent Name - ID:	**2010 2010 - 2010**										
4/1/2013	0	0	00:00:00	08:56:01	00:01:06	0	00:00:00	0	00:00:00	08:57:07	0.20
4/2/2013	0	0	00:00:00	13:50:45	00:00:21	0	00:00:00	0	00:00:00	13:51:06	0.04
4/5/2013	0	0	00:00:00	00:16:02	00:00:00	0	00:00:00	0	00:00:00	00:16:02	0.00
Agent	0	0	00:00:00	23:02:48	00:01:27	0	00:00:00	0	00:00:00	23:04:15	0.10

Figure 2.7 – Report example on agent performance.

2.6. Mitel UC solution

In 2013, Mitel changed its offerings, simplifying the buy experience. In the end of 2013 Mitel and Aastra entered in a merge process. The deal was closed in January 2014 and gave birth to a new company under the name of Mitel, while continuing to leverage Aastra’s strong brand recognition in European markets. Aastra had a strong portfolio for telephony, video, call centers and UC. Mitel pretends to use MiCollab suite as the common UC solution across the various call management platforms. MiCloud service is also available as UCaaS based on MiCollab, MiVoice and MiContact Center. The incorporation of Aastra video capabilities into the existing MiCollab video option and into MiVoice Video Unit are on track. One big concern about Mitel and their products is that the expanded portfolio may reduce Mitel ability to research, development and support.

The Mitel MiVoice, MiCollab and MiContact Center solutions provide a mature and comprehensive software suite. The common based software architecture makes it possible to use a distributed solution or centralized solution in a data center. They run on industry-standard servers, and are certified for virtualizations on VMware environments. The same solutions are also available as cloud offerings.

Mitel MiContact Center sits on top of MiVoice product and has two versions, Business Edition delivers robust contact center, IVR, multimedia functionality and reporting but packaged specifically for small contact centers. For large scale Mitel MiContact Center Enterprise Edition is a robust, highly flexible solution that delivers feature-rich IVR capabilities, contact center monitoring and reporting. Reporting can be made by request or by schedule. It's very similar to other vendors solutions, in this case the result is an excel file that can be later worked by a statistician or imported to data warehouse for data mining. Mitel also allows access to its database raw data, for custom reports.

2.7. Asterisk UC solution

Asterisk system is built by a community of developers and is targeted to communication systems developers. The result is an engine that is able to handle the low-level details of initiating, maintaining and manipulating calls between endpoints (phone terminals). Since the initial release it's been tested and refined by a community of more than 80,000 developers. With the right developer skills, raw configuration files and custom scripts, it's possible to provide UC capabilities like: voice messaging, instant messaging, desktop fax, drag/drop call control, multi-party conferencing and IVR [5].

There are many forks from Asterisk, some are free to use and usually have no support, like: FreePBX or Trixbox CE, while others, are commercial versions and typically provide software, hardware and support for both. Some commercial versions of Asterisk are: Switchvox, Masip, Zycoo, Enterprise Courier and Xorcom, among many others.

Asterisk includes a wealth of functions that make it a powerful IVR platform: audio playback, audio recording, queues and calendar integration. For Asterisk, IVR is just a special use of the Dialplan, as showed in Figure 2.8 [26].

```
exten => 90,1,Answer()
exten => 90,2,Background(hello_world_audio_file)

;If you press 1
exten => 1,1,Playback(you_pressed_1_audio_file)
exten => 1,2,Hangup()

;If you press 2
exten => 2,1,Playback(you_pressed_2_audio_file)
exten => 2,2,Hangup()
```

Figure 2.8 - Asterisk IVR Hello World, using Dialplan language.

Figure 2.8, shows how the Dialplan can be used to create IVR function in Asterisk. In this example, if someone dials the extension 90, the system will pick up the call and will play the audio file called `hello_world_audio_file`. Meanwhile if the key 1 is pressed the system will play the audio file called `you_pressed_1_audio_file` and afterwards the system will hang-up the call. A similar action will occur if key number 2 is pressed.

IVR are just calls and being so, one could think about using Call Detail Record (CDR) for reporting.

In Asterisk based systems, CDR-stats [27] is a separate application that analyses the CDR and creates statistics from there. There are two options: professional support (commercial version) or community support (free to use). It's one interesting tool for CDR reporting, but it's not by any mean IVR specialized, the report will have inconsistent results. Being IVR just a special use of the Dialplan, creates a few difficulties for IVR reporting, because generally calls that enter IVR will generate more than one CDR record.

QueueMetrics is a call center analysis, monitoring and reporting software application for Asterisk. Because it is specialized for call centers it has a special treatment for IVR. QueueMetrics can be integrated with some Asterisk based systems and it's a commercial solution with a license renewal every four years. This software creates reports for total calls, answered calls, unanswered calls, Call distribution, Agents and Call detail, as showed in Figure 2.9 [28].

Answered call distribution per hour

Hour	Num		Answered calls	Avg	Min	Max	Avg duration
00:00	72	3.4%		60.0 s.	20 s.	150 s.	
01:00	72	3.4%		60.0 s.	20 s.	150 s.	
02:00	72	3.4%		60.0 s.	20 s.	150 s.	
03:00	72	3.4%		60.0 s.	20 s.	150 s.	
04:00	72	3.4%		60.0 s.	20 s.	150 s.	
05:00	72	3.4%		60.0 s.	20 s.	150 s.	
06:00	72	3.4%		60.0 s.	20 s.	150 s.	
07:00	72	3.4%		60.0 s.	20 s.	150 s.	
08:00	72	3.4%		60.0 s.	20 s.	150 s.	
09:00	72	3.4%		60.0 s.	20 s.	150 s.	
10:00	72	3.4%		60.0 s.	20 s.	150 s.	
11:00	72	3.4%		60.0 s.	20 s.	150 s.	
12:00	72	3.4%		60.0 s.	20 s.	150 s.	
13:00	72	3.4%		60.0 s.	20 s.	150 s.	
14:00	72	3.4%		60.0 s.	20 s.	150 s.	
15:00	72	3.4%		60.0 s.	20 s.	150 s.	
16:00	72	3.4%		60.0 s.	20 s.	150 s.	
17:00	119	5.7%		60.3 s.	20 s.	150 s.	
18:00	144	6.9%		60.0 s.	20 s.	150 s.	
19:00	144	6.9%		60.0 s.	20 s.	150 s.	
20:00	144	6.9%		60.0 s.	20 s.	150 s.	
21:00	144	6.9%		60.0 s.	20 s.	150 s.	
22:00	97	4.6%		59.6 s.	20 s.	150 s.	
23:00	72	3.4%		60.0 s.	20 s.	150 s.	

Figure 2.9 – Scheme depicting QueueMetrics Answered call distribution per hour.

Figure 2.9 shows peak hours of IVR use, in this example, peak hours are between 17:00 and 22:00, this can be very useful for assigning more workforce at this hours.

2.8. Call Center Management and Reporting

Call Center or Contact Center equipment currently available generates vast amounts of data related to the daily operation of a call center. This data is used by the UC equipment to route calls and logging, but also has the potential to reveal important details about agent's productivity and performance. This data can reveal user experience provided by the contact center to each person that placed a call in the system. To date there has been no public system or method which has been able to gather, organize, interpret, and report vital data to managers and supervisors in a manner which unleashes the full potential and value of such data [29].

Chapter 3

Architecture and solution to the problem: a comprehensive IVR analysis model using online analytical processing on a multidimensional data cube

3.1. Requirements

As presented in the introduction section, there was a need to develop new solutions for a system analysis approach in order to respond to the UBI's 2020 mission call for constant improvement on all University services.

The relevant stakeholders were, primarily, the Vice-Rector for the Academic Affairs, and also, too much lesser extent, the users at the Academic Services, and external users that contact UBI using the telephone.

Based on the interviews conducted with the stakeholders, the following list of functional requirements (FR) and non-functional requirements (NFR) was produced for the answered and unanswered calls:

FR1: The system must be able to provide metrics on the efficiency of the incoming calls, *e.g.*, the system must provide information about the answered and unanswered calls per day or set of consecutive days, month or set of consecutive months, quarter or set of consecutive quarters and year or set of consecutive years;

FR2: The system must be able to provide information about the answered calls per operator number;

FR3: The system must be able to provide information about the distribution of the incoming calls, *i.e.* the system must provide information about the answered and/or unanswered calls per IVR menu option;

FR4: The system must be able to provide information about the time spent, *i.e.* the system must provide information about the waiting time and conversation time.

NRF1: The system must not install any software in the IP PBX Enterprise Courier.

NRF2: The system must not interfere with the IP PBX system, *e.g.* the system must not alter the IP PBX databases.

NRF3: The system must not interfere with the VoIP performance.

It must be noted that this list of requirements is not final, as the approach to the development of the system was performed in a prototyping manner, thus allowing the stakeholders to refine the requirements list, *i.e.*, the results that the system must output will be subject of continuous improvement.

3.2. Call Detail Record structure

The IP PBX Enterprise Courier stores locally, in a MySQL database, information about all calls in a format named Call Detail Record (CDR) and in particular, calls that enters into the IVR. The CDR structure has the following fields: `cdr_id`, `calldate`, `clid`, `src`, `dst`, `dcontext`, `channel`, `dstchannel`, `lastapp`, `lastdata`, `duration`, `billsec`, `disposition`, `amaflags`, `accountcode`, `uniqueid` and `userfield`.

3.3. Retrieved fields to Data WareHouse

From all of the fields, there are a few with special interest for this study: `cdr_id`, `calldate`, `src`, `dst`, `duration`, `billsec`, `disposition` and `uniqueid`.

The `cdr_id` field is a sequential number generated when the database record is created.

The `calldate` field has the timestamp for the call beginning and is a datetime type.

The `src` field has the call source, *e.g.* “+351275319700” and is of varchar type.

The `dst` field has the call destination, *e.g.* “1113” and is a varchar type.

The `duration` field has the call total time in seconds and it is of int type.

The `billsec` field has the call conversation time in seconds and it is of int type.

The `disposition` field is a varchar type and can take two possible strings: “ANSWERED” or “UNANSWERED”.

The `uniqueid` field has the id for the call in question and is a varchar type.

The relevant data fields [30] for this study are retrieved from Enterprise Courier IP PBX by the data warehouse and are stored for comprehensive and detailed analysis as it is showed in Figure 3.1. The data warehouse retrieval SQL job, fetches new data, based on `CDR_id`, i.e, twice a day and it can be adjusted for more sensitive real-time data, if required.

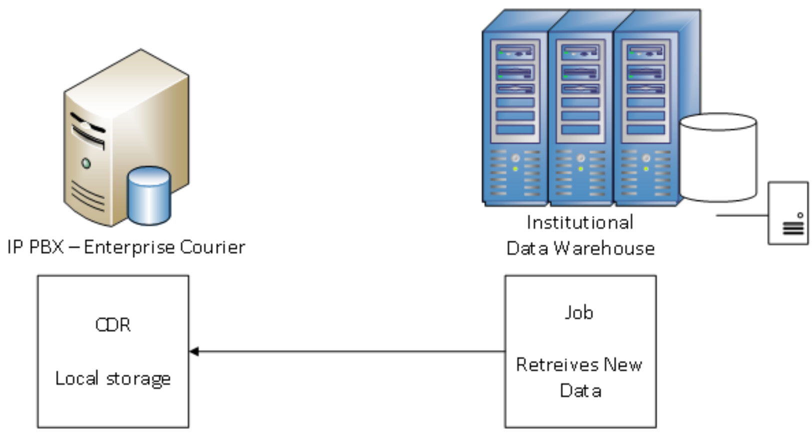


Figure 3.1 - Data retrieval scheme.

As stated before in Chapter 2, usually there are two database rows for the same call. Actually, this only occurs if the call is answered. The first CDR is generated for the receiving call stating the queue of the call. This queue can then be translated to the appropriate IVR option. The second CDR record is generated for the operator that took the call, as showed in Table 3.1.

Table 3.1 – Records representing one answered call.

cdr_id	*calldate*	*src*	*dst*	*duration*	*billsec*	*disposition*
2448554	2014-07-15 15:57:01	91XXX6962	9000005	223	223	ANSWERED
2448239	2014-07-15 15:57:34	91XXX6962	1113	192	188	ANSWERED

In Table 3.1 it was omitted the field `uniqueid`, the value is "`asterisk-1405436252.64896`" for both records.

The total call length, since the system pick up the call, until the operator or user ended the call, is given by the first record in the `duration` field, in this case 223 seconds.

Total conversation time is given by the second record, in the field `billsec`, that in this example is 188 seconds.

The waiting time for answered calls is defined in this study as showed in Equation 3.1 and Equation 3.2.

$$wait\_time = Total\_call\_lengh - Conversation\_time. \tag{3.1}$$

Substituting with the values of Table 3.1, the result will be:

$$wait\_time = 223 - 188 = 34 \, seconds. \tag{3.2}$$

The destination field (`dst`), in the first record with the value "`9000005`" represents the queue that took the call. This value can be translated to IVR option number "5", and according to Figure 1.3, represents "Other issues".

The destination field (`dst`), in the second record with the value "`1113`" represents the actual phone extension that took this call.

The source field (`src`), in both records was partially omitted for privacy reasons.

Table 3.2 – Record representing one unanswered call.

cdr_id	*calldate*	*src*	*dst*	*duration*	*billsec*	*disposition*
1847989	2014-04-22 15:13:37	24XXX5728	9000005	173	173	ANSWERED

Unanswered calls only generate one record, like it's showed in Table 3.2. The `disposition` field shows the value "`ANSWERED`" because the system picked up the call and presented the IVR menu options to the user. The second record doesn't exist because the operator didn't took this call.

The `duration` field represents the total time of this call. This value is defined for this study, as the waiting time, that in this case was 173 seconds, approximately 2 minutes and 53 seconds.

The destination field (`dst`), with the value "`9000005`" represents the queue that took the call. This value can be translated to IVR option number "5", and according to Figure 1.3, represents "Other issues".

In Table 3.2 it was omitted the field `uniqueid`, the value is "`asterisk-1398176017.17671`" and in the case of unanswered calls, this value is in fact unique.

3.4. Cube Structure

Online Analytical Processing (OLAP) is a technology that is used to organize large business databases and support business intelligence. OLAP databases are constituted by cubes.

Online Analytical Processing (OLAP) databases facilitate business intelligence queries. OLAP is a database technology that has been optimized for querying and reporting, instead of processing transactions.

OLAP data results from historical data and is aggregated into structures that permit sophisticated analysis. OLAP data is also organized hierarchically and stored in cubes instead of tables. It is a sophisticated technology that uses multidimensional structures to provide rapid analysis.

With the imported data, a multidimensional data model is constructed at the institutional data warehouse. This model views data in the form of a data cube [31] [32] [33]. A data cube allows data to be modeled, viewed and filtered in multiple dimensions.

Cube is a data structure that aggregates measures by the levels and hierarchies of each of the dimensions that are needed to analyze. Cubes combine several dimensions, such as time, geography, and product lines, with summarized data. Cubes are not "cubes" in the strictly mathematical sense, however, they are an appropriate representation for the concept.

The cube is defined by dimensions and facts [34]. Dimension tables are integral companions to a fact table. Dimensions are a set of one or more organized hierarchies of levels in a cube that a user understands and uses as the base for data analysis. The dimension tables contain the textual descriptors. Each dimension is defined by its single primary key, designated by the PK notation in Figure 3.2, which serves as the basis for referential integrity with any given fact table to which it is joined. Dimension table attributes play a vital role in the data warehouse, since they are the source of virtually all interesting constraints [35].

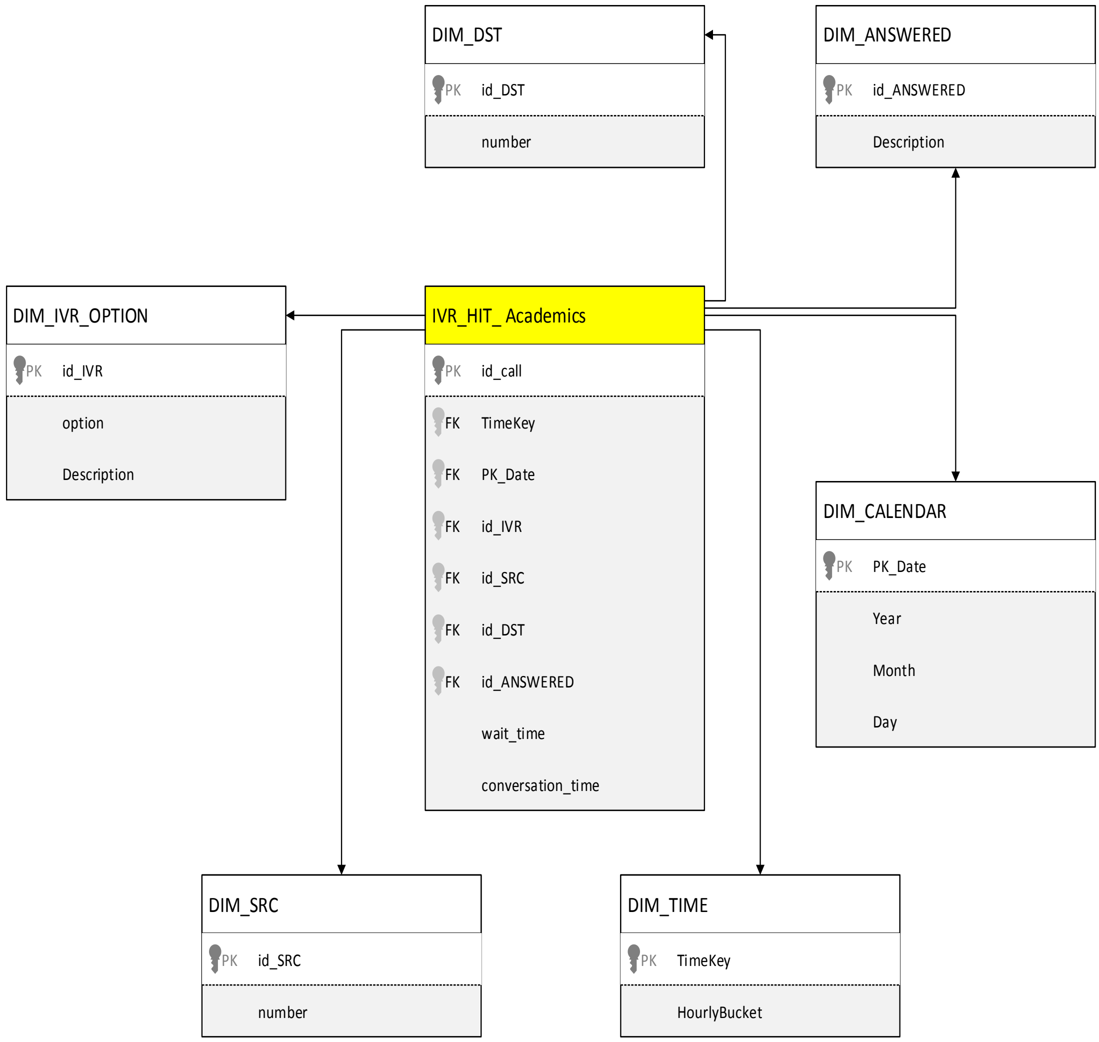


Figure 3.2 - Cube structure view.

A fact table is the primary table in a dimensional model [34]. The fact table contains the names of the facts, or measures, according to Figure 3.2, "`wait_time`" and "`conversation_time`". Measures are the central values in the cube that are processed, aggregated and analyzed.

In addition to these measures, calculated measures were introduced based on this two primary measures, `MAX_wait_time` as the maximum value for `wait_time`, `MIN_wait_time` as the minimum value for `wait_time`, `Avg_wait_time` as the average value for `wait_time`, `MAX_conversation_time` as the maximum value for `conversation_time`, `MIN_conversation_time` as the minimum value for `conversation_time` and `Avg_conversation_time` as the average value for `conversation_time` were introduced, as well foreign keys to each related dimension tables, designated by the FK notation in Figure 3.2.

The fact table itself generally has its own primary key made up of a subset of the foreign keys. This key is often called a composite or concatenated key, in Figure 3.2 is showed as `id_call`.

A data warehouse requires a concise, subject-oriented schema that facilitates online data analysis. A star schema [32] for the multidimensional analysis model, Figure 3.2, with the fact table in the middle. As this schema was applied on IVR data, it was decided to call it IVRCube.

For the multidimensional analysis model were defined six dimensions [36]: `DIM_CALENDAR`, `DIM_TIME`, `DIM_DST` (stands for destination), `DIM_SRC` (stands for source), `DIM_ANSWERED` and `DIM_IVR_OPTION`.

The `DIM_DST` dimension has all possible destination numbers, in this study, it's the operator numbers. `DIM_SRC` dimension has all source numbers from the data source. `DIM_SRC` and `DIM_DST` dimensions has all sources and destinations respectively, from the data source and are populated from the retrieved data (Figure 3.1).

`DIM_CALENDAR` dimension is the one dimension nearly guaranteed to be in every data mart, because virtually every data mart is a time series. Calendar dimension was created with an SQL script and has all possible data from 2014 until 2020.

`DIM_TIME` dimension gives a perspective of how hours of the day may influence the service (for example peak hours). TIME dimension was created with an SQL script and has all possible hours from 00h until 23 hours 59 minutes 59 seconds, distributed in hourly bucket.

The `DIM_ANSWERED` dimension takes just two possible values, according to Table 3.1 it will take the "`ANSWERED`" value or according to Table 3.2 it will take the "`UNANSWERED`" value, *i.e.*, the value in `DIM_ANSWERED` dimension is manipulated to represent answered or unanswered calls.

The `DIM_IVR_OPTION` dimension has all possible options the clients can choose from the IVR menu. This was the only dimension that was manually populated with the names of the possible options of the IVR menu, according to Figure 1.3, "Access to University", "Application and Enrollment", "Certificates and Diplomas", "Tariff and Accreditations", "Equivalence and Recognition of academic degrees" and "Other issues".

Chapter 4

Results and Discussion

High level decisions often want to see a big picture indicating broader trends based on aggregated data, or to see these trends broken down by any number of variables. Business intelligence is the process of extracting data from an OLAP database and then analyzing that data for information that can be used to make informed business decisions.

The IVR in study is running since late January of 2014 and the presented results include processed information since then until September 5, 2014. OLAP data was retrieved using Microsoft Excel. Results may be presented in table form, or as charts. In this section, some of the most relevant views from the multidimensional analysis are presented.

According to Figure 4.1 the three most used IVR options are: “Application and Enrollment”, followed by “Certificates and Diplomas” and “Access to University”.

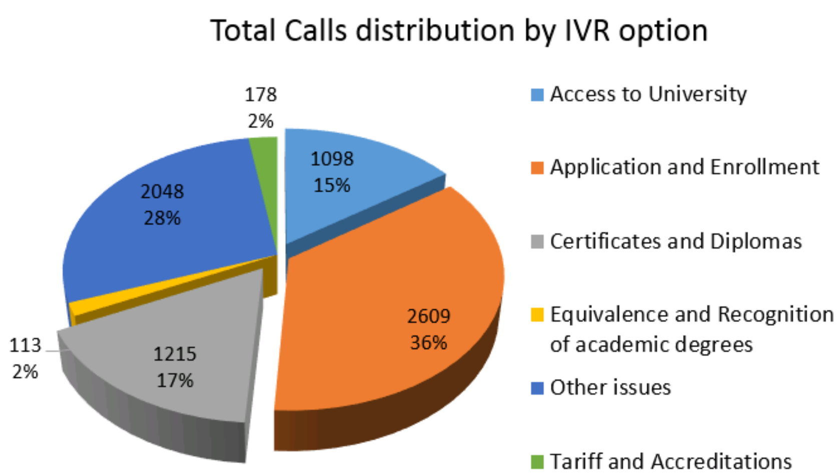


Figure 4.1 - Total Calls (Answered and Unanswered) distribution by IVR option.

As stated before, there were great concerns ordering the six menu options. As data showing suggests in Figure 4.1, a change in IVR menu order should take place in a near future, as proposed in Table 4.1.

Table 4.1 - Proposed reorder of IVR menu options.

New IVR menu position	*IVR menu Option Description*	*Older Position*
1st	Application and Enrollment	2nd
2nd	Certificates and Diplomas	3rd
3rd	Access to University	1st
4th	Tariff and Accreditations	4th
5th	Equivalence and Recognition of academic degrees	5th
6th	Other issues	6th

The three first options are actually the options with more hits. The empirical reasons behind the ordering were reasonable, since shifting option “Access to University” to third place matched the correct IVR menu order.

Options in fourth, fifth and sixth positions maintain the same positions as the originals. “Other issues” option should maintain the last option, because it’s a last resort option.

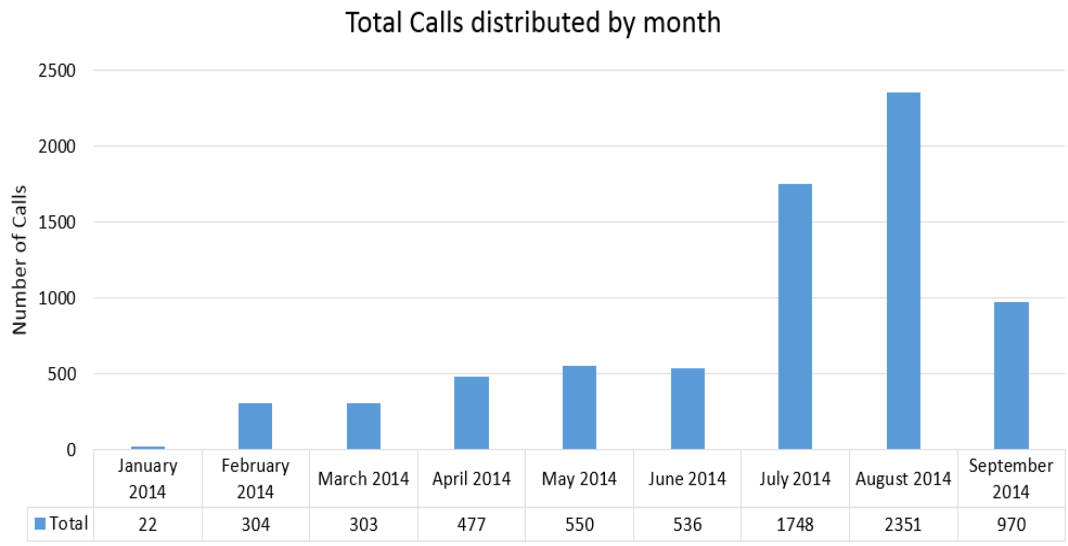


Figure 4.2 - Total (Answered and Unanswered) calls distribution by month.

In the monthly distribution, January had a very small number of calls, most likely because it was when the IVR service was created and it wasn't properly advertised. Calls volume triples in July and August compared to previous months. Data available for September only includes the first week, according to Figure 4.2 it's expectable that September total calls will exceed August total calls.

In Figure 4.3, additional detail shows the evolution on the motivation for the calls in the months of July, August and September and compares it to calls in the month of June.

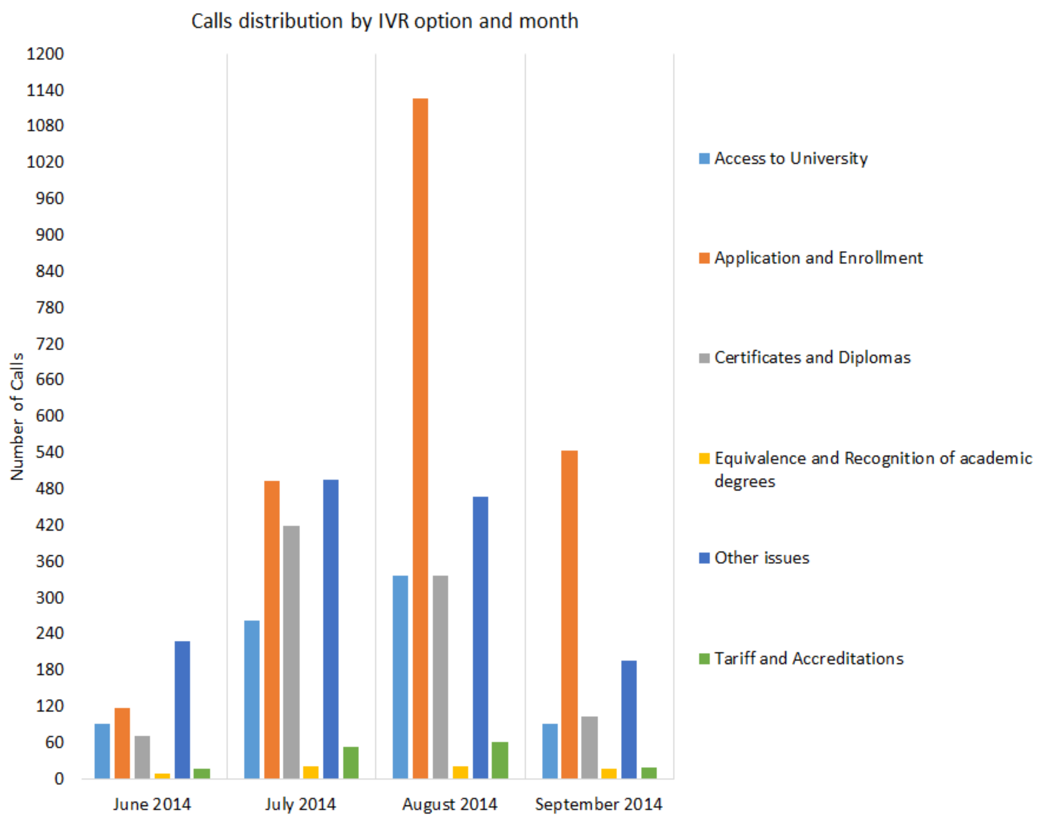


Figure 4.3 - Total Calls distribution by IVR option and by month.

The month of July, relative to June (Figure 4.3) shows a global increase in all IVR options. In particular the options: “Certificates and Diplomas”, “Access to University” and “Application and Enrollment”.

July is the end of the scholar year, and recent graduated students need their certificates and diplomas to compete in the labor market, explaining the abnormal amount of calls related to “Certificates and Diplomas”.

In Portugal, from July 17 until August 8 and from September 8 until September 12, are periods of the year where students choose which University they want to enroll. From September 6 until September 10 is the first period for new students to enroll in their courses. Both of these two situations explain the call increase in options “Access to University” and “Applications and Enrollment” for the months of July, August and September.

Figure 4.4 presents call distribution by operator. Extension 1113 answerers most of the calls 52% and, according to Figure 1.3, this is perfectly acceptable since this operator responds to three options of IVR menu, as showed in Figure 1.3.

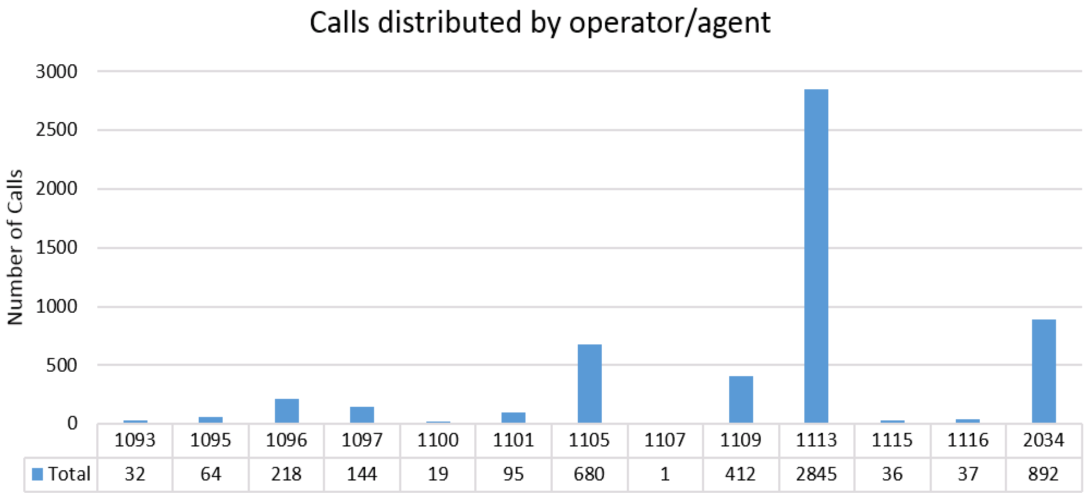


Figure 4.4 - Calls distribution by operator extension.

With exception for extension 1113, extensions 2034 and 1105 together have 1572 calls, representing more calls than all other extensions combined (1058 calls).

Extension 1107 is not referred before in Figure 1.3, as this is a very recent change to Figure 1.3 – Flowchart for the IVR Academic Services costumer contact and ticketing. It's yet uncertain if this change is to persist or not, but it's reflected automatically in this study.

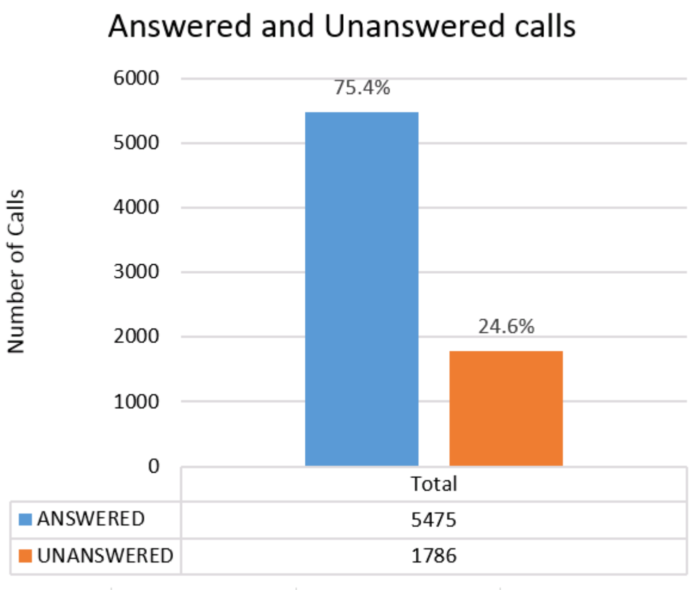


Figure 4.5 - Calls distribution by answered and unanswered type.

One major concern is how many calls are being answered and how many calls are being lost, as this might indicate the performance of the entire IVR service. Figure 4.5 indicates the distribution of calls by answered and unanswered type. From a total of 7261 calls, 24.6% are unanswered, also interesting is how answered and unanswered calls are distributed per IVR option (Figure 4.6).

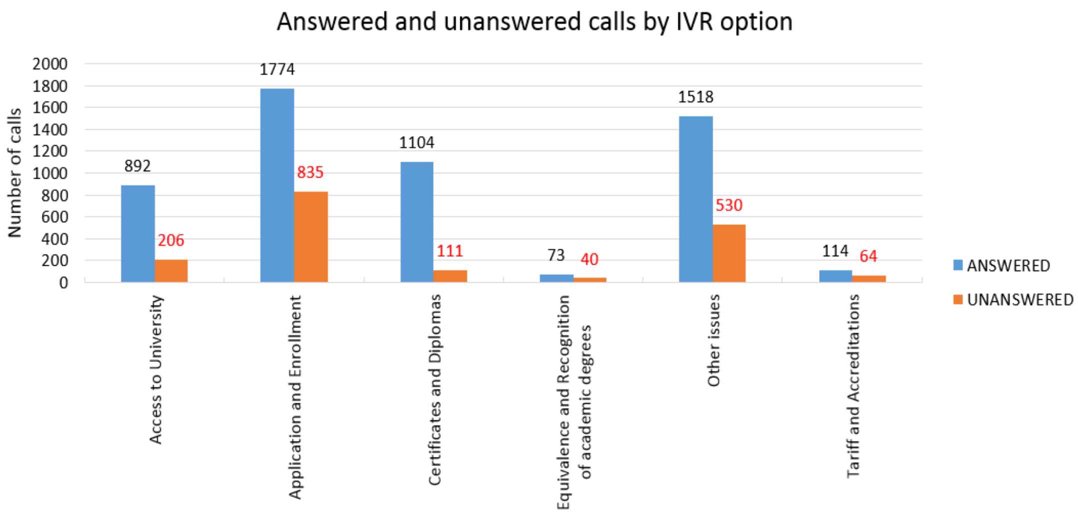
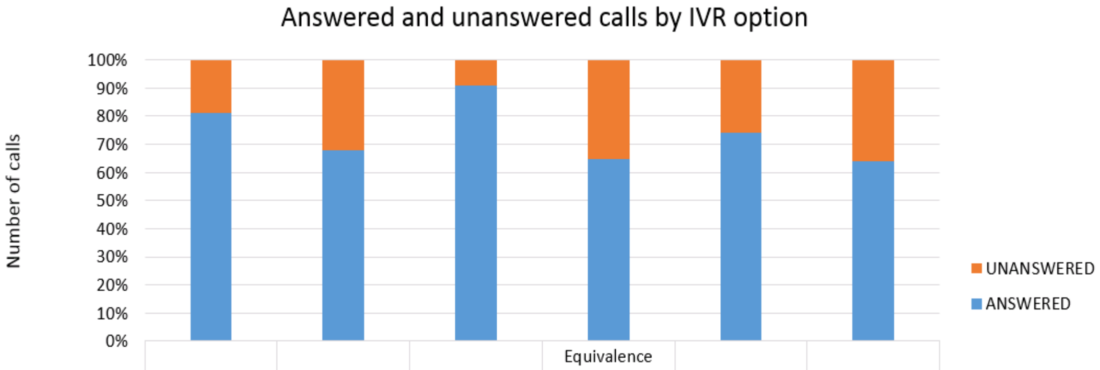


Figure 4.6 - Answered and unanswered calls distributed by IVR option.

Figure 4.6 indicates the number of unanswered calls per IVR option, indicating exactly 206 and 835 unanswered calls in options “Access to University” and “Application and Enrollment”, respectively. These two options combined represent the possible interest of students to become UBI alumni, and have a total of 1041 unanswered calls.

Option “Equivalence and Recognition of academic degrees” has the lowest number of unanswered calls. This doesn’t mean that it has the best rate of answered calls, as it’s possible to confirm in Figure 4.7.



	Access to University	Application and Enrollment	Certificates and Diplomas	Equivalence and Recognition of academic degrees	Other issues	Tariff and Accreditations
UNANSWERED	206	835	111	40	530	64
ANSWERED	892	1774	1104	73	1518	114

Figure 4.7 - Answered and unanswered calls distributed by IVR option, a different perspective.

According to Figure 4.7, option “Equivalence and Recognition of academic degrees”, had a total a 73 calls with nearly 35.4% lost calls. This was only surpassed by “Tariff and Accreditation” option with nearly 35.96% of unanswered calls. “Certificates and Diplomas” present the best rate of answered calls, only leaving about 9.14% of unanswered calls.

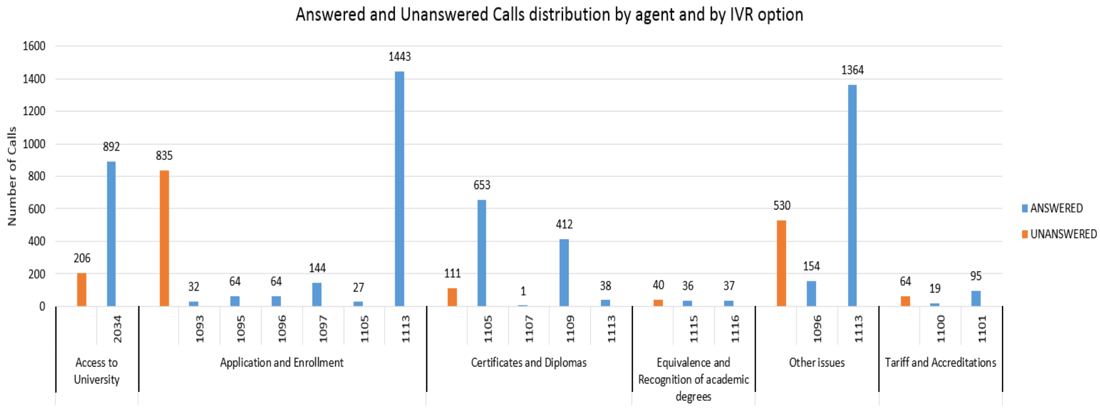


Figure 4.8 - Calls distribution by IVR option and by operator.

Figure 4.8 shows an interesting fact, there are two operators responding to IVR option “Other issues”. This is because in the first days the IVR was implemented there were actually two operators for option 5 of the IVR menu, which was changed afterwards. It also suggests that option “Access to University” and “Other issues” should have at least one more agent to answer calls, this will help to balance work load and will function has a redundancy if one of those agents is out. Figure 4.8 indicates a big variation on agents answering calls for option “Application and Enrollment”, *i.e.*, there are agents taking a very small amount of calls. This is furthered detailed in Figure 4.9.

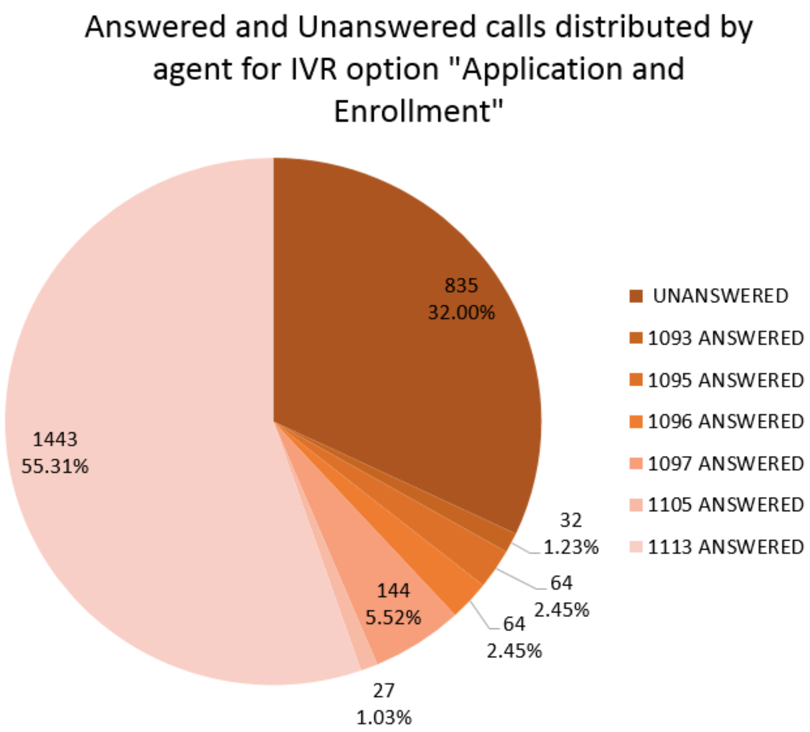


Figure 4.9 - Answered and Unanswered calls distributed by agent for IVR option "Application and Enrollment"

The unanswered calls represents 32% of calls received by IVR option “Application and Enrollment”. Extension 1113 takes about 55.31% of total calls received by this option.

Figure 4.9 shows that the combined five agents taking fewer calls represent less than 13% of the total calls, with four of those operators having individually less than 3% of all calls for this option. The unanswered calls account for more than the double of those five agents answered calls.

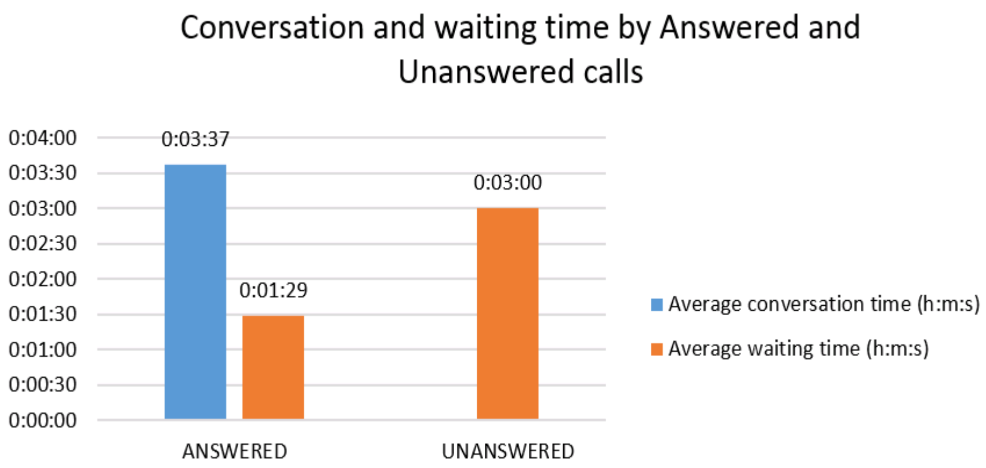


Figure 4.10 - Average conversation and average waiting time by Answered and Unanswered calls.

Figure 4.10 indicates that average waiting time for answered calls is about 1 minute and 29 seconds. The average call length is 3 minutes and 37 seconds. The average frustration time before a client ends the call before an agent add the chance to pick it up is about 3 minutes.

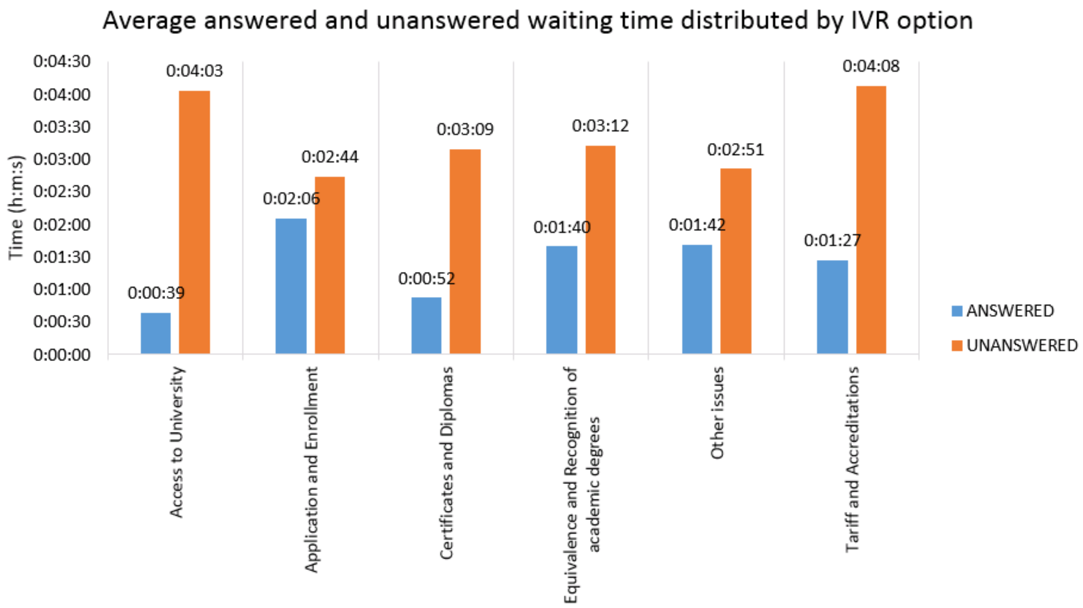
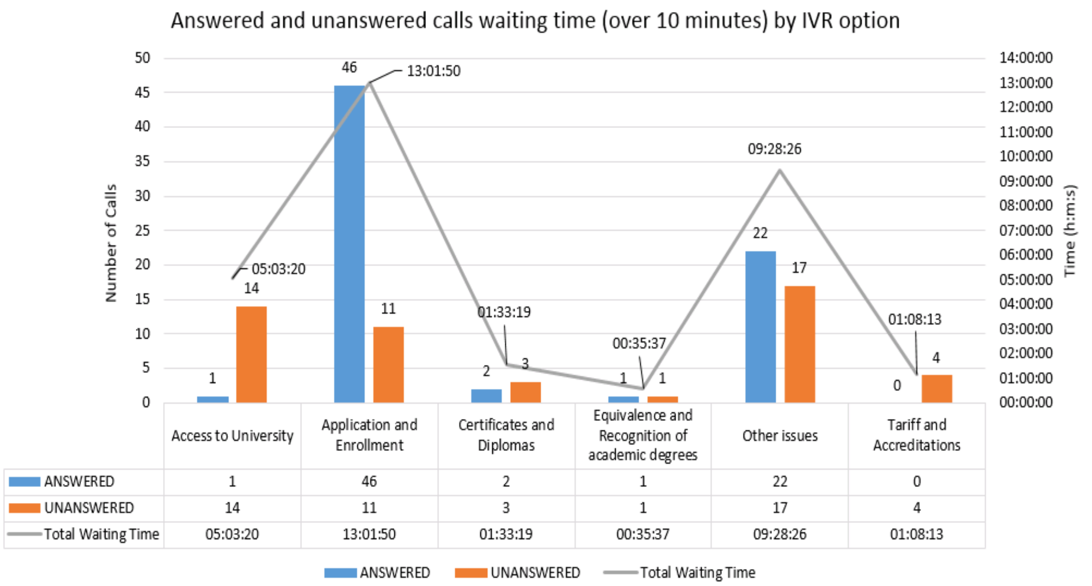


Figure 4.11 - Average answered and unanswered waiting time distributed by IVR option.

The waiting time before an agent picks up the call varies according to IVR option. For answered calls, Figure 4.11 shows less waiting time for "Access to University" option with an average of 39 seconds and greater waiting time for "Application and Enrollment" with an average of 2 minutes and 6 seconds.

Average frustration time also differs according to selected IVR option. This shows that clients are more willing to wait for option "Access to University" with 4 minutes and 3 seconds and "Tariff and Accreditations" with and 4 minutes and 8 seconds. Clients are less willing to wait for "Application and Enrollment" with 2 minutes and 44 seconds.



	Access to University	Application and Enrollment	Certificates and Diplomas	Equivalence and Recognition of academic degrees	Other issues	Tariff and Accreditations
ANSWERED	1	46	2	1	22	0
UNANSWERED	14	11	3	1	17	4
Total Waiting Time	05:03:20	13:01:50	01:33:19	00:35:37	09:28:26	01:08:13

Figure 4.12 - Answered and unanswered calls waiting time (over 10 minutes) by IVR option.

Figure 4.12 shows the number of calls over 10 minutes for answered and unanswered groups. It also presents cumulative waiting time for those calls per IVR option.

"Application and Enrollment" has 46 answered calls with more than 10 minutes, the answered and unanswered calls for this option sum 13 hours 1 minute and 50 seconds of waiting. "Other issues" option has 17 unanswered calls with more than 10 minutes. The answered and unanswered calls with more than 10 minutes for this option, represent 9 hours 28 minutes and 26 seconds of waiting time.

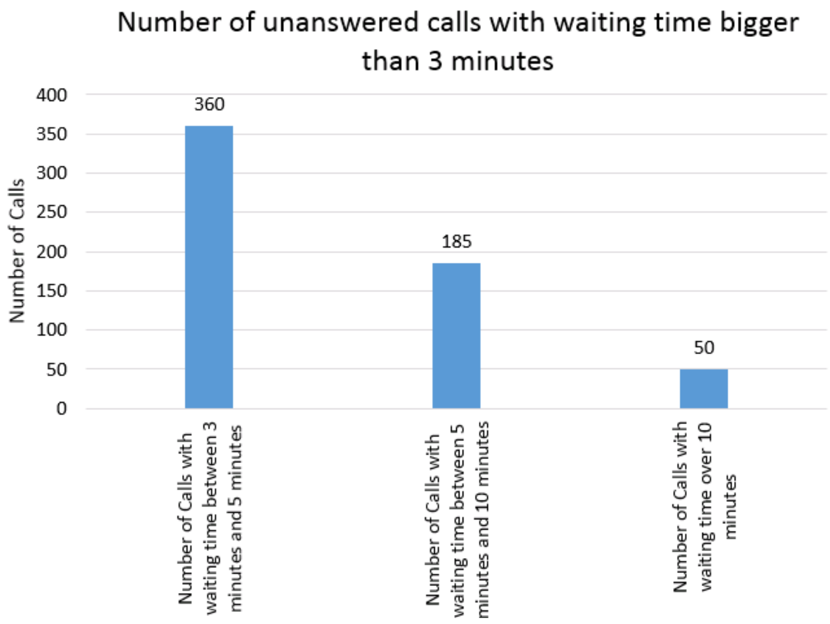


Figure 4.13 - Number of unanswered calls with waiting time bigger than 3 minutes.

According to Figure 4.13 there are 20.16% of calls lost from unanswered calls, for waiting times between for 3 minutes and 5 minutes. For this study 3 minutes is the average waiting time before frustration takes place. For waiting times between 5 minutes and 10 minutes there are 185 calls and represents about 10.35% of unanswered calls.

There are 50 unanswered calls with more than 10 minutes of waiting time, which represent 0.689% of total calls (answered and unanswered) and only 2.8% of unanswered calls.

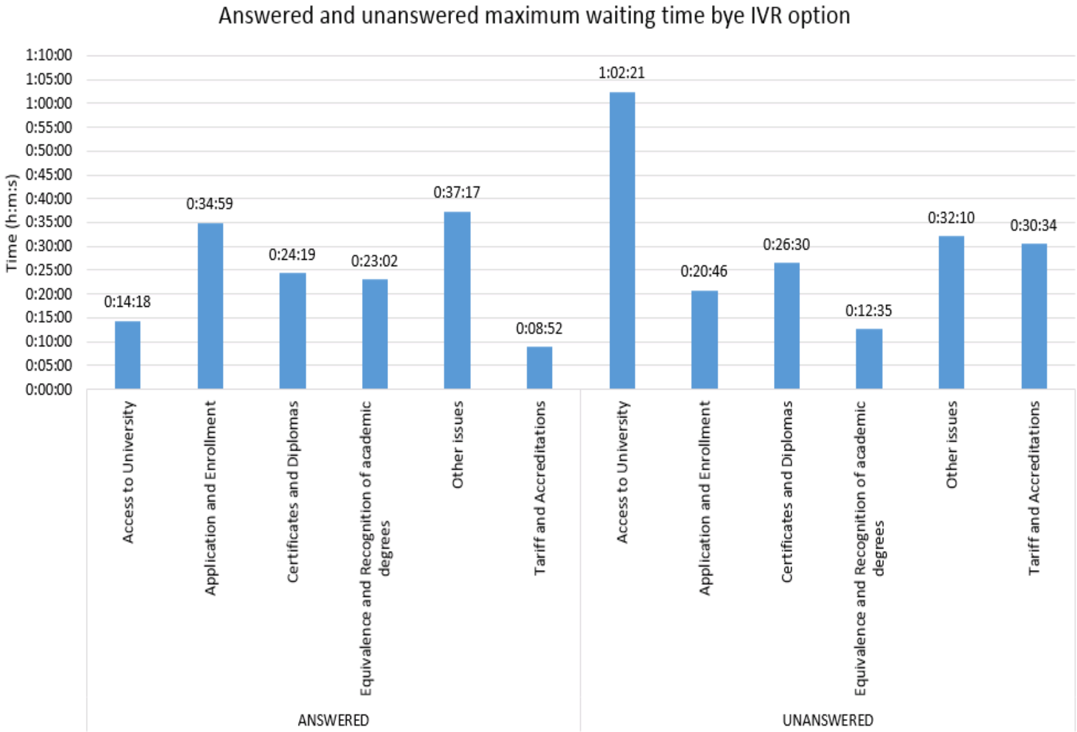


Figure 4.14 – Answered and unanswered maximum waiting time bye IVR option.

Figure 4.14 shows the maximum waiting times ever recorded during this study. For answered calls the best time goes to “Tariff and Accreditation” option, while the worst goes to “Other issues”.

For unanswered calls there as one call willing to wait for 1 hour 2 minutes and 21 seconds before giving up in option “Access to University”.

The minimum waiting time for answered calls vary from 3 seconds to 11 seconds.

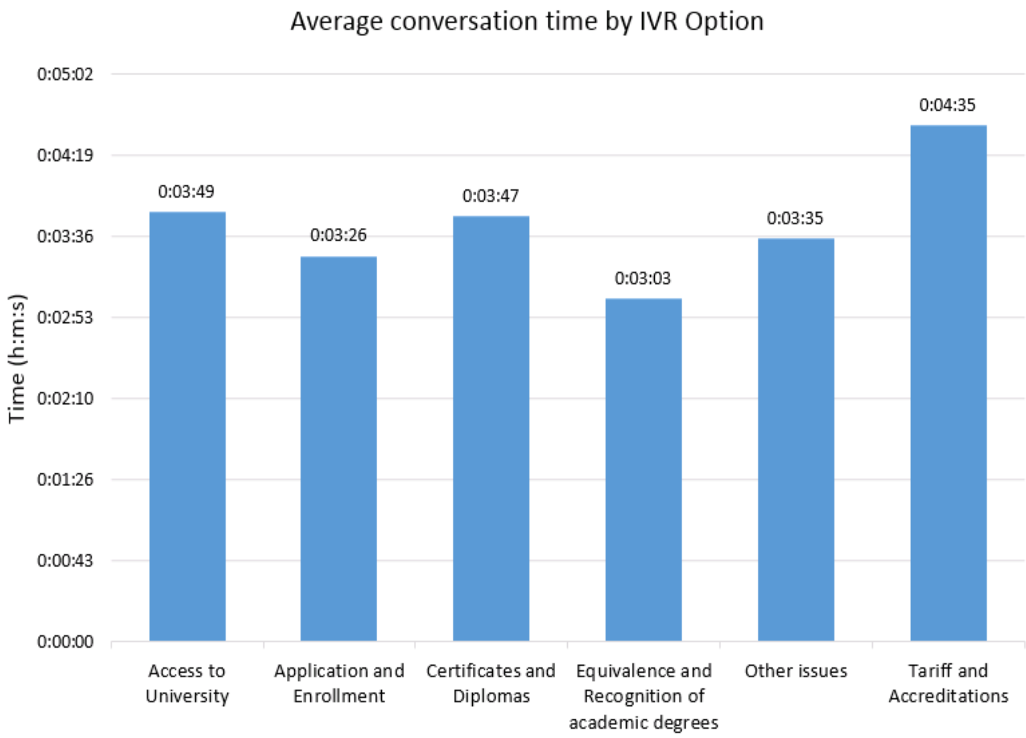


Figure 4.15 - Average conversation time by IVR option.

The average conversation time presents the smaller value for "Equivalence and Recognition of academic degrees" option. The biggest average value is for "Tariff and Accreditations".

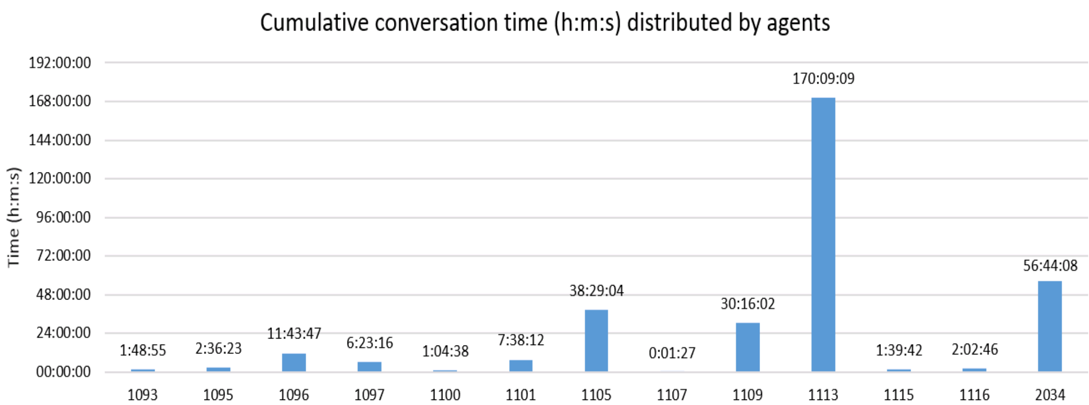


Figure 4.16 - Cumulative conversation time distributed by agents.

Figure 4.16 shows the cumulative conversation time by every agent on the IVR option menu. It is no big surprise that extension 1113 is the one with more time spent on the phone, since it was the one that answered to more calls, with 170 hours 9 minutes and 9 seconds of conversation time.

Total conversation time from all agents is 330 hours 37 minutes and 29 seconds. Putting this value to perspective, it means 41 days 3 hours 37 minutes and 29 seconds of one person workdays doing nothing more than talking to the phone.

The following six graphics are presented separately because they present very different characteristics, according to IVR menu option.

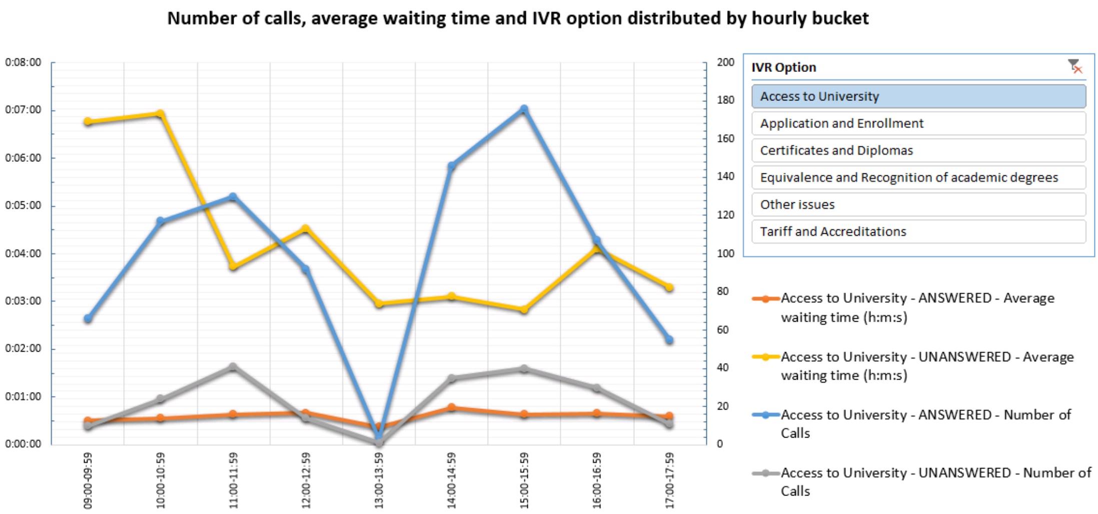


Figure 4.17 – Number of calls, average waiting time and IVR option “Access to University” distributed by hourly bucket.

From Figure 4.17 it is perceived that there are two periods with more call inflows. One period in the morning between 10h and 12h59m, the other period refers to the afternoon period between 14h and 16h59m. The answered average waiting time seems to be unaffected by peak hours.

Interesting is the fact that it is in the morning, between 9h and 10h59m, where clients are willing to wait more time before hang-up the call.

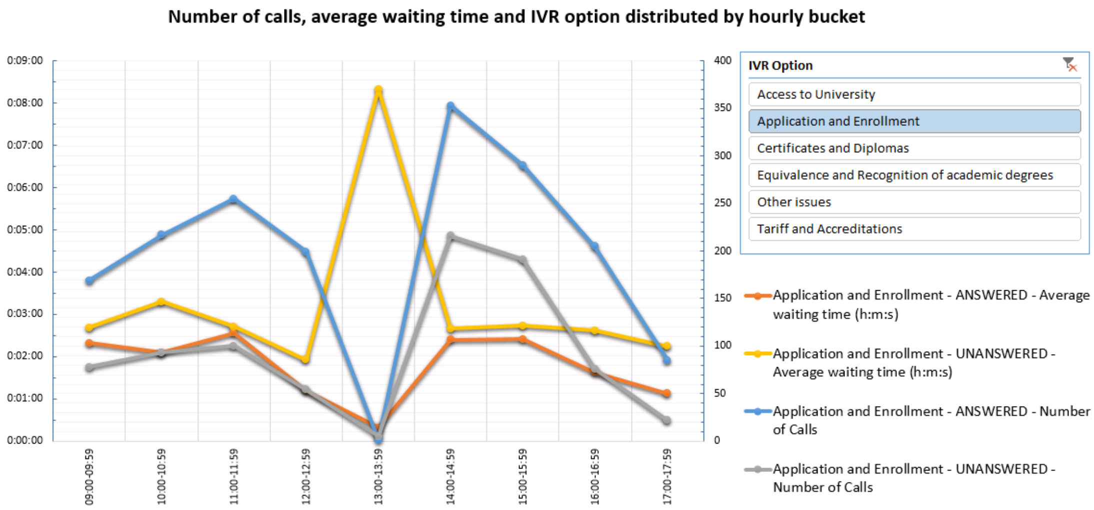


Figure 4.18 – Number of calls, average waiting time and IVR option "Application and Enrollment" distributed by hourly bucket.

Figure 4.18 presents two peak periods for receiving calls, one in the morning between 10h and 12h59m and one bigger in the afternoon from 14h to 16h59m. Answered average waiting time is affected by peak hours and clients are more willing to wait in the period 13h to 13h59m.

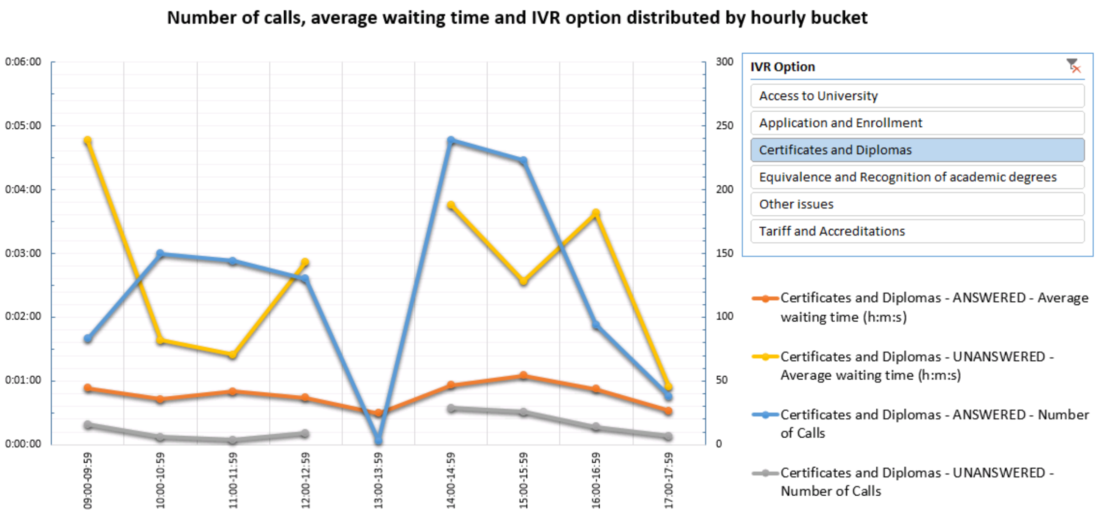


Figure 4.19 – Number of calls, average waiting time and IVR option "Certificates and Diplomas" distributed by hourly bucket.

Figure 4.19 presents two periods with more call inflows. One period in the morning between 10h and 12h59m, the other period refers to the afternoon period between 14h and 15h59m. The answered average waiting time has a slight inflection in peak hours. Interesting is the fact that unanswered number of calls doesn't follow the peak of answered calls in the morning period.

The periods where clients are willing to wait more are from 9h to 9h59m, 12h to 12h59m and 14h to 16h59m.

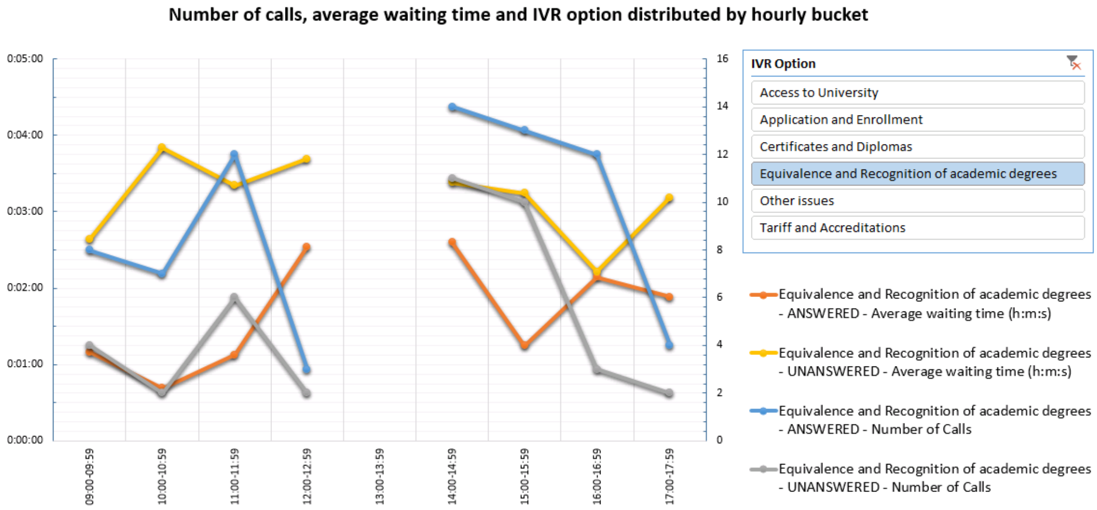


Figure 4.20 - Number of calls, average waiting time and IVR option "Equivalence and Recognition of academic degrees" distributed by hourly bucket.

Figure 4.20 presents two peak periods for receiving calls, one is in the morning between 11h and 11h59m and the other one in the afternoon from 14h to 16h59m. Answered average waiting time appears to have no relation to peak hours. Unanswered calls follow the trend of answered calls.

Clients are more willing to wait from 10h to 12h59m, from 14h to 15h59m and from 17h to 17h59m.

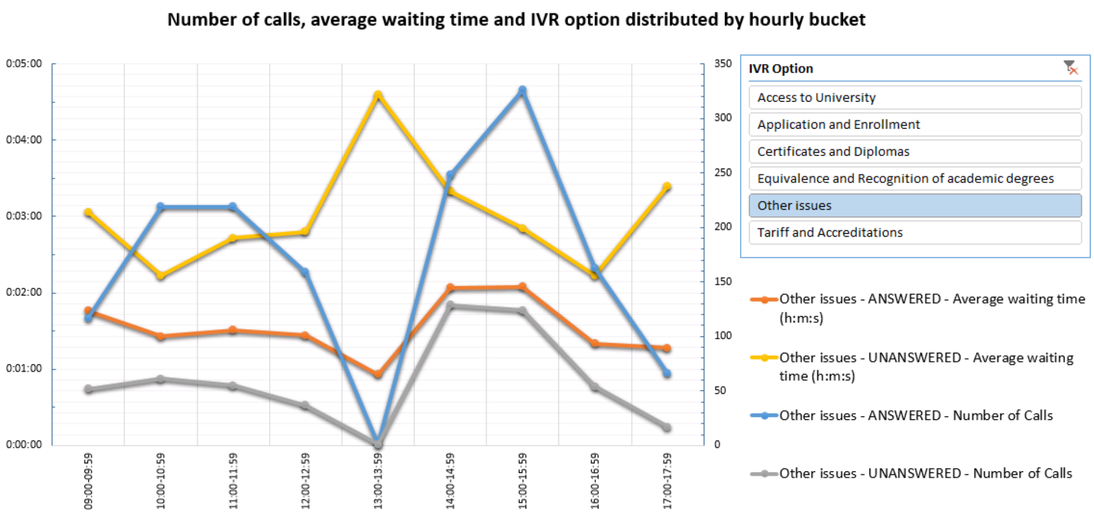


Figure 4.21 - Number of calls, average waiting time and IVR option "Other issues" distributed by hourly bucket.

Figure 4.21 shows two periods with more incoming calls. One period in the morning between 10h and 11h59m, the other period refers to the afternoon period between 14h and 15h59m. The answered average waiting time appears to be affected by peak hours. The periods where clients are willing to wait more are from 13h to 13h59m.

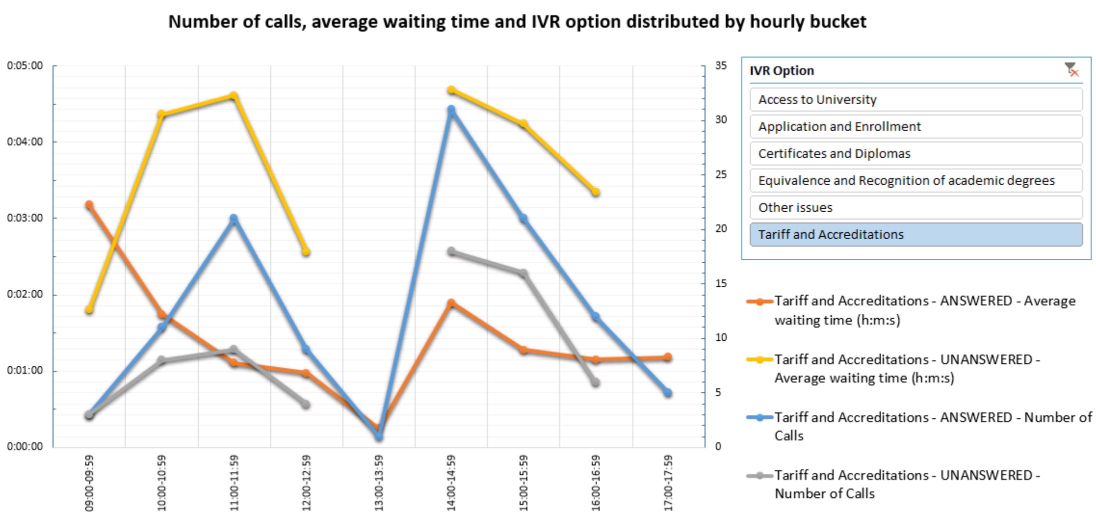


Figure 4.22 - Number of calls, average waiting time and IVR option "Tariff and Accreditations" distributed by hourly bucket.

Figure 4.22 presents two periods with more call inflows. One period in the morning between 11h and 11h59m, the other period refers to the afternoon period between 14h and 14h59m. The answered average waiting time appears affected in the afternoon period, but shows no relation to peak hours in the morning period. Unanswered number of calls follows the trend revealed by answered calls.

The periods where clients are willing to wait more are from 10h to 11h59m and from 14h to 15h59m.

From Figure 4.17, Figure 4.18, Figure 4.19, Figure 4.21 and Figure 4.22 it's possible to observe that there are some calls being answered after 13h. This overlaps with the lunch break for workers at the Academic Services. Figure 4.20 is the only one that doesn't present such fact. This happens, according to Figure 1.3, because IVR Academic Services costumer contact and ticketing is available until 13h. So a call entering the system near 13h will span over lunch break. As data suggests, a change in IVR availability hours should take place in a near future.

According to Figure 4.10 the average frustration time is about 3 minutes and in consonance to Figure 4.15, the longest average conversation time is 4 minutes and 35 seconds. The sum of this values is 7 minutes and 35 seconds. The IVR for Academic Services costumer contact and ticketing should stop taking new calls before 12 hours 52 minutes and 25 second.

These preliminary results will allow high level management to re-organize the human resources around the real workload as evidenced by the collected data, therefore allowing the external users to perceive a more efficient service, and the internal users to focus where the workload is increased.

Chapter 5

Conclusions and future work

Despite the massive spread of self-service Web applications, Gartner states that 92% of all customer transactions still take place over the telephone. This number is more likely to increase if the destination is a toll-free number [23].

Many systems that use IVRs have a very bad reputation, in many cases well deserved, although it's a precious tool for many call/contact centers. Well-designed IVRs can provide the caller the most skilled agent to handle his/her matter at the first contact.

Poorly designed they can become a source of frustration and annoyance, one good example of this is the call being routed to someone that can't help the caller. Even worst is the call being transferred over and over again by agents that still can't help. It's not the IVR systems that customers hate, it's the way they are configured, as the technology itself is completely neutral.

IVR is usually the first touch-point most consumers have. It either facilitates positive experiences or negative experiences. Many information technology administrators configure these systems and then leave them to be like a self sustaining system. This was already established to be a big mistake, with many enterprises loosing clients, as research proves it.

An IVR is like any piece of software or hardware, it needs maintenance and monitoring. Maintenance might be from a software/hardware perspective, *i.e.*:

- Are all options well configured and routed to at least one operator?
- Are equipment's still handling traffic volume?
- Do logs reveal any malfunction?
- How is sound quality?

These are very important questions, but aren't the only ones. From a business intelligence perspective there are still questions to be answered, for example:

- How many calls the system gets?
- Are menu options still relevant?
- Are they in the correct order (most required in first place)?
- Do the options with most hits have the correct number of workers?
- What is the percentage of answered/unanswered calls?
- Are menu options intuitive and the wording clear?
- What is the average frustration time?
- Do peak times have more agents to answer calls?

In monitoring, it's possible to identify positive and negative aspects. IVR systems are like any other system, they need maintenance and monitoring. In a quality culture, an important insight is that failure or success is always a value formed after certain facts.

Treating large amounts of data and extract concise information in a timely manner from it it's vital nowadays. Iterate, monitor, and iterate in a constant cycle will guarantee the best decisions based on current experience, information and understanding, so that the IVR continue running efficiently and be an effective tool for both users and agents.

The suggested a multidimensional data analysis model using data cube extracts comprehensive analysis results that fits the two proposed assumptions: maintain comprehensive historical data and service monitoring.

The views presented in the results section are representative of the full power of this analysis, as the IVRCube makes it possible to create all types of views on the data by gathering and filtering the various dimensions.

The system is used to conduct a multidimensional analysis through an OLAP operation according to the level of abstraction for each dimension. Analysis system using data cube can easily and timely extract comprehensive analysis results showing relevant trends. It can be used as a decision support system for business intelligence. A properly optimized IVR can lead to higher rates of satisfaction while lowering costs. The continuous improvement provides the ability to find potential enhancements that may otherwise go unnoticed.

5.1. Future Work

The ultimate objective is to create a dashboard for the University's high level management, so the information that the system provides can be effectively and timely used to make correct management decisions, namely by allowing the re-allocation of resources considering date, time and workload constraints.

Efficiency and productivity are key aspects in network and telecommunication systems, therefore in the long term, I foresee many opportunities to apply analytical technologies as the one presented in this work, giving me purpose and motivation for future developments.

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Appendices

Paper:

A comprehensive IVR (Interactive Voice Response) analysis model using online analytical processing (OLAP) on a multidimensional data cube

A comprehensive IVR (Interactive Voice Response) analysis model using online analytical processing (OLAP) on a multidimensional data cube

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Abstract

Private Branch eXchange (PBX) is a tool indispensable in the business world. The telephone exchanges allows employees to perform internal connections between telephones, or make calls to the external network (PSTN) also known as Public Switched Telephone Network. With increasing Internet usage, there is interest in understanding what services are offered. FreePBX is an open source Asterisk web-based PBX software for Linux, which supports multiple protocols and services, like Interactive Voice Response (IVR). Cisco Unified Communications Manager (CUCM) or CallManager, is a software-based call-processing system developed by Cisco Systems. CUCM tracks all active VoIP network components; including phones, gateways, conference bridges, among others. IVR is part of the Academic Services costumer contact and ticketing of University of Beira Interior (UBI). IVR monitoring and analysis are essential for effective operation and resource management, in particular, multidimensional analysis for long-term data is necessary for comprehensive understanding of the trend, the quality of customer service and costumer experience. In this paper, we propose a new IVR analysis model for large volumes of IVR data accumulated over a long period of time. The IVRCube proposed is an analysis model using online analytical processing (OLAP) on a multidimensional data cube that provides an easy and fast way to construct a multidimensional IVR analysis system for comprehensive and detailed analysis of long-term data. The feasibility and applicability are validate, as the proposed IVRCube analysis model is implemented and applied it to Academic Services costumer contact and ticketing IVR data.

I. Introduction:

University of Beira Interior (UBI) [1] started the convergence of the phone system and the data network in 2005, after acquiring a commercial Internet Protocol (IP) Private Branch Exchange (PBX) software from Cisco called CallManager [2]. The long term UBI objective, was to have 100% Voice over IP (VoIP) network until 2020. The Internet Protocol Private Branch eXchange (IP PBX) was integrated with the analogical PBX's and with the Public switched telephone network (PSTN). Figure 1 presents a scheme depicting CallManager integration in the UBI voice architecture. In Figure 1 E-carrier (E1) represents a Primary Rate Interface (PRI). PRI is a standardized telecommunications service level within the Integrated Services Digital Network (ISDN). E1 provides 30 communication channels and 2 signaling channels, this is also referred to as 30B channels plus 2D channels. In this case Media Gateway has 60 voice channels to interconnect VoIP and legacy telephony.

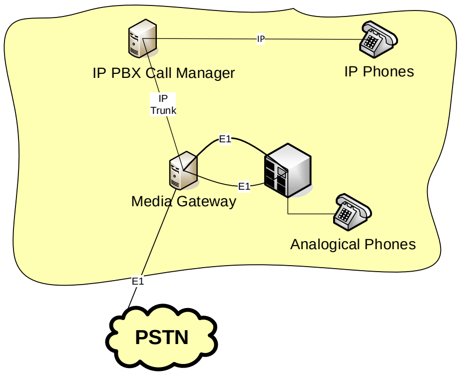

Figure 1 – Scheme depicting CallManager Integration.

In 2007, Portuguese Universities started to transform their phone systems in favor of a converged system using the data network to route phone calls. This project, named VoIP@RCTS [3] was launched by the *Fundação para a Computação Científica Nacional (FCCN)*, the Portuguese National Research and Education Network (NREN). The chosen solution was a software platform based on FreePBX [4] witch uses the Asterisk core [5]. This software although it's based on FreePBX it has been redesigned to suite Universities specifications. This new software is called Enterprise Courier from itCenter [6]. UBI integrated that same project called VoIP@RCTS and incorporated in its telephone system. Figure 2 illustrates UBI integration of VoIP@RCTS in the existing infrastructure of UBI. In Figure 2 we can also see the Session Based Controller (SBC) that acts as a firewall for the VoIP world.

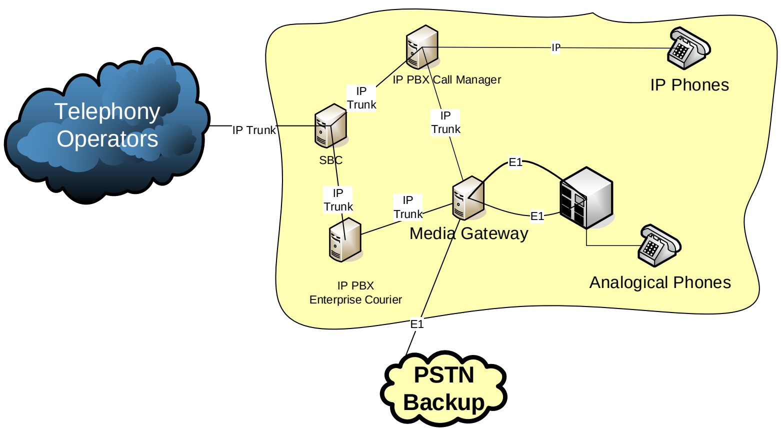

Figure 2 – Integration of VoIP@RCTS in UBI infrastructure.

Although the provided solution was based on open source software, it was customized for the specific needs of the universities, therefore, Enterprise Courier offers new possibilities to explore. One of the offered possibilities is the use of IVR, which suited some of the specific services, by allowing the calling user select the most adequate operator for his/her particular call motivation.

The actual flow diagram for the IVR in study (Academic Services costumer contact and ticketing) is showed in Figure 3.

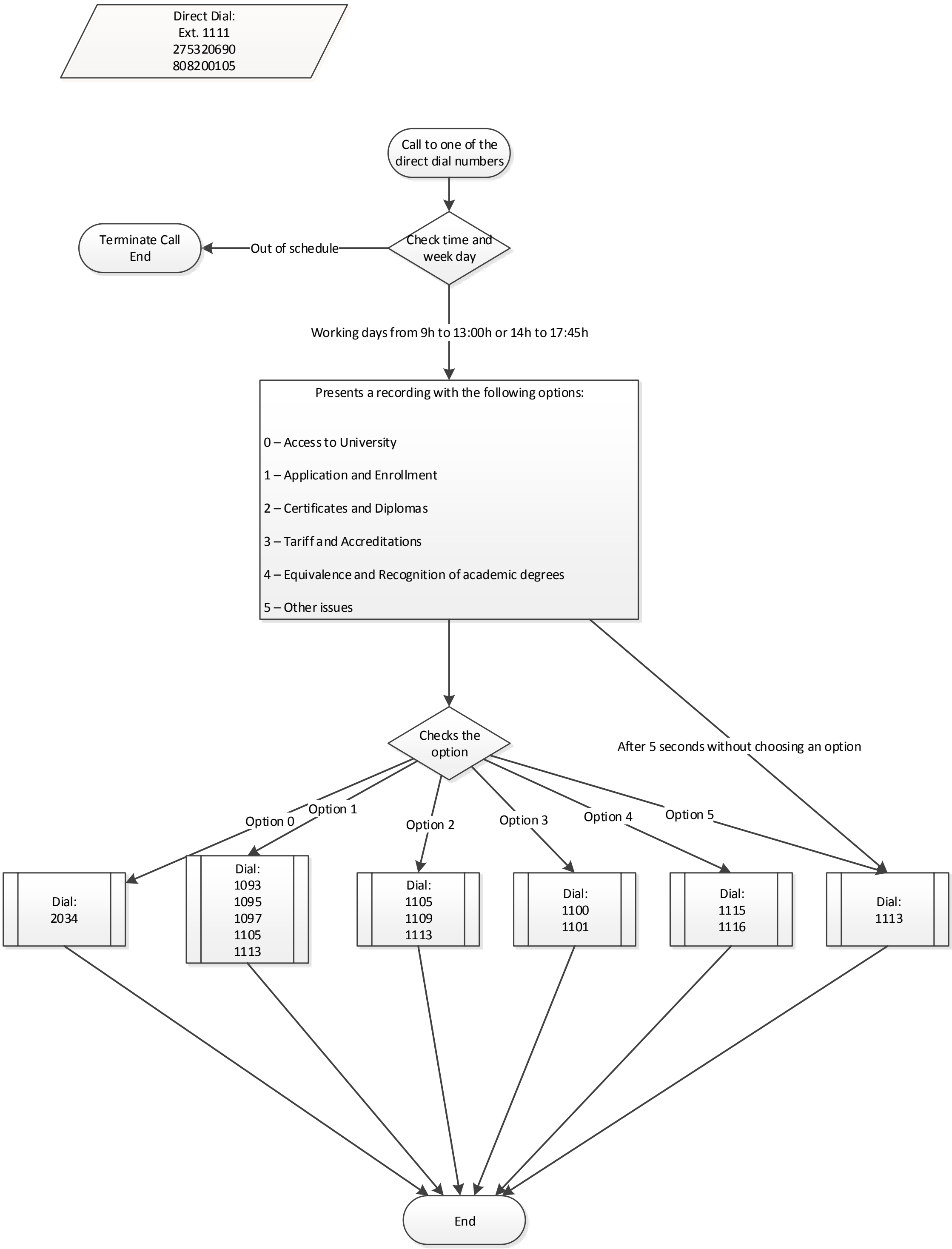

Figure 3 – Flowchart for the IVR Academic Services costumer contact and ticketing.

UBI seeks to continuously improve its organizational structure, its processes and its control methods, aiming to satisfy and anticipate the requirements of its customers and other stakeholders.

Yet, the Enterprise Courier does not provide IVR statistics, and therefore, there are no metrics that allow UBI to evaluate and plan improvements in the area of automated telephone response.

Therefore, in 2013, the Vice-Rector for Academic Affairs, recognized as the main stakeholder, required that monitoring and supervision would need to be implemented in the Academic Services costumer contact and ticketing IVR system.

Having this in consideration, this paper describes the research carried out that will allow the future creation of a solution that allows monitoring and supervision, for example, get a relationship between how many calls an option receives and which operator takes it, or how many minutes does a call last, or how many seconds is a client on hold until he starts to talk with the operator, on an Asterisk based software.

The remainder of this paper is organized as follows: this paragraph concludes section 1, the Introduction, where the problem, motivations and objectives of the research were outlined; section 2 follows, describing the state of the art; section 3 presents the architecture and the solution; section 4 describes and discusses the results obtained; and finally section 5 concludes de the paper presenting conclusions and future work.

II. State of the Art:

Modern networks are constantly evolving to meet user demands. Initial data networks were limited to exchanging character-based information between connected computer systems. Traditional telephone, radio, and television networks were maintained separately from data networks. In the past, every one of these services required a dedicated network, with different communication channels and different technologies to carry a particular communication signal. Each service had its own set of rules and standards to ensure successful communication.

Advances in technology are enabling us to consolidate these different kinds of networks onto one platform referred to as the "converged network". Unlike dedicated networks, converged networks are capable of delivering voice, video streams, text, and graphics between many different types of devices over the same communication channel and network structure. Previously separate and distinct communication forms have converged into a common platform [7].

Compared to traditional telephony (circuit switching), VoIP is now more scalable, allows the integration of new features and the possibility of reducing costs. From the perspective of data networks, VoIP has transform the voice into just another application [5]. In particular some VoIP applications allow IVR configuration. IVR allows call handling in order to give some sort of response to them without human interaction. Being the caller responsible to choose one of the options that are presented. IVR is usually presented in a form of a menu that can have many layers or submenus and are composed by audio files played to the caller, witch, through touch keys on the phone can interact with the system, for example, allowing the creation of a voting system via phone, money transfer from a bank account or to choose the more skilled operator in certain subject [8].

Some VoIP applications have add-on tools that allow to create IVR processes using drag and drop. In this category there at least three popular solutions, one from Alcatel Lucent called OmniTouch

Contact Center [9] and the other from Cisco called Unified Contact Center Express [2]. Both are commercial, have various license options according to the desired functionality range and integrate their respective unified VoIP solutions. On the other hand, in the open source world we have Asterisk [5] or Asterisk based systems, where IVRs are abstractions created from fragments of the dial plan and have no special treatment.

The commercial solution from Alcatel Lucent also has built-in statistics reports. According to their documentation, this product has a range of reports of real-time details on service activity, individual agent, agent group activity and statistics on each type of interaction [9].

The Cisco platform has one add-on tool called Cisco Unified IP IVR that has this functionality. This tool can create reports for application performance analysis report, detailed call by call report and traffic analysis report [2].

In Asterisk based systems, CDR-stats [10] is a separate application that analyses the Call Detail Records (CDR) and creates statistics from there. It's not by any mean IVR specialized, but is one interesting tool for CDR reporting. There are two options: professional support (commercial version) or community support (free to use). QueueMetrics [11] is specialized for call centers and has a special treatment for IVR, it can be integrated with some Asterisk based systems and it's a commercial solution with a license renewal every four years. This software creates reports for total calls, answered calls, unanswered calls, area code breakdown, inbound call attempts, Call distribution, Agents, Outcomes and Call detail.

III. Architecture and solution to the problem: A comprehensive IVR analysis model using online analytical processing on a multidimensional data cube:

Responding to the constant need for improvement stated in UBI's 2020 mission and previously presented in the introduction section, a system analysis approach was used to define the solution development process.

The identified relevant stakeholders were, primarily, the Vice-Rector for the Academic Affairs, and also, too much lesser extent, the users at the Academic Services, and external users that contact UBI using the telephone.

Based on the interviews conducted with the stakeholders, the following list of functional requirements (FR) and non-functional requirements (NFR) was produced for the answered calls:

FR1: The system must be able to provide metrics on the efficiency of the incoming calls, *e.g.*, the system must provide information about the answered calls per day or set of consecutive days, month or set of consecutive months, quarter or set of consecutive months and year or set of consecutive years;

FR2: The system must be able to provide information about the answered calls per operator number; FR3: The system must be able to provide information about the distribution of the incoming calls, *i.e.* the system must provide information about the answered calls per IVR menu option;

FR4: The system must be able to provide information about the time spent, *i.e.* the system must provide information about the waiting time and conversation time.

NRF1: The system must not install any software in the IP PBX Enterprise Courier.

NRF2: The system must not interfere with the IP PBX system, *e.g.* the system must not alter the IP PBX databases.

NRF3: The system must not interfere with the VoIP performance.

It must be noted that this list of requirements is not final, as the approach to the development of the system was performed in a prototyping manner, thus allowing the stakeholders to refine the requirements list, *i.e.*, the results that the system must output will be subject of continuous improvement.

The IP PBX Enterprise Courier stores locally, in a MySQL database, information about all calls in a format named Call Detail Record (CDR) and in particular, calls that enters into the IVR. The structure of the CDR has the following fields: `calldate`, `clid`, `src`, `dst`, `dcontext`, `channel`, `dstchannel`, `lastapp`, `lastdata`, `duration`, `billsec`, `disposition`, `amaflags`, `accountcode`, `uniqueid` and `userfield`. From this list of fields we have special interest in: `calldate`, `src`, `dst`, `duration`, `billsec` and `uniqueid`. The field `calldate` has the timestamp for the call beginning and is a datetime type. The field `src` has the call source, *e.g.* "+351275319700"and is a varchar type. The field `dst` has the call destination, *e.g.* "1113" and is a varchar type. The field duration has the call total time and it is of int type. The field `billsec` has the call conversation time and it is of int type. The field `uniqueid` has the id for the call in question and is a varchar type.

The relevant data fields [12] for this study are retrieved from Enterprise Courier IP PBX by the data warehouse and are stored for comprehensive and detailed analysis as it is showed in Figure 4. The data warehouse retrieval SQL job, fetches new data and runs twice a day, but it can be adjusted for more sensitive real-time data, if required.

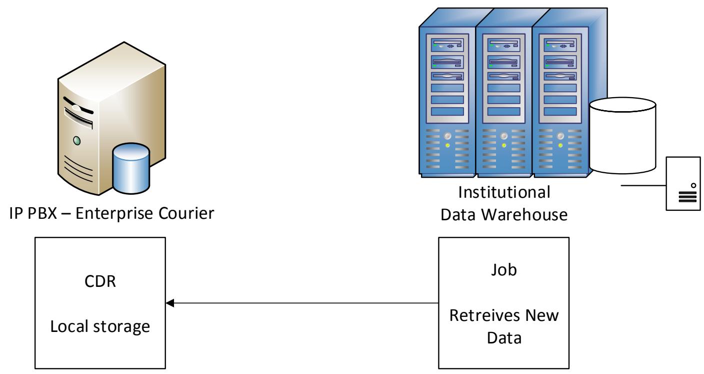

Figure 4 – Data retrieval scheme.

With the imported data, a multidimensional data model is constructed at the institutional data warehouse. This model views data in the form of a data cube [13] [14] [15]. A data cube allows data to be modeled, viewed and filtered in multiple dimensions. The cube is defined by dimensions and facts [16]. Dimension tables are integral companions to a fact table. The dimension tables contain the textual descriptors. Each dimension is defined by its single primary key, designated by the PK notation in Figure 5, which serves as the basis for referential integrity with any given fact table to which it is joined. Dimension table attributes play a vital role in the data warehouse. Since they are the source of virtually all interesting constraints. [17]

A fact table is the primary table in a dimensional model [16]. The fact table contains the names of the facts, or measures, in this case "wait_time" and "conversation_time", according to Figure 5. In addition to these measures, calculated measures where introduced based on this this two primary measures, MAX_wait_time as the maximum value for wait_time, MIN_wait_time as the minimum value for wait_time, Avg_wait_time as the average value for wait_time, MAX_conversation_time as the maximum value for conversation_time, MIN_conversation_time as the minimum value for conversation_time and Avg_conversation_time as the average value for conversation_time were introduced, as well foreign keys to each related dimension tables, designated by the FK notation in Figure 5. The fact table itself generally has its own primary key made up of a subset of the foreign keys. This key is often called a composite or concatenated key, in Figure 5 is showed as id_call. A data warehouse requires a concise, subject-oriented schema that facilitates online data analysis. A star schema [14] for the multidimensional analysis model, figure 5, with the fact table in the middle. As this schema was applied on IVR data, it was decided to call it IVRCube.

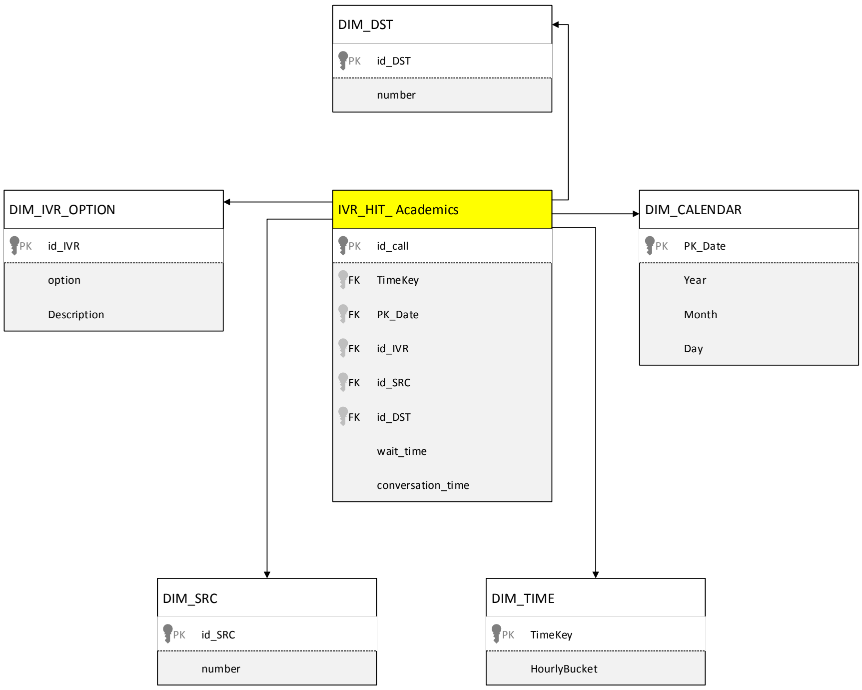

Figure 5 – Cube structure view.

In this paper, we defined five dimensions [18]: CALENDAR, TIME, DST (stands for destination), SRC (stands for source) and IVR_OPTION. The DST dimension has all possible destination numbers, in this study, it's the operator numbers. SRC dimension has all source numbers from the data source. SRC and DST dimensions has all sources and destinations respectively, from the data source, and are automatically populate in the retrieval job, Figure 4. The calendar dimension is the one dimension nearly guaranteed to be in every data mart, because virtually every data mart is a time series. Calendar Dimension was created with an SQL script and has all possible data from 2014 until 2020. TIME dimension gives a perspective of how hours of the day may influence the service (for example peak hours). TIME Dimension was created with an SQL script and has all possible hours from 00h until 24h, distributed in hourly bucket. The IVR_OPTION dimension has all possible options the clients can choose from the IVR menu. This was the only dimension that was manually populated with the names of the possible options of the IVR menu, e.g. "Access to University", according to Figure 3.

IV. Results

The IVR is running since late January of 2014 and the presented results span this period until late July 2014. In this study only answered calls will be analyzed. Results may be presented in table form, or as charts. In this section, some of the most relevant views from the multidimensional analysis are presented, describing views from the multidimensional analysis in the form of graphics.

In Figure 6 we can see the distribution of calls by the various IVR options.

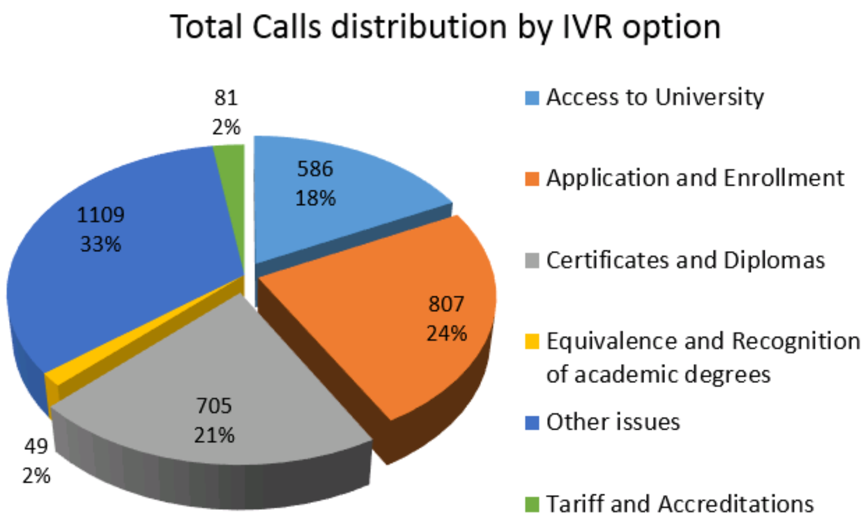

Figure 6 – Answered calls distribution of by IVR options.

The “Access to University” and “Application and Enrollment” options represent possible interest in the University and combined have a total of 42% of total calls, followed by the “Other issues” option that represents a total a 33% from a total of 3337 answered calls.

Figure 7 represents the distribution of calls during the period of this research.

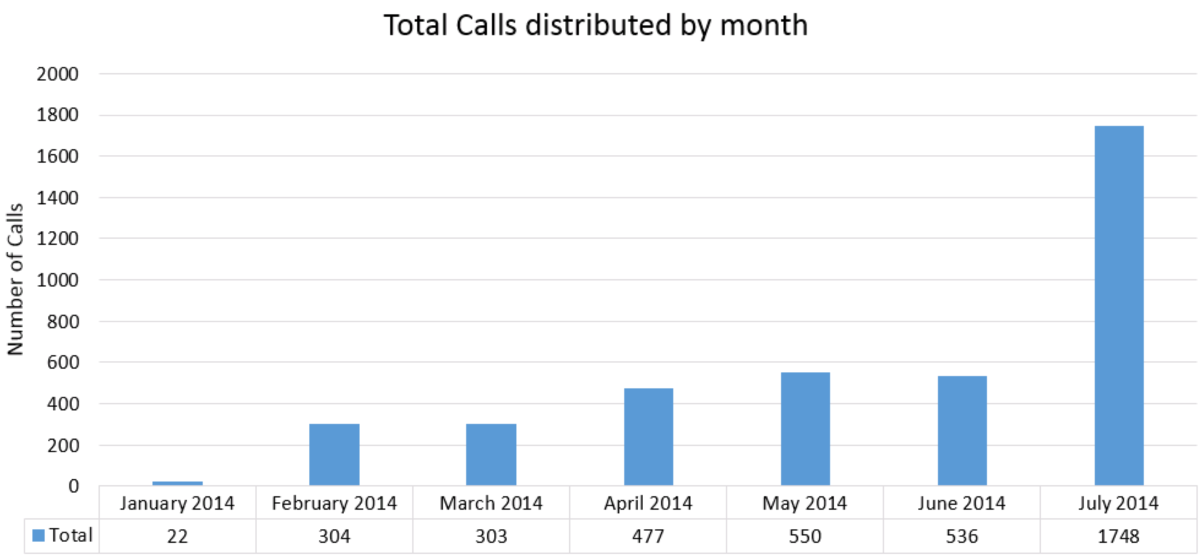

Figure 7 – Calls distributed by month.

In the monthly distribution, January had a very small number of calls, most likely because it was when the IVR service was created and it wasn't properly advertised. Calls volume doubles in July compared to previous months. In Figure 8, additional detail shows the evolution on the motivation for the calls in month of July and compares it to calls in the month of June.

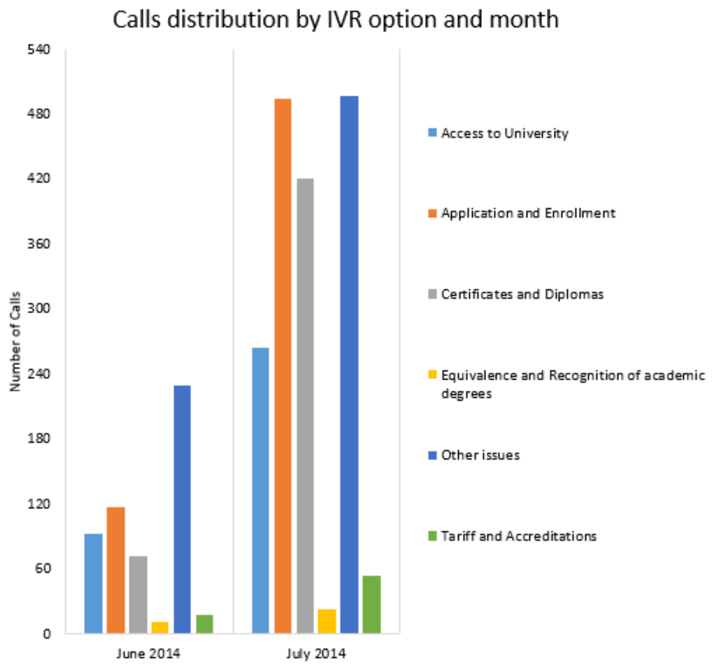

Figure 8 – Call details for June and July.

The month of July, Figure 8, shows a global increase in all call options. In particular the options: "Access to University", "Application and Enrollment". In Portugal, July is period of the year where students choose which University they want to enroll, which accounts for the increase in these options. The "Certificate and Diplomas" option also has a significant increase due to the school year end.

Figure 9 presents the distribution of calls per IVR operator.

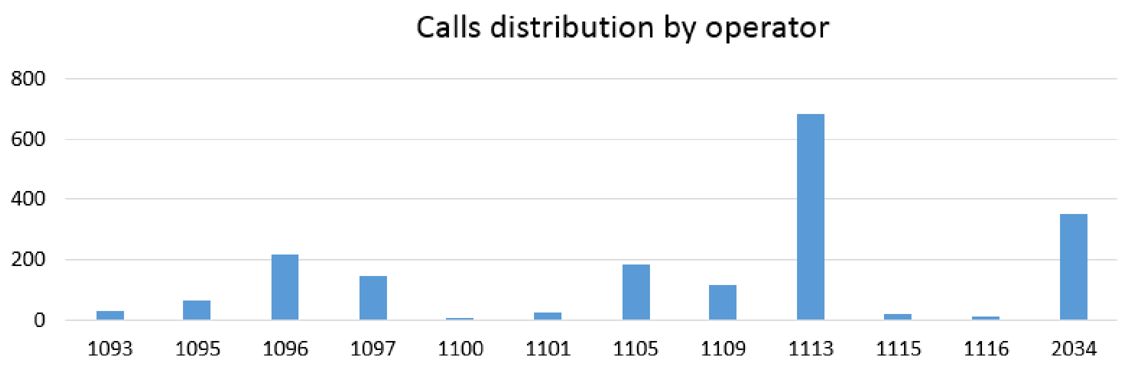

Figure 9 – Calls distribution by operator.

This figure shows all 3337 answered calls distributed by all IVR operators. The operator that as most of the calls is extension 1113, and according to Figure 3, this is perfectly acceptable since this operator responds to three options of IVR menu, as showed in the Figure 10.

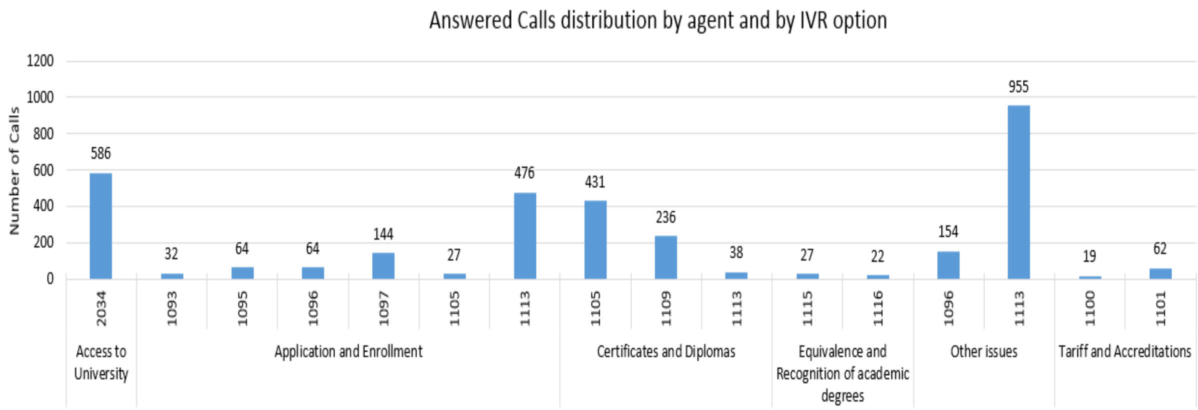

Figure 10 – Calls distribution by IVR option and by operator.

Figure 10 also shows an interesting fact, as there are two operators responding to IVR option “Other issues”. This is because in the first days the IVR was implemented there were actually two operators for option 5 of the IVR menu, which was changed afterwards.

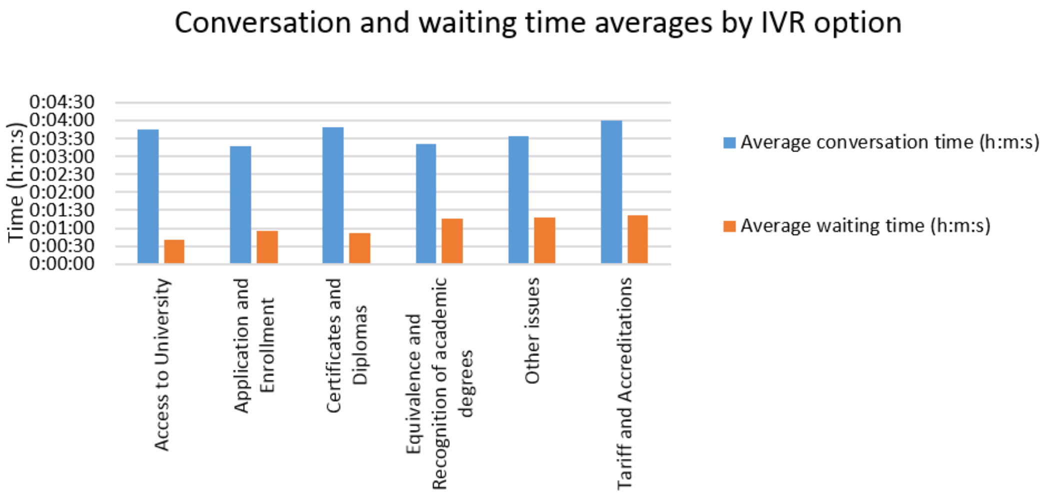

Figure 11 – Conversation and waiting time averages by IVR option.

Figure 11 presents the time, in minutes, users spend in the telephony system. It shows that the average conversation time spans from 3 minutes and 18 seconds in option “Application and Enrollment” to 4 minutes and 1 second in “Tariff and Accreditations” option. Figure 11 also shows that the average waiting time spans from 41 seconds in “Access to University” option to 1 minute and 21 seconds in “Tariff and Accreditations”.

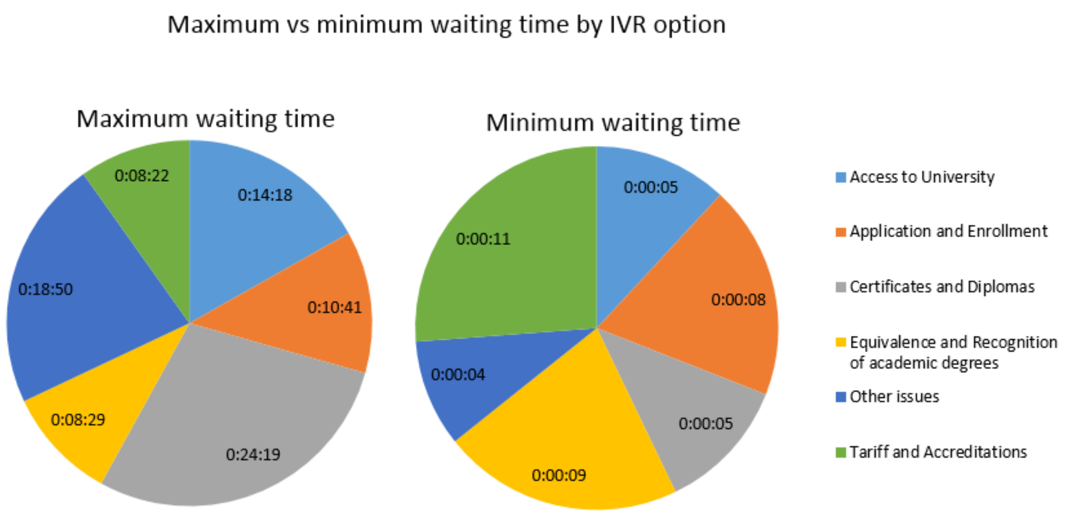

Figure 12 – Maximum vs minimum waiting time by IVR option.

According to Figure 12, the maximum waiting time spans from 8 minutes and 22 seconds in "Tariff and Accreditations" to 24 minutes and 19 seconds in "Certificates and Diplomas". The minimum waiting time spans from 4 seconds in "Other issues" to 11 seconds in "Tariff and Accreditations".

These preliminary results will allow high level management to re-organize the human resources around the real workload as evidenced by the collected data, therefore allowing the external users to perceive a more efficient service, and the internal users to focus where the workload is increased.

V. Conclusion

With the emergence of technologies such as Voice over IP and the easy inherited ability to store records in a database, new challenges regarding the use of this data are addressed. Treating large amounts of data and extracting a concise and timely relevant information allow to make decisions that may improve the performance of the systems and the way people deal with these systems.

The views presented in the results section are only representative of the full power of this analysis, as the IVRCube makes it possible to create all types of views on the data by gathering and filtering the various dimensions.

We suggested a multidimensional data analysis model using data cube to elicit comprehensive analysis results that fits at least two purposes: keep comprehensive historical data and monitoring the service. The system can be used to conduct a multidimensional analysis through an OLAP operation according to the level of abstraction for each dimension. Because the analysis system using data cube can easily extract comprehensive analysis results within the viewpoints of trend it can be used beyond monitoring.

The ultimate objective is to create a dashboard for the University's high level management, so that the information the system provides can be effectively and timely used to make correct management decisions, namely by allowing the re-allocation of resources considering date and time workload constraints.

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