Identifying the Main Paths of Knowledge Diffusion in the Voice over Internet Protocol

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Abstract

With the rapid growth of the Internet, new applications and services are continuously being proposed for the telecommunication systems. Voice over Internet Protocol (VoIP) is one of the most prominent models for enabling convenient, high-rate, and on-demand voice conversation services through any IP-based network. The increased use of VoIP as an accepted communication form has resulted in a growing volume of academic research dedicated to their assessment.

This study integrates the key-route main path analysis approach and edge-betweenness clustering technique to analyze the knowledge diffusion structures in VoIP domain. The former reveals a roadmap of VoIP including the overall development trajectory and the linkage among different research groups. The latter technique uncovers five major research groups in the VoIP-related literature: wireless network, quality of service, security, traffic and capacity, and VoIP services and architecture. The findings derived from this study highlight the significant development trajectories and contributors of VoIP-related studies.

Keywords: VoIP, Voice over Internet Protocol, Citation analysis, Key-route main path, Clustering-betweenness

1 Introduction

This era of burgeoning Internet use has not only stimulated significant changes in the area of telecommunications but has also become an essential part of modern life. One of the popular applications in this Internet age is voice over internet protocol (VoIP), a widely used and new way of voice communication in public switched telephone networks (PSTN) and cellular networks. The VoIP system may be the best alternative to the traditional circuit network since it has several benefits and advanced features, such as lower cost, integrated services, highly scalability and ease of updating, disaster discovery, and security [1]. VoIP has therefore attracted intense and growing attention that range from the communication industries to academic communities.

Research related to VoIP technology has focused on different themes over the years. In the 1970s, people were trying to find an alternate to the traditional telephony for carrying voice signals, and till 1981, researchers focused on the development in the IP area [2]. After the rapid growth of packet based voice research, researchers switched their focus to the problems of jitter and delays of real-time communication applications [3]. Currently, VoIP applications have gained increasing popularity in wireless networks, with much work being done on congestion control, scheduling, security, and capacity analysis of VoIP over wireless networks. The possibility of the VoIP communication over satellite linkage has introduced security concerns and has highlighted the role of digital filters to improve signal quality [4]. Thousands of papers have been published in journals, conferences, and books on the VoIP domain, and among these are several review papers such as Singh et al. [2], Kazemitabar et al. [5], and Karapantazis and Pavlidou [6]. These authors elaborate on the development of this domain, present clear and concise frameworks, and address technical problems for VoIP. From the view of disciplinary development, recognition of the significant topics of VoIP is fundamentally reshaping the traditional communication field.

Main path analysis, a citation-based bibliometric method, is commonly used to uncover the most significant paths in a citation network and to trace the development trajectory of a research field. The approach of main path analysis has been implemented in investigating the development of various fields such as data quality [7], data envelopment [8], and e-tourism [9]. However, little research has examined the communication-related domain development in terms of the main path analysis method.

Shih et al. [10] reveal and compare the knowledge diffusion trajectories for the VoIP studies using the methods of local/global main path which reflect the

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major developments of VoIP and show the knowledge diffusion process. However, it is hard to realize the comprehensive development of the VoIP field by these two methods.

In this study we will adopt advanced and effective methods to handle thousands of papers in VoIP investigate and visualize VoIP articles. With a citation-based main path analysis to trace the knowledge diffusion paths in VoIP domain, we can uncover the development trends of VoIP. This paper thus contributes to the VoIP research field by providing an overall knowledge diffusion trajectory, which can shed light on the progress of telecommunication systems.

2 Related Works

2.1 Review Studies on VoIP

In terms of communication service, VoIP has been described as a revolutionary technology that transmits voice conversations over an internet-based telephony network [11-12]. Due to its attractive flexibility and low cost, VoIP was rapidly accepted and has been made available in both the enterprise and consumer markets for several years [13-14]. Although previous studies mainly focused on engineering technologies, the recent applications of VoIP have also attracted academic attention, especial in comprehensive reviews and analyses [15].

The development of VoIP covers multiple research themes and the academic field has been aggressively exploring the themes and developmental structures of VoIP from various perspectives. Karapantazis and Pavlidou [6] examined all the themes that have the greatest impact on the quality of voice communication. These themes include the Quality of Service (QoS) requirements of voice, the evaluated methods of the performance of VoIP systems, the voice codecs and header compression techniques, the signaling protocols, and the meaningful issue of security in VoIP networks, and so on. Nisar et al. [14] discussed the fundamental principles of VoIP and identified the scheduler and algorithm issues at that time. The authors further classified the scheduling algorithms into real-time traffic schedulers, VoIP traffic scheduler issues and representative schedulers. In a survey of 245 publications on the topic of VoIP security, Keromytis [13] classified them according to the VoIP security alliance threat taxonomy and provided a roadmap of VoIP security for researchers seeking to understand existing capabilities and to identify gaps in addressing the numerous threats and vulnerabilities present in VoIP systems. Singh and Kaur [1] investigated QoS scheduling services and performance related metrics such as jitter, mean opinion score (MOS) and packet end to end delay, and concluded that mobile WiMAX can not only be used to fulfill the demand for high internet speed but can also be used to provide VoIP services.

Although many authors have undertaken overviews of VoIP research, their methods primarily used content analysis or subjective induction, both of which have significant limitations and difficulty capturing all the diverse research issues in the VoIP field. Given these problems in the previous work, this study applies citation-based main path analysis, a more objective research technique designed to collect both dynamic data and cluster keywords into thematic groups.

2.2 Citation-based Main Path Analysis

The methodologies of this study are based on citation networks, i.e., networks constructed from the citation relationships among articles. In a citation network created in this study, each node represents an article, and it is linked to other nodes that it references or is referenced by. Using network terminology, a citation network is a non-weighted and directed network. It is non-weighted because the importance of each citation is regarded as the same. It is directed because presumed knowledge flows directionally from a cited article to the article that references it.

Hummon and Doreian [16] introduced a network analysis technique named main path analysis to identify the key idea flows in the development of a scientific field based on citations over a period of time. Compared with other scientometric methods (e.g., co-citation, co-authorship), main path analysis takes into account the temporal structure of scientific development more than the semantic structure of scientific work [17].

There are several benefits to using main path analysis [18]. First, it simplifies the complexity of the citation network, retaining only the most significant trajectories and ignoring paths of lesser significance. Second, it stresses the relationship of a series of historical-development events, which is helpful for scientists to trace a particular scientific domain. Third, it identifies the important works in a field's historical development. The main path analysis considers not only direct influences through a high citation count, but also takes indirect influences into account.

A number of theoretical themes, as well as successful application domains, have successfully applied main path analysis to trace the most significant paths in a citation network and helped researchers navigate the literature maze [7-9, 19-20]; however, comparatively little attention has been paid to the developmental trajectories of VoIP domain using bibliographical citation data.

3 Research Materials

3.1 Retrieving the Articles from Databases

Although there exist many databases for literature analysis, Web of Science covers more scientific field over time and with a majority of journals written in English [21]. In fact, its Science Citation Index covers more than 250 disciplines and 160,000 journals, with data ranging as early as 1900; it also provides a Conference Proceeding Citation Index in Science and covers 5.2 billion papers published in books and journals from 1990. Web of Science was designed with the intention of satisfying users in citation analysis, a field discussed and debated by scientists for decades [22].

Data retrieval began with a keyword search which was separated into three steps. First, based on the reading of several highly-cited papers and books, we define a query string which consisted of keywords such as "ip telephony", "internet on telecommunication", "voice over broadband", "internet telephony", "broadband phone", "packet based telephony", "packet-based telephony", "internet phone", "packet voice" and "packetized voice." We picked out the related papers that contain these keywords in the title as the initial dataset. Second, we removed articles that were not in our research scope or had missing data, and then chose the articles that belong to the domains of telecommunications, computer science information systems, engineering electrical electronic, computer science information systems, computer science hardware architecture, computer science software engineering, computer science artificial intelligence, transportation science technology and computer science interdisciplinary applications. Third, in order to avoid missing any important papers, we referred to some prior review works [2, 5-6] to make sure all relevant articles referenced by them were within the scope of this study.

Figure 1 shows the growth trend of VoIP articles in our dataset. From the years 1990 to 1997, the number of papers published each year in the VoIP area was less than 10, but in 1998, the number of paper published per year started to grow dramatically until 2008 when the accumulated numbers of VoIP papers surpassed 1000, reaching 1013. In total, the dataset contains 1679 articles from 1990 to 2016 that can be used for further investigation. Before 1998, the development of VoIP was still in the early stages. With the increasing growth of internet technology, VoIP entered into the growth stage in the past decade. However, for the past 5 years, the papers published per year have declined steadily year by year, which shows that the knowledge structure of the field of VoIP has gradually become mature. In the near future, the field will enter into the maturity stage in which the applications, rather than technologies, of VoIP may become the main research topics.

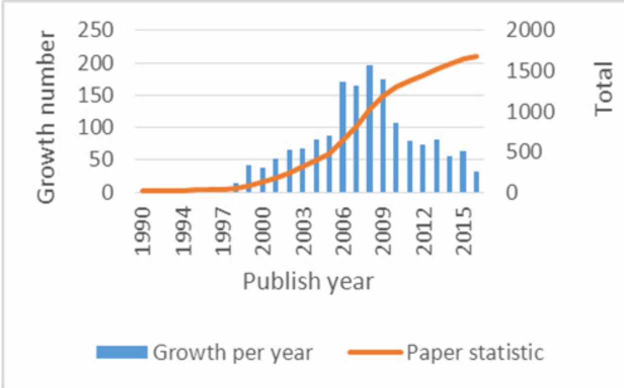


Figure 1. Growth trend of VoIP articles

3.2 Using the Methodologies to Analyze Citation Network

This study applies the methods of main path analysis and clustering technique to explore the knowledge diffusion paths and major VoIP topics. The main path analysis is one of the methodologies to quantitatively analyze academic literature and can handle a large amount of studies in the literature. The purpose of main path analysis is to trace the significant development trajectories in a specified science & technology domain. Clustering is a kind of unsupervised classification algorithm which divides target patterns into groups and it has been addressed in many contexts and by researchers in many disciplines. In our study, the key-route main path analysis and edge-betweenness clustering algorithm were employed to estimate the intellectual structures and relationships of VoIP.

This study uses an in-house software developed by Da Vincier Lab, National Taiwan University of Science and Technology, to calculate the main paths.

3.2.1 Key-route Main Path Analysis

The key-route main path approach proposed by Liu and Lu [18] is one of the more widely used extensions of main path analysis. A key-route in a citation network is a significant link that represents a major channel for knowledge diffusion. Key-route main path analysis guarantees that the main paths will pass through the specified key-routes. It begins the path search from the top significant links, thus guaranteeing the inclusion of these links. The approach requires specifying the number of the top links, e.g., key-route 10, which specifies that the search begins from the top 10 most significant links. By varying the number of key-routes, one is able to control the level of main path details.

3.2.2 Edge-betweenness Clustering

The edge-betweenness of a network link is defined as the number of shortest paths between pairs of vertices that run along it [23]. The links that connect groups in the network have high edge-betweenness. Edge-betweenness clustering removes the highest edge-betweenness links, and then separates the groups from one another during the process. In the end, the underlying group structure will be revealed. There are four procedures to reveal the underlying group. First, one needs to calculate the edge-betweenness of all existing links in the network, and then remove the highest edge-betweenness link. Second, one recalculates edge-betweenness for the links that are affected by the previous removal, which is followed by removal of the current highest-rank link. Third, if the network is divided into two groups, the modularity for such division is computed and recorded, and the recalculation and removal steps continue. The modularity is calculated whenever separation occurs. The recalculation and removal steps are repeated until all the links in the network are removed. Finally, we trace back to the network division that has the largest modularity and obtain the grouping result.

4 Exploring the Diffusion Path of VoIP Studies

4.1 Overall Main Paths

We conducted a key-route main path analysis on all of the literature selected (see Figure 2). The number of top key-routes was set to 10. In the figure, the arrow indicates the direction of knowledge flow and the line thickness shows the size of its traversal count: the thicker the line, the more significant the route. Each paper in the figure is attached with a notation that begins with the last name of the first author, continues with the first initials of the co-authors (in capital letters), and ends with the publishing year. For example, the original paper by Hassan, Nayandoro, and Atiquzzaman published in 2000 is indicated as HassanNA2000 [24].

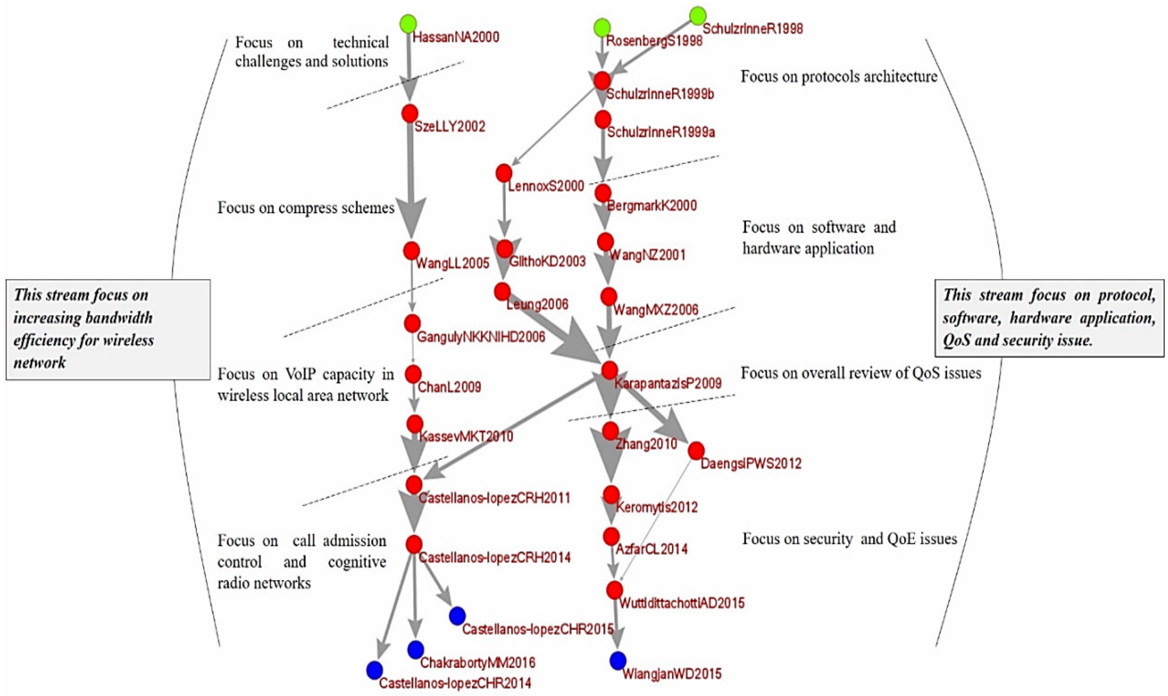


Figure 2. Overall VoIP main paths (for key-route 10)

Starting from the left-hand side, it can be seen that HassanNA2000 [24] proposed the following: a new service that can be expected from VoIP, the technical challenges and solutions, and the emerging products that promise to support VoIP. SzeLLY2002 [25] focused on improving the bandwidth efficiency of existing VoIP applications by proposing a multiplexing scheme, and the results show that the multiplexing scheme can increase bandwidth efficiency by as much as 300%. A significant work following [25] is WangLL2005 [26], which used similar compression scheme as SzeLLY2002 [25] and investigated a scheme that can improve the VoIP capacity by close to 100% without changing the standard 802.11 CSMA/CA protocol. GangulyNKKNIHD2006 [27] discuss the basic requirements for efficient deployment of VoIP services over a mesh network then present and evaluate practical optimizing techniques that can enhance the network capacity, maintain the VoIP quality and handle user mobility efficiently.

ChanL2009 [28] is the first paper attempt to examine the VoIP capacity in the "multicell" environment in which many infrastructure WLANs are deployed in the same geographical area. KassevMKT2010 [29] presented a VoIP packet loss model applicable to wireless access transmission media to satisfy QoS constraints in terms of packet dropping and call blocking probabilities. After KassevMKT2010 [29], VoIP has drawn much interest in wireless networks because of the convergence of all IP architecture in wireless and wireline networks. Therefore, it is followed by a series of works by Castellanos-Lopez et al. Castellanos-lopezCRH2011 [30] discussed call admission control strategies for voice over IP (VoIP) traffic over wireless access networks. Then, Castellanos-lopezCRH2014 [31] proposed a novel joint connection-level and packet-level analytical model for the performance evaluation of cognitive radio networks (CRNs) under VoIP traffic. Castellanos-lopezCHR2014 [32], Castellanos-lopezCHR2015 [33], and Chakraborty-MM2016 [34] focus on the issue of call admission control scheme and cognitive radio networks with VoIP traffic. In summary, all of the articles on the left hand side talked about increasing bandwidth efficiency and then proposed a new scheme for improving the QoS in wireless networks.

On the right hand side, it begins with two researches that merge with Schulzrinne's work [35] on internet telephony architecture and protocols, and then divides into two research activities. The first two studies are RosenbergS1998 [36] and SchulzrinneR1998 [37], which focus on protocol issues. The former proposes a new protocol architecture, called Brokered Multicast Advertisements (BMA) which can be applied to location of any service across a wide area network, while the latter describes Session Initiation Protocol (SIP), and shows how its basic primitives can be used to construct a wide range of telephony services. Then in 1999, SchulzrinneR1999b [38] describes the upper-layer protocol components that are specific to Internet telephony services: the Real-Time Transport Protocol (RTP) to carry voice and video data, and the SIP for signaling. Furthermore, it also mentions some complementary protocols, including the Real Time Streaming Protocol for control of streaming media, and the Wide Area Service Discovery Protocol for location of telephony gateways. It is clear to see that Schulzrinne and Rosenberg are the pioneers of the protocol architecture of Internet telephony.

After SchulzrinneR1999b [38], there are two research streams. The first one is also led by Schulzrinne followed by BergmarkK2000 [39], WangNZ2001 [40], and WangMXZ2006 [41]. BergmarkK2000 [39] observed that convergence between the existing telephone networks and data transfer over the Internet required new software. WangNZ2001 [40] focused on an evaluation of media gateway performance (in terms of voice quality) that is affected by impairments of an IP network in a practical environment. The author points out the crucial factors that affect a successful VoIP network and possible remedies are also presented in this work. WangMXZ2006 [41] designed and implemented a QoS-provisioning system that can be seamlessly integrated with the VoIP systems to overcome their weakness in offering QoS guarantees.

The second stream was headed by LennoxS2000 [42], and followed by GlithoKD2003 [43] and Leung2006 [44]. In this stream, the authors saw the potential of the usage of VoIP and tried to add some services in VoIP. GlithoKD2003 [43] studied the cases of high-level service creation environment. Leung2006 [44] noticed that VoIP is promising for long distance calls, so he focused on the issue of QoS by proposing a dynamic bandwidth allocation scheme to meet related requirements.

These two streams merged to KarapantazisP2009 [6], which is a comprehensive review paper for QoS. Beginning with the QoS requirements of voice, the author employed methods to evaluate the performance of VoIP systems, then surveyed voice codecs and header compression techniques. It also discussed signaling protocols and continued with a comprehensive study on call admission control (CAC) schemes tailored for the needs of VoIP. Furthermore, it shed light on the meaningful issue of security in VoIP networks. Finally, it discussed the potential of satellite systems to provide VoIP and claimed that Satellite Internet service will support VoIP in the future. Zhang2010 [45] discussed requirements for a VoIP anonymization Service (VAS) in terms of functionality, performance and usability. Keromytis2012 [13] presented a comprehensive survey of VoIP security academic research, and then discussed further work on understanding cross-protocol and cross-mechanism vulnerabilities, which are the byproduct of a highly complex system-of-systems and an indication of the issues in future large-scale systems. AzfarCL2014 [46] first examined Skype and nine other popular Mobile VoIP applications for Android mobile devices, and analyzed the intercepted communications to determine whether the captured voice and text communications were encrypted or not. The results indicated that most of the applications encrypt text communications. However, voice communications might not have been encrypted in six of the ten applications examined. WuttidittachottiAD2015 [47] and WiangjanWD2015 [48] presented a serious study of VoIP-Quality of Experience of a well-known VoIP application and three modern ones, Skype, LINE, Tango and Viber.

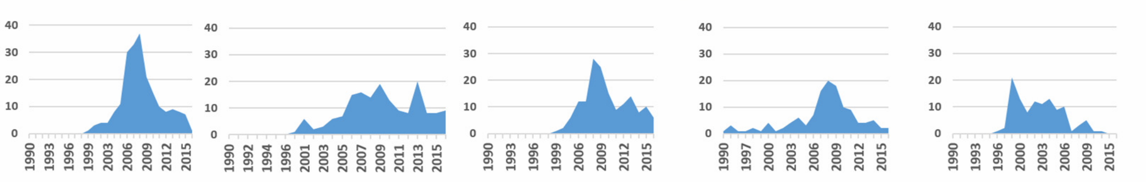
4.2 Clustering in VoIP

We applied the edge-betweenness clustering to separate the citation network into different groups. The clustering was conducted on the dataset of size 1676, but 3 review articles were removed. The reasons for this is review articles tend to reference documents belonging to different subareas within the discipline, and from the perspective of knowledge transmission, they play the role of knowledge brokers and link different subareas together [9], thus blurring the group boundaries within the citation network. One previous study [8] showed that removing the review articles actually improves the clustering results.

Edge-betweenness clustering separated the citation network into many coherent groups with size ranges from 25 to 210. Among these groups, the top 5 groups had sizes of 210, 164, 159, 136 and 111. We discuss these top five groups, which represent 47% of our dataset. The sizes of these groups ranked number 6th to 10th are 81, 77, 61, 53, and 47. Taken together, the top 10 groups contain 1099 (65.6%) of our data. The remaining 10 groups are of size 40 and smaller. The majority of these small-size groups are either isolated nodes or 'islands' in the citation network. Reporting results with so many small-sized groups is actually one of the advantages of edge-betweenness clustering methods—it does not enforce an attachment of an entity to a seemingly irrelevant group, which is a shortcoming of the K-means method. It therefore leaves remotely relevant entities alone as small groups, and the larger-sized groups are the subareas that have a large amount of coherent research activities. In the following discussion, we concentrate on the five largest groups and apply the key-route main paths which are created from the top 5 key-routes to analyze each group.

In order to recognize the contents of these groups, we examine the keywords in their titles and abstracts. The keywords are selected from a list of terms used in all the articles excluding words that are too common in this area such as 'VoIP', 'network', 'voice', etc. Variations of terms with identical or similar meanings are regarded as the same keywords—for example, 'VoIP', 'voice over IP' and 'voice over internet protocol' are all treated as the same keyword. Figure 3 presents the top 3 keywords in each group and the growth curve for the number of papers in each subarea. The parentheses attached with each keyword indicate the keyword's mean number of appearance in the group. For example, in the 1st group "ieee 802 11" appears on average 2.05 times per article. From the keywords, the emphasis of the 1st group is largely on "wireless network" as it contains keywords such as wireless network, wireless, WLAN, from similar reasoning, the topic of the remaining groups are "QoS issue," "security issue," "Traffic and capacity issue," and "VoIP services and architecture." From the growth curves 1, 3, and 4, it is evident that the amount of papers grew extremely fast during 2003 to 2008, but after 2008, the amount of papers gradually decreased. Moreover, in groups 2 and 3, the "QoS issue" and "security issue" are remaining popular research topics after 2012. Another noticeable finding is that the first subarea shows that the VoIP wireless network issue is relatively popular in VoIP field during 2004 to 2012 since it contains the highest number of papers at 60, compared to the other topics whose largest annual amount never surpasses 40. In contrast, group 5's topic "VoIP services and architecture" was only discussed in the early years and the amount of papers it generated is much lower than the other groups.

	Group 1 (210 articles)	Group 2 (164 articles)	Group 3 (159 articles)	Group 4 (136 articles)	Group 5 (111 articles)
1	ieee 802 11 (2.05)	Quality (4.93)	Attacks (1.23)	Traffic (1.15)	Service (1.56)
2	Wireless (1.5)	Quality of service (0.8)	Security (1.12)	Model (0.74)	Architecture (0.77)
3	WLan (1.47)	Packet loss (0.63)	Spam (0.92)	Capacity (0.72)	SIP (0.69)
Focus	Wireless network	QoS issue	Security issue	Traffic and capacity issue	VoIP services and architecture
Growth trend					



Note. the number in the parentheses attached with each keyword indicates its mean appearance in the group

Figure 3. Clustering results and growth trend

In order to examine the development within this subarea, we applied the main path method to analyze the papers contained in the subnetwork.

4.2.1 Wireless Network

The resulting main paths identified 10 papers (Figure 4). The majority of the articles on the main paths are about the development of improving quality of experience (QoE) in wireless VoIP and performance optimizations for deploying VoIP services in a wireless LAN. The treatment of website design on the main paths begins with Goodman1999 [49], who focused on understanding the causes of delay within analog modems, and continues with the objective of developing recommendations to minimize delay for VoIP applications. Then, SzeLLY2002 [25] and WangLL2005 [26] are shown in our key-route main paths to focus on improving the bandwidth efficiency and to investigate a scheme that can improve the VoIP capacity without changing the standard 802.11 CSMA/CA protocol. Another starting point is the citation of WangLL2005 [26] is HiragurllLM2002b [50], which proposed a novel multiple access protocol based on autonomous distributed control that allows wireless LANs to satisfy the VoIP requirements. The other start point is GargK2003 [51], who studied the inherent limitations of the 802.11 (a/b) distributed coordination function in supporting VoIP calls over a wireless LAN. Then, HoleT2004 [52] presented a means of estimating the capacity of a voice-only 802.11b network, and validated the estimation by means of limited experiments. After HoleT2004 [52] and WangLL2005 [26], GangulyNKKNIHD2006 [27] discussed the basic requirements for the efficient deployment of VoIP services over a mesh network, and then he presented and evaluated practical optimizing techniques that can enhance the network capacity, maintain the VoIP quality and handle user mobility efficiently. KashyapGDB2007 [53] studied the problem of supporting VoIP calls in a wireless mesh network. The authors also developed a new method for routing using call statistics that uses prior calling patterns to avoid potentially critical links. LambrinosD2009 [54] and LambrinosD2013 [55] are a series of research lines for improving the quality of experience in wireless VoIP through novel call scheduling.

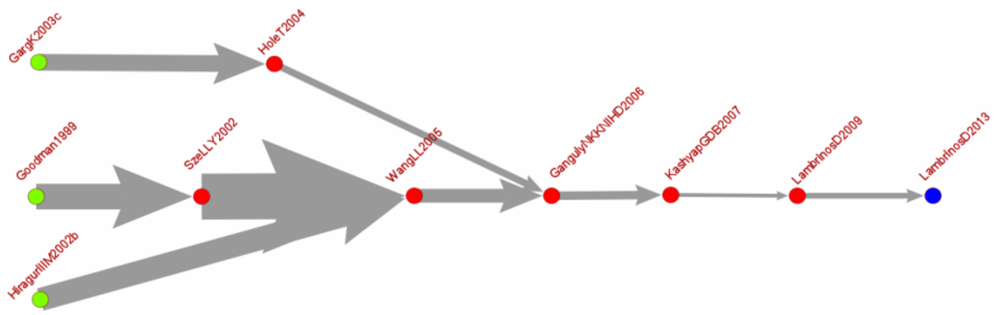


Figure 4. Key-route main path for “Wireless Network”

4.2.2 QoS Issue

In Figure 5, the discussion on the main paths begins with ConwayMS1997 [56], which developed a simulation tool for performance analysis and evaluation of Internet phone applications. ConwayZ2002 [57] proposed a simulation-based methodology to analyze the subjective quality of VoIP calls as a function of network QoS parameters and choices in implementation and configuration. BirkeMPR2007 [58] and BirkeMPR2010 [59] presented their experience in the real-time monitoring of VoIP calls from a commercial operational network. GibeliBMZM2013 [60] presented the development of baselines based on Internet Protocol Detail Record (IPDR) to support VoIP traffic management in open-access Metropolitan Area Networks (MAN). Garcla-doradoDRMMDA2014 [61] proposed a novel architecture that provides very high performance, novel services and significant cost reduction by using commodity hardware with minimal interference.

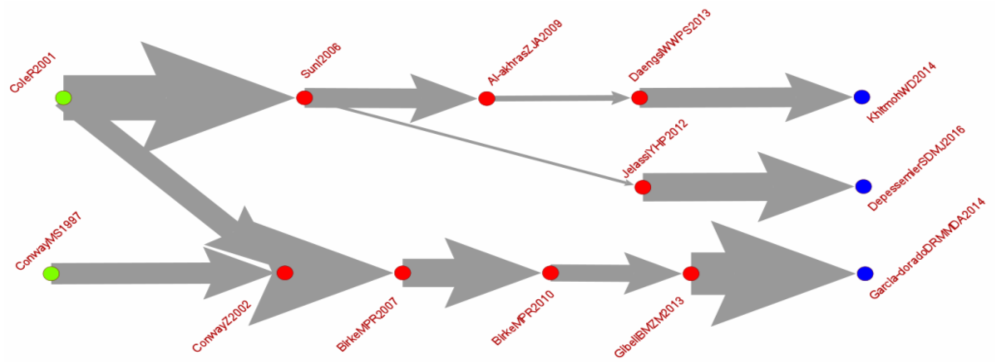


Figure 5. Key-route main path for the “QoS issue”

ColeR2001 [62] discovered that the relevant transport level quantities are the delay, network packet loss and the decoder’s de-jitter buffer packet loss. Furthermore, it found that an in-path monitor requires the definition of a reference de-jitter buffer implementation to estimate voice quality based upon observed transport measurements. SunI2006 [63] presented new models for objective, nonintrusive, prediction of voice quality for IP networks and to illustrate their application to voice quality monitoring and playout buffer control in VoIP networks. JelassiYHP2012 [64] accounted for the voicing feature of signal wave included in missing packets to extend conventional parametric no-reference speech quality models. DepessemierSDMJ2016 [65] identified device characteristics, context parameters, and user aspects that influence the usage behavior and experience during VoIP calls. Al-akhrasZJA2009 [66] presented a method for predicting effective factors of equipment impairment. DaengsiWWPS2013 [67] improved on the accuracy and reliability of VoIP quality measurements by proposing the subjective VoIP Quality Evaluation model. KhitmohWD2014 [68] proposed VoIP quality estimation that is reliable and suitable for estimating MOS of G. 729 in a QoS environment. Overall, this group focused on monitoring and evaluating the VoIP quality using different methods and extended the monitoring models to improve VoIP systems.

4.2.3 Security Issue

The emergence of VoIP has not only offered numerous advantages for end users and providers alike, but also simultaneously introduced security threats, vulnerabilities and attacks not previously encountered in networks with a closed architecture like the PSTN. In this group, which is illustrated in Figure 6, there are five start points, which then divide into two different streams. All the sources are published after 2004, indicating that this issue was followed by the explosion of VoIP usage. Beginning from the top stream led by SengarDW2006 [69], we can see this research paper mentioned that integrated signaling environment can become a security threat to emerging VoIP and PSTN networks, so it proposed a security solution as a fix. Because SengarWWJ2006b [70] noticed that VoIP is more susceptible to Denial-of-Service (DoS) attacks than regular Internet services, it presented an online statistical detection mechanism to detect DoS attacks in the context of VoIP. GeneiatakisLK2008 [71] provided a categorization of potential attacks against VoIP services, which was followed by specific security recommendations and guidelines for protecting the underlying infrastructure from these attacks to ensure the provision of robust and secure services. LiKT2011 [72] applied the building security gateway (BSG) procedure to VoIP to achieve secure sessions; VoIP call monitoring and intercepting functions are also implemented in BSG. IrshadSRCUG2015 [73] proposed a key-agreement protocol to servers that can authenticate the user on the request message received, rather than the response received upon sending the challenge message; in this way, another round-trip of exchanged messages was not needed and thus prevents a possible DoS attack. ZhangTZ2016 [74] pointed out that an energy-efficient authenticated key agreement protocol for SIP should be provided to ensure the confidentiality and integrity of data communications over VoIP networks, and so they proposed an efficient authentication protocol for SIP by using smartcards based on elliptic curve cryptography. NaqviCM2016 [75] proposed an improved anonymous authentication scheme for SIP.

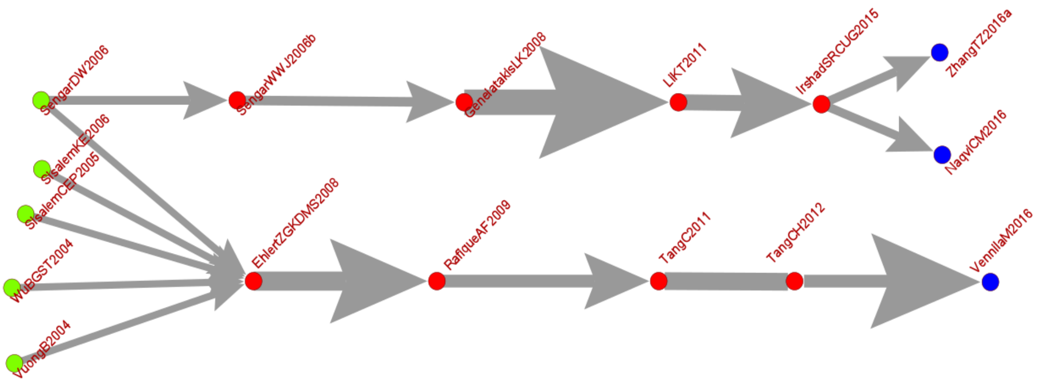


Figure 6. Key-route main path for the “Security issue”

The second stream is composed of four start points. SisalemCEP2005 [76] provided the description of a VoIP architecture that is optimized to offer VoIP services in various scenarios. SisalemKE2006 [77] addressed the issue of denial of service attacks targeting the hardware and software of voice over IP servers or by misusing specific signaling protocol features. WuBGST2004 [78] proposed the design of an intrusion detection system that targets VoIP systems. VuongB2004 [79] presented a survey of the security problems in VoIP networks, with an emphasis on both intrusions and intrusion detection methods. EhlertZGKDMS2008 [80] proposed a two-layer architecture to prevent denial of service attacks on VoIP systems based on the SIP. RafiqueAF2009 [81] evaluated DoS attacks against SIP-based VoIP systems. TangC2011 [82] and TangCH2012 [83] discussed the methods for preventing the SIP flooding attacks in VoIP network. VennilaM2016 [84] proposed a detection technique for RTP flooding attacks.

4.2.4 Traffic and Capacity Issue

This group discusses the key challenges of the telecommunication era, which represent the development of new Internet architecture models that aim to satisfy the QoS requirements of IP-based services. The relevance of this issue is related to the transformation of the Internet to a commercial infrastructure able to provide differentiated services to users with widely different service requirements.

In Figure 7, BrunoGG2000 [85] presented a procedure to evaluate the parameters of the LEAP (Linear Bounded A-rival Processes) traffic characterization with a stochastic model accounting for traffic flow. There are five articles presented by [86-90]. EstepaEV2003 [86] presented an experimental study that extends the current knowledge of the VAD/DTX codec influence in the transmission rate. EstepaEV2004 [87] proposed a new frame generation model for those audio codecs which handle SID frames and deduced an analytical expression for the mean bit rate at the input of the IP network as a function of the number of frames per packet. EstepaEV2005 [88] presented a form to accurately predict the VoIP traffic mean bit rate. EstepaE2007b [89] adapted the fluid model to obtain accurate loss prediction in the multiplexing process of VoIP sources equipped with comfort noise generation and provided an algorithmic solution for its dimensioning application. EstepaE2008 [90] discussed VoIP codecs and proposed a simple but efficient algorithm which can be applied in dimensioning or admission control to find out the bandwidth reservation required to guarantee delay and loss in a packet-switch multiplexer node for VoIP traffic. The articles after KassevMKT2010 [29] were discussed in the previous section, which focused on CAC and CRNs issues.

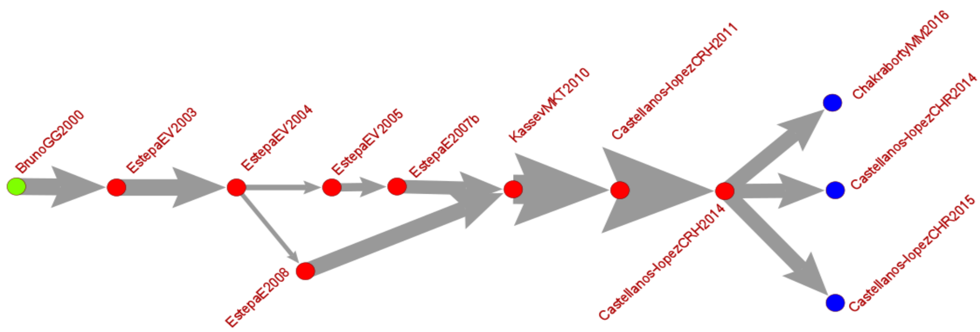


Figure 7. Key-route main path for the “Traffic and capacity issue”

4.2.5 VoIP Services and Architecture

The articles in Figure 8 were published relatively earlier. At that time, the Internet had already become mature and VoIP was a promising new technology which introduced a wealth of new service possibilities, and so these articles focused on enhancing the service architectures and standard protocols.

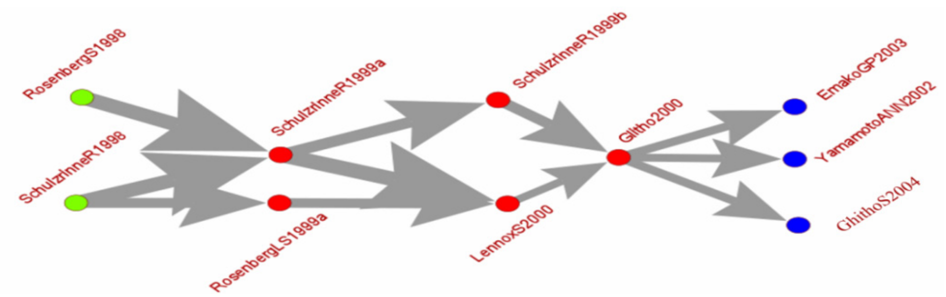


Figure 8. Key-route main path for “VoIP services and architecture”

There are five papers presented by Rosenberg and Schulzrinne in 1998 and 1999, which also appeared in key-route main path we mentioned in the last section, with the exception of RosenbergLS1999a [91]. This paper developed requirements for programming VoIP services. In this group, we can see Rosenberg and Schulzrinne first focused on studying the architecture and protocols for VoIP. Since the Internet was increasingly being used at that time to carry voice and video, it was critical for VoIP service to smoothly interoperate with PSTN. For this reason, these authors strived to find Internet Telephony Gateway and proposed a new protocol architecture, called BMA. After their work, LennoxS2000 [42] scrutinized the ITU-T and IETF advanced services architectures for VoIP, and summarized the salient features and weaknesses. Glitho2000 [92] presented a critical overview on advanced service architectures for VoIP, and this paper was followed by three end points of this group: GhithoS2004 [93] was a case study on the use of parlay call control APIs in SIP networks; YamamotoANN2002 [94] described a traffic control scheme for carrier-scale VoIP services; and EmakoGP2003 [95] presented a mobile agent-based advanced service architecture for wireless VoIP.

5 Conclusion

This study explores the knowledge diffusion paths of VoIP by using citation-based main path analysis and clustering methods to analyze the 1679 related articles published from 1990 to 2016. This was done in order to reveal the main research trends and emphases, as well as to classify the VoIP publications. We identified the two research streams of bandwidth efficiency of wireless network and multifaceted applications of VoIP, and five major VoIP subarea issues: Wireless network, QoS, Security, Traffic and capacity, and VoIP services and architecture. This study then examined the developmental trajectory of the VoIP subareas, its practical applications and its structural breakdown.

This paper uses graphical representations to present the developmental trajectories so that readers can instantly recognize the interrelationships between these important research themes along evolutional timelines. This clear intellectual structure of VoIP is helpful for new researchers who are considering entering into the VoIP research field. It is also hoped that this evolutional structure will bring researchers new insights into understanding VoIP in terms of not only past but perhaps also future development.

One limitation of this study is worth mentioning. Main path analysis relies on the citing/cited relationships among articles, which makes it not conducive to the newest publications because citations usually need time to accumulate. The results of the main path are a snapshot of the field at the time the data is collected and that the latest topics may be overlooked. For example, we do not see any of the software-defined networking (SDN) articles on the main paths. SDN is one of the most potential technologies to network management for making the configuration of network dynamic, flexible and programmatically efficient, and to strengthen network performance and monitoring [96-98]. Recently, many scholars have paid their attention to the impact of SDN on VoIP quality and efficiency. Bakhshi [96] proposed an SDN-based architecture to solve call admission control (CAC) problem, Bahnasse et al. [97, 99] developed a hybrid SDN model to manage dynamic bandwidth allocation for Multi-Protocol Label Switching (MPLS) network, and Siniarski et al. [100] implemented a monitoring system of quality of experience (QoE) with low communication overhead. All these studies reveal that SDN as an advanced approach is helpful for enhancing VoIP quality. The limitation of this study can be turned into an opportunity for future works to examine further the impact of SDN on VoIP.

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