# Comparative analysis of traditional telephone and VoIP systems (April, 2014)

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***ABSTRACT*** **- Pakistan is in the period of upgrading communication infrastructure. Therefore many communication techniques are taking birth day by day and building the effective way of communication and one of many is IP Telephony system. In Pakistan the adoption of IP telephony is slow because of variety of reasons, but the main factor is the reliability of legacy phone systems. With a weak economy in Pakistan, the management team of organizations is happy to get as much mileage as possible out of their existing infrastructure, and if the legacy phones are working fine, IT team will need a really strong reason to change it to IP telephony. The purpose of this independent study is to drive the correct adoption of new IT & communications technologies in Pakistan to enable increased productivity and added value for businesses. This paper investigates the issues of service quality and cost lie in Pakistan and on the basis of this issues what steps we need to consider while deciding to implement VoIP. After implementation of VoIP in two different organizations and we then analysis their comparative results with legacy phone systems.**

## I. INTRODUCTION

In the past twenty or so years, the world of telephone has transformed from being an analog network to a digital one which requires a circuit switch. Lately, the recent development is that of the transfer from the circuit-switch to technology involving packet-switch for the transmission of data and voice via Internet Protocol (IP) networks. Another technology that has now surfaced is that of Voice-over-IP (VoIP) technology. This technology has been promoted by the reasonably priced long distance charges and no access charges.

Though the issue of the effectiveness of Public Switched Telephone Networks (PSTNs) and VoIP is still going on, an escalating number of businesses are choosing to swap their (PSTNs) for the reasonably priced VoIP substitutes. The time when VoIP first emerged on the surface, internet telephone was regarded as having performance problems. It earned a bad name for dropping calls and pathetic voice quality. Nevertheless, important developments have been made with regard to VoIP. Moreover, there are several reasons why bringing a change could prove to be beneficial. The regions where VoIP presently has an edge over PSTN consist of benefits like special feature accessibility, cost and scalability.[15]

Conventional long distance and regional phone service providers have started paying proper attention to internet telephony which is a comparatively novel form of technology. Using this technology, telephone users can interact through real-time audio message using five applications. Such form of communication can be carried out by means of fax to fax, PC to PC, phone to phone, PC to fax and PC to PC. The main notion associated with Internet telephony is that audio is transmitted across packet-switched networks as compared to conventional switching circuits. Because of the sustained ever-increasing development in the use of Internet, this technology has started challenging the conventional long distance and regional phone service providers. The final success and widespread use of this technology will depend on the capability of Internet telephony to offer value added features like incorporation of voice and data, flexibility and speed at a reasonable price as compared to a conventional telephone service.

Internet telephony requires packet-switched networks that are considerably diverse from the conventional circuit-switched networks that used by the traditional telecommunication sector. A framework known as Public Switched Telephone Network (PSTN) forms the basis of circuit-switched networks. In this framework, the phone connection is completed by switching the calls to open circuits. The assumption under which this system was designed stated that at one time, not greater than 30% of subscriber would be using the phone service. Owing to the current increase in Internet use, this assumption is being closely analyzed and is compelling Local Exchange Carriers to increase the capacity. This intricate framework is not required for packet-switched networks. In order to transmit data, a packet-switched network employs routers. Every packet takes a header with it which relays the location of the sender and the order of data. Then a hub computer breaks the information packets and transfers it to routers which find out the least jammed path to the end location. After this, the destination computer again constructs the information into its authentic form. Considerably less intricate framework is required for packet-switched networks as compared to the circuit-switched networks when it is delivering the data of same capacity.

## II. TRADITIONAL TELEPHONE ISSUES IN PAKISTAN

### *A. Service Issues*

Pakistan faces a dearth of telephone service providers who encounters many hurdles in the innovative technological aspects.

- PSTN contains no service which offers audio conferencing features to their clients
- Remote Connectivity is not supported by PST.
- Inter Branch Connectivity is non-existent.
- No support is provided for the establishment of Call Center by the conventional telephone.
- If there any problem occur in PSTN setup then support require from External Vendor.
- PSTN does not provide any such feature of Voice Mailbox as VoIP provide.
- NO native Voice Recording
- Reporting and monitoring features are non-existent.
- PSTN network does not have any Security Feature.

### B. *Management Issues*

- Due to huge inter office calling, a lot of difference is created in the billing
- The feature of scalability and flexibility is not sufficient to meet the future needs as extensions are less in number.[2]
- Cost Savings and Return on Investment (ROI) is not offered by PSTN.
- The cost is increased by the expansion of PSTN.
- Upgradation of PSTN is not possible.
- The Extended Features on PSTN need Licensing Cost.
- There is no guarantee that it will be accessible or reliable.
- It would soon become obsolete.
- VoIP Network Equipment uses electricity
- Backup Phones, Backup Equipment and Backup Power

## III. REPLACEMENT OF PSTN TECHNOLOGY

Various internet telephone services can take the place of PSTN. Such services include Skype and Viber etc. However, VoIP is still the most reasonable technology which can take its place. By having widespread control on it, VoIP can be changed or expanded as per the requirement. Unlike other telephone technology, VoIP provides more business features like safety parameters. VoIP and PSTN technologies can be compared thus: VoIP technology transmits packetized voice signal over IP network and disintegrates the audio into small portions and transfers then as a torrent of packets across a digital network. Here such packets follow the best path to arrive at their destination. Such packets can travel via different paths to arrive at the same destination. However, at the end point they change back into their original state.

Very less of the VOIP call providers for long distance are cheap, it is east to connect to PTSN network if the providers have efficient call management, rich features and also control over the call routes. Hence VOIP technology is in the lead in the world. Packet loss is still a major setback of VOIP services. [1]. Internet devices, smart phones and computers all have VoIP serviced. And one can send sms or make calls over Wi-Fi or 3G.

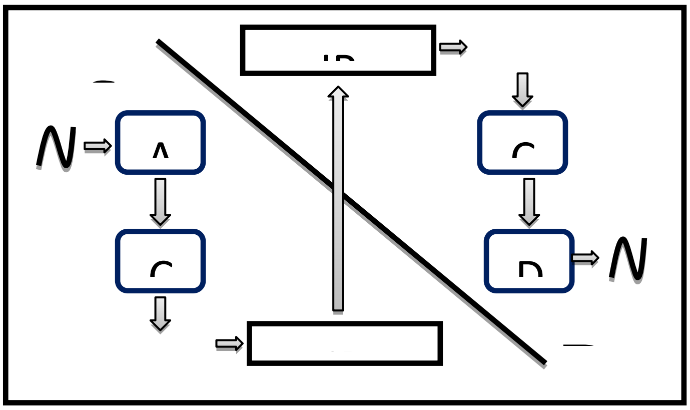

Fig. 1. Basic Transmission Flow of VoIP

## IV. VOIP SSYSTEM

Distributing video and audio as packets via network by dividing and converting them to digital form is what VoIP does. The chunks are put together simply so the picture of an ordinary circuit switching call is in the mind of people having a phone conversation. The private Branch Exchange systems are used for external telephone calls and for internal communication which is free; this is a shift from corporate voice communication. All calls are operated by the PBX in VoIP.

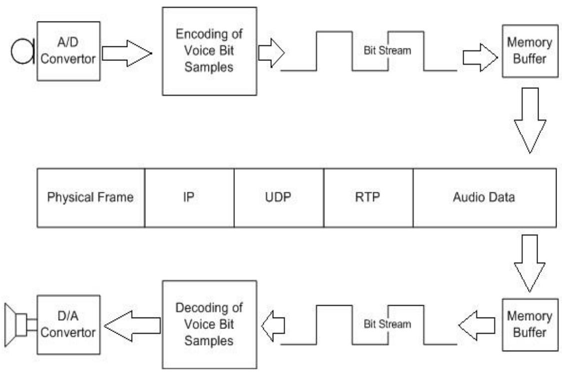


Fig. 2. Flow of Basic VoIP communication

IP phones instead of ordinary PBX are used for VoIP. The simple way of connecting wide area IP network to an IP phone system is shown above. LAN is used to connect the phones and local calls can be made through it. The diagram in Fig. 2 illustrates a simple connection of IP telephone system with wide area IP network. The IP phones are linked via LAN and local voice calls can be done over a LAN. However, the voice calls between different sites can be made on wide area IP network. The IP phones can De packetize and Packetize encoded speech also. The IP phone also contain codec’s which are used to encode/decode and digitize the speech. The connections among different sites can be done via wide area IP networks. The IP phones are registered by proxy servers and coordinate signals also. PSTN can be connected via VoIP gateways. [14]

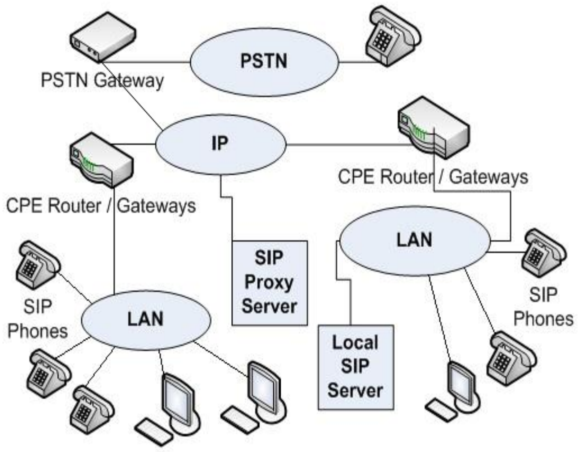


Fig. 3. VoIP from end to end

## V. VOIP PROTOCOLS

The communication services of IP telephone systems depends upon the protocol stack signals to call and to disconnect and also to share the abilities. Those signals also help the system to transfer information needed to search the users. The standard protocols discussed in this literature are: MGCP (Media Gateway Control Protocol), Session Initiation Protocol (SIP), H.323 and Megaco/ H.248. Some of the other proprietary protocols discussed are: Viber, Skype, MiNET, Skinny Client Control Protocol (SCCP), Inter-Asterick eXchange (IAX) and OpenScape Voice. The main characteristics, protocol stacks and architecture of each standard protocol are given below.

### *A. H.323*

The protocol H.323 is the first international multimedia communication protocol. Its main task is to convert the video, voice and the information on the packet networks. This protocol features internet integration and World Wide Web together via PSTN interfacing. Other services provided are: prepaid calling card services, wholesale transit of voice, enterprise/residential video and voice services. Remote users are allowed to make video calls along with editing of any document in the real time on internet. Other services performed by H. 323 are it allows call/ customization of phone calls, call transfer, user location and many other tasks with the help of HTTP interface between client/server on network. [5]

### *B. SIP*

SIP (Session Initiation Protocol) is specially designed by Internet Engineering Task Force (IETF) for initiating, terminating and maintaining interactive communication sessions among users. Instant messaging, video, voice, virtual reality and interactive games are included in these sessions. The functions of SIP are to allow user availability, location, endpoint capabilities, session management and setup [3]. This protocol also enables the unified messaging, voicemails, location services, context-aware communication, internet collaboration and conferencing and integration of applications and communications.

### *C. MGCP*

The Media Gateway Control Protocol (MGCP) is actually an IETF VoIP protocol which is designed for built-up gateways, IP devices, and big trunk gateways. In December of 2003, an updated version with additional features was launched in RFC 3661 [4]. Amongst the features of a broken-down Multimedia Port, MGCP is used. A Multimedia Gateway mainly comprises of a Media Gateway Controller which consists of the call-management program, as well as a Media Gateway that has a controls over media. [6] Its main purpose is to provide assistance to central call-management unit. Because the Call Agents are situated towards the ends of the network, fall in touch via the call management protocol like H.323 or SIP [3], MGCP cannot connect to fellow protocols. And because they are located towards the ends, SIP or H.323 must be installed in the main network.

After bring together and matching up various VoIP transmitting protocols, it was seen the H.323 and SIP benefit by passing the signal to the following protocol, whereas MGCO and Megaco employ a master-slave transmission technique. After being the trendy technique, and capturing the market early on, H.323 has lately been overtaken by SIP. Generally, SIP is considered as the fitting technique when communicating between call agents, or when handling trunk groupings. Apart from that, SIP provides a number of other facilities other than the phone. These consist of messaging, supervising attendance, and verbal e-commerce through the web.

### *D. Other protocols*

This study has been conducted to deliver a full overview of the architecture and evolution of VoIP protocols. Additionally, we also look some other substitute proprietary protocols. One of them is Skype [9] which is the name of the company as well as the name of the software. Skype utilizes an overlay network and an exclusive VoIP protocol. Instead of using the old client server model Skype uses a peer-to-peer model. Skype offers both on-net (SkypeIn) and off-net services (SkypeOut), it on-net service is free while it charges a fee for its off-net services that supports calls to mobile phones and PSTN. Viber [10] is some other protocols that utilize peer-to-peer model for VoIP systems. A Cisco protocol known as Skinny Client Control Protocol (SCCP) [11] is used for conferencing using the IP, and real-time calls. It is used with Cisco VoIP phones and Cisco Call Manager. Another exclusive VoIP protocol is MiNET [12], it is used by PBXs and Mitel phones. OpenScape Voice [13] was developed by Unify Inc. (formerly Siemens Enterprise Communications), it allows the interworking and communication in hybrid, packet-switched and line switched network environments. [6] Another protocol that provides transmission and control to stream media through IP networks is the Inter-Asterisk eXchange (IAX) protocol. It is being used as a substitute for H.323 and SIP by Asterisk VoIP PBX to connect to devices supporting IAX.

## VI. MANAGEMENT GOALS

Telephony adoption in Pakistan is moving at a slow pace which encompasses a number of reasons, but the vital factor for the slow growth can be attributed to legacy phone systems reliability. The reason why businesses shift to IP is for the reduction of telecommunication costs and this change usually occurs as a result of a change agent for instance end-of-life for the phone systems. While the economy is weak, the IT industry would be happy to receive as much mileage as it can with the infrastructure that exists , and strong reasons for changing would be required if the legacy phones work fine. Certainly, the level of value as a result of Unified Communications and Voice over IP are major factors for adopting this technology, however hard they might be to quantify. Moreover, a vast number of decision makers have a mindset that is rather focused on hardware when measuring telephony's value. With the passage of time, it might change, but this is one of the main reasons why legacy systems are still enduring.

The paper will be focusing on the particular decision of implementing VoIP as well as the optimum techniques for measuring its various efficiencies. A certain objective needs to be established for implementing such activity based systems, after taking appropriate actions. The goals of management while carrying out an implementation of VoIP are listed below:

- Establishing effective processes for calculating overall costs of existing PBX networks as well as comparing them with the costs that would incur as a result of shifting to VoIP networks.
- Some vital segments that can be traced to the PBX network consist of the amount of employees engaged in long-distance calls, the charges for long distance calls, and the numbers of calls that are made annually that are long-distance in nature.
- Assisting activities that apply to the PBX long-distance calls involve yearly labor costs, hardware expenses, and the tax shield that is a result of a higher amount of capitalization costs via a conventional PBX network.
- The main segments that can be traced to VoIP networks are the annual costs per user for IP telephones and the amount of users.
- Assisting activities that are of application the VoIP installations involve the opportunity costs involved with additional depreciation for the earlier PBX systems as well as the assumptions that the obsolescence of technology will occur at a rapid pace in conventional networks.
- The goal of the analysis is computing the profitability and cost ramifications that are associated with discarding/replacing PBX systems for VoIP networks.[7]

To resolve these problems we have to follow few steps that include [8]:

- Identify Cost Drivers
- Keep the records of all activity and the time assign to each cost
- Implementation of strategic plan
- Calculate results and build up conclusion

## VII. ORGANIZATION "A"

This organization is using Panasonic PABX TDA 200 which now 5 year old having number of extension is 80 and number of line is 32. Now the company needs more number of extensions for their future requirement and they also need to replace their current PABX because it complete it's depreciate limit. Now keeping this scenario in front then we finds two options;

1. The option one is that they replace their existing PABX with new one with more extensions support
2. The option two is that they move toward IP Telephony solution and purchase IP based PBX that is not restricted to extensions, there is no resource pool and it is customizable to provide and extend extensions like that of computer IP and this option have rich features.

### A. *Financial Analysis*

- *Option 1: PABX*

Panasonic PABX = 1,267,300PKR

Panasonic Phone = 80,000 PKR

DP and Cabling = 300,000 PKR

Switches = 700,000 PKR

Total = 2,347,300 PKR

- *Option 2: IP Based PABX*

IP PABX = 300,000 PKR

IP Phones (Qty-152) = 1200,000 PKR

Switches (Qty-8) = 700,000 PKR

Total = 2,200,000PKR

### B. *Architecture*

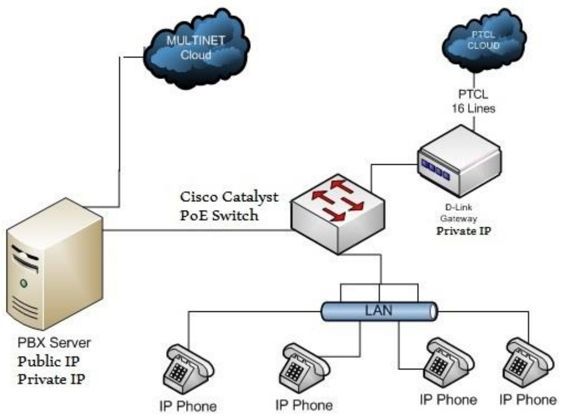


*Fig. 4. VoIP Architecture of Organization A.*

In figure 4 we found the architecture of our organization "A" in which fiber of data link is terminated into media converter and from media converter a Ethernet cable is terminated to LAN card of PBX server and from second LAN card of PBX server a Ethernet cable is terminated to switch where other users are also terminated. For backup purpose they terminated 16 lines of PTCL to FXO gateway that is then terminated to switch. Here they used Cisco catalyst switches are used here which possess PoE technology. This sent power to Ethernet to on IP phones.

### C. *Performance Analysis*

Below chart is the ratio of call that made by the organization after implementation of VoIP. They made approximately 3180 call out of which 3040 calls reached to its destination successfully without any congestion or dropping and only 140 call failed to reached its destination due to many other reason including congestion, delay etc.

- Number of Extensions = 152
- Number of Successful Calls = 3040
- Number of Unsuccessful Calls = 140

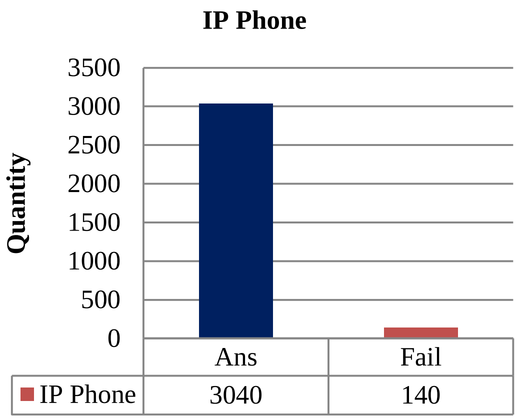


Fig. 5. IP phone versus pass/fail quantity

### D. *Cost Benefit Analysis*

- Dated PTCL Billing = 140,000 PKR per month
- 60% of billing comprise of long distance and Mobile Phone calls
- Current Billing per month with option 2 = 105,000 PKR
- So after implementation we save 35,000 PKR per month
- ROI = 2,200,000/35000/12 = 5.2 years.

## VIII. ORGANIZATION "B"

The organization is facing tremendous issues of downtime and noise using landline and they were getting poor support from their landline government sector. And the other problem is that they were getting costly bills due to frequent calls at outside the country side. Due to poor performance and high cost they decide to move IP Telephony.

### A. *Financial Analysis*

IP Based PABX

| | |
|---|---|
| IP PABX | = 300,000 PKR |
| IP Phones (Qty-550) | = 3,850,000 PKR |
| Voice Gateway | = 250,000 PKR |
| Cisco Call Manager setup | = 100,000 PKR |
| Total | = 4,500,000 PKR |

### B. *Architecture*

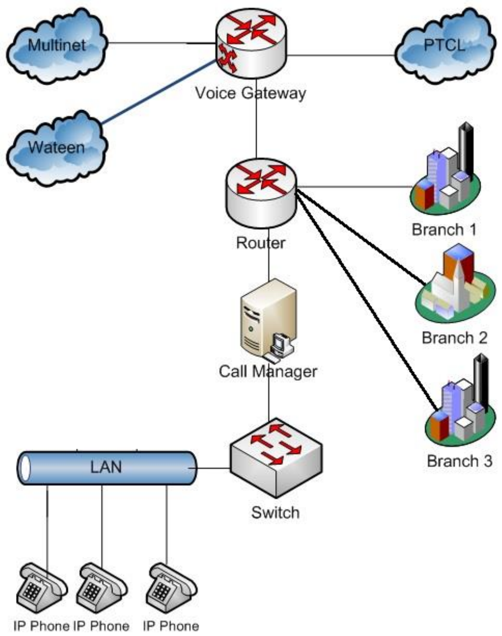


Fig. 6. VoIP Architecture diagram of Organization B

Above figure 6 is the architectural diagram of our Organization "B", in which two data link from two different vendors is terminated to voice gateway so that if one link will down they have another one as backup and will not face any down time. Few PTCL lines are also terminating into voice gateway. Then the voice gateway is connecting to a router so that they can connect other branches to each other and to head office and from router call manager is connect so that they can manage their VoIP infrastructure and then this call manager is connect to switch and LAN to provide interface to user to use IP Phones.

### C. *Performance Analysis*

Below chart is the ratio of call that made by the organization after implementation of VoIP. They made approximately 11,478 call out of which 10,252 calls reached to its destination successfully without any congestion or dropping and only

1,226 call failed to reached its destination due to many other reason including congestion, delay etc.

- Number of Extensions = 544
- Number of Successful Calls = 10,252
- Number of Unsuccessful Calls = 1,226

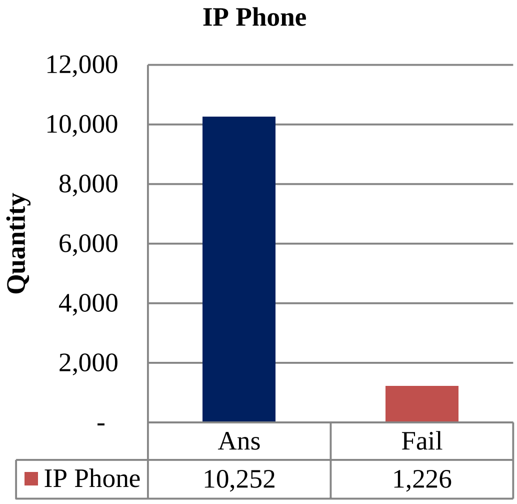


Fig. 7. IP phone versus pass/fail quantity

*D. Cost Benefit Analysis*

- Dated PTCL Billing = 501,000 PKR per month
- 40% of billing comprise of long distance and Mobile Phone calls
- Current Billing per month after implementation of VoIP = 375,800 PKR
- So after implementation we save 125,200 PKR per month
- ROI = 4,500,000/125200/12 = 3 years.

## CONCLUSION

Considering the fact that international phone call were steeply-prices, VoIP was designed, to allow people to be able to connect vocally across the globe. Imagine having to make a phone call in a country on the other side of the world. The cost of the call would be the first thing that would cross a person's mind. However, VoIP came up with solution for this, and other related problems.

When a new application is developed, or new software is put together, there are bound to be some shortcomings. However, the benefits that VoIP served totally outnumbered. In the above research paper, we have been looking up the advantages that VoIP serves and the numerous ways in which it has helped strengthened verbal interaction using phones. It is cheap, Using PSTN setup in Pakistan, it really costly, here time is money. We have to pay money for a single minute that we spend talking to others on phone. Making an international call is costly, and the caller adds to the bills with every passing minute. Because VoIP works using the web, the user only has to pay the internet expenses. What is important to note is that VoIP not only allows voice calls, but also a number of other services, such as the vibrant media offerings; the traditional telephones only offered voice and fax. And in a country like Pakistan, the demand is sky high.

Rich media service is made possible by VoIP technology as it integrates with applications and protocols. VoIP protocols (like H.323, Session Initiation Protocol [SIP]) work on the application level and have the capability to join or work in coordination with several other applications like web browser, social-networking applications, instant messenger and email etc. It can even be incorporated with the present operational conventional telephone system. Majority of the VoIP service suppliers offer an interface which can be controlled by the user, particularly GUI. This can help the customers modify options, service and features in a dynamic manner.

We investigate the issues facing by two organizations related to poor performance of traditional telephone setup and they were paying high cost. We first understand the scenario of these two organizations and then implement VoIP setup in these two organizations using best practices. After implementation they both organization gets good performance, rich features and saving cost. Research has revealed that as compared to the use of PSTN line, the use of VoIP can help the customer save approximately 90 % on long distance calls and 40 % on local calls.

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