ISSN XXXX XXXX © 2019 IJESC

Research Article Volume 9 Issue No.1

A Novel Approach for Estimating Bandwidth Controlled Topological Architecture for Distributed Hybrid Networks

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Abstract:
At present, all things have become very compact, in case of communication they can be defined by different attributes like digital documents, internet Medias, telemedicine's etc. People have become internet dependent these days. Now days communication over the internet like–Skype, messengers, g-talk are very familiar tools for chatting. Facebook is also an all-in-one communication media these days. But these software applications that most of the people use need more bandwidth. As a number of enormous internets using people are increasing day by day for that a proper way is needed to control and distribute bandwidth within a conjugated network nodes or network clouds. These days' people prefer video chatting for more secured and dedicated communication. To send videos as data packets consider the topologies such as- mesh, bus, ring, star, hybrid within the communication networks which uses unlimited bandwidth results a huge wastage but creates a problem for the higher growth of internet users. Thinking of this a controlled algorithm for bandwidth optimization as well as bandwidth management is developed. This algorithm is developed considering a video communication system that has the ability of running PABX, Fax, video via-web conference, video presentation for a defined time through the creation of admin defined extensions. The algorithm has developed considering a local LAN (no internet) and can process user defined data by checking the processing capacity of a processor that ensures 0% drop probability of packets which make it special. The overall mechanism shows better possibilities as an estimation scheme throughout the network nodes to ensure the administrators to design a proper and better network for any distributed network architectures. After evaluating the results it is assured that this topology works better than any other existing topologies for a distributed system that follows M/M/1 queuing theory.

Keywords: LAN, Wireless, Wi-Fi, VoIP, SIP, IAX2, Network Topology, Distributed Networks, Bandwidth(BW) Optimization, Traffic Engineering, M/M/1 Queuing, Time and Memory Complexity, Dhrystone, Graph Theory, Moore Graph.

I. INTRODUCTION

A video communicative system can be described as integrated real-time services shaped into a compact manner. Integration of communication services such as chat (instant messaging), IP telephony, data sharing (including peer-to-peer communication), call control and speech recognition comply with non-real time communication services like- voice-mail, e-mail, fax, SMS. It is not meant to be a single components but a series of components that dedicates a consistency of unified user interface and user experience across several media types and multiple devices. A video communicative system can have the components included like- instant messaging, chat, VoIP, Video over IP. In here a system model is constructed for video calling and conferencing, a video communicative system-that is itself referred to a unified communication system regarding the constructing criteria.

A video communicative system consists of one or more SIP phones, VoIP phones; the server optionally includes a gateway. It is very similar to a proxy server: SIP clients can be either soft phones or hardware based phones. The first process is to register with the video server and when anyone wishes to make a call one have to ask the video server to establish the connection. The video server has a directory of all phones or users and their corresponding SIP addresses. Users of the video server system share a number of lines for making external or internal phone calls. One of the latest tendencies in communicative server based phone system development is the Video PABX via conferencing, also known as video over IP PABX which uses the Internet Protocol to transmit calls. Now at the age of latest technology virtual and existing communicative server systems are filled through the communication market because of different stand-alone servers. With these systems people can communicate through ease sitting their home instead of large systems and going outside.

Thinking at the existing system a custom video server is built referred as an video communicative server for its construction criteria which is very easy to use, cheap in cost and also has different usability like-no internet access, optimized local LAN bandwidth, Video calling, video conferences through document uploading, communicating through windows phone or android smart phones and many more. This prescribed system is made especially for video calling and conferencing. In this video communicative system every user is assigned with a unique ID number. Using this ID number one user can communicate with others.

It uses the video standard H.273 codec for video calling, conferencing and also uses both protocols SIP and IAX2. A new algorithm is developed to optimize the bandwidth. With the help of the algorithm it ensures 0% drop probability of packets, for the link and end point also in case of the caller and called party. No matter what the packet size is, the delay is constant and it is 1 sec for a particular data interval time, provides an optimized process.

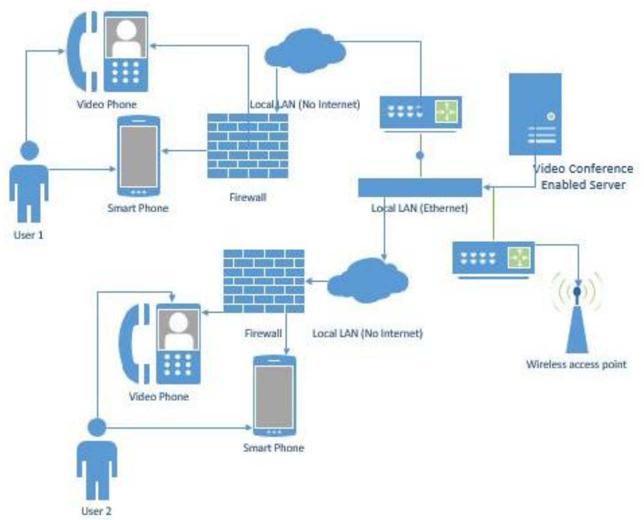


Figure. 1. Video Conferencing System (M/M/1)

In case of HD video the delay (sec) Vs Packets (number) won't be changed. The algorithm has developed considering a local LAN (no internet) and can process user defined data considering the processing capacity of a processor estimates 0% drop probability of data packets. The defined algorithm controlled the bandwidth by transferring data packets through measuring delay time. The full process shows better performance as an estimation scheme throughout the network nodes to ensure the system administrators to design proper and better network node architecture by measuring the best efficiency for video calling and conferencing since it follows M/M/1 queuing theory.

II. ARCHITECTURE OF VCS (VIDEO COMMUNICATIVE SYSTEM)

A video communication system is used to allow users to access voice, e-mail and other mixed media from a single mailbox that is independent of the access device. The system also includes messages of mixed media types such sound clips, pictures, and communication via short message services (SMS) [1].

Collaboration and interaction of systems redirects a new way on applications such as calendaring, scheduling, workflow, integrated voice response (IVR), and other enterprise applications that helps individuals and workgroups to communicate efficiently. Real-time and near real-time communications systems redirects a new dimension of fundamental communication between individuals using applications or systems such as conferencing, instant messaging, next-generation private branch exchanges (PABX), and paging [2].

A. Components of VCS (Video communicative system)

- Communications: Voice, data, and video
- Messaging: Voice, email, video, and IM
- Conferencing: Online, offline, audio, and video
- Presence: IP phone, desktop clients, and call connectors
- Common user experience: Desktop, phone, and mobility

A. Features of VCS (Video communicative system)

- Movement toward Software Approaches
- Distance from hardware-centric products
- Service Oriented Architecture
- Increasing Web Services
- Demanding Mobility
- Always Connected
- Faster and Quick Response
- Increased Competition

III. LAN (LOCAL AREA NETWORK)

A local area network (LAN) is a computer network that interconnects computers within a limited area such as a residence, school, laboratory, office, departments or supermarkets. A local area network is built in principle to a wide area network (WAN), which can cover a larger geographic distance and may involve with telecommunication leased circuitry materials. Ethernet over twisted pair cabling and Wi-Fi are the two most common transmission technologies that are used for local area networks.

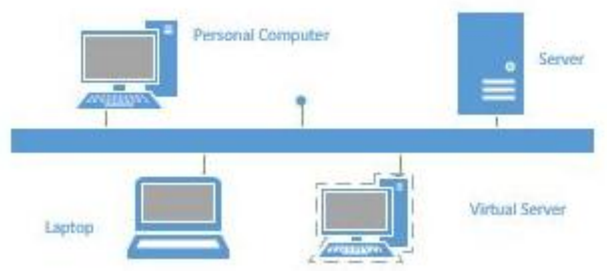


Figure.2. a conceptual diagram of a local area network using 10 Base 5 Ethernet

*A. **Wireless***

It is the communication in which information can be transferred between two or more points without any wire communications. Wireless can work in larger distance communication offering radio frequency to connect with television or other deep-scale electronic components. It can define different type of systems like- mobility, mobile, fixed length personal digital assistants (PDAs) and wireless networking.

*A. **Wi-Fi***

Wi-fi or wireless fidelity, allows the access of the internet while on the move or running environment; one person can simply remain online while moving from one area to another residing in a wireless mode. Wi-fi enabled computers send and receive data's from inside and outside anywhere on a location. But must be resided within the range of a base station. It is just similar as a cable modem connection. Wi-fi is a generic term that redirects to the IEEE 802.11 communications standard for wireless local area networks (WLANs).

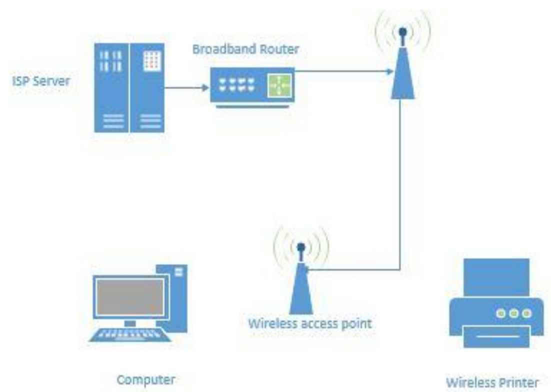


Figure.3.A printer prints a paper connected by a computer wirelessly (both are in same LAN)

IV. VIDEO OVER IP

Video over IP is a trend of communication property which works via the internet protocol. The system uses the standard video codec mechanism to shorten the program components into a bit stream and then transports it over the network to carry that bit stream which is encapsulated in a data flow of IP packets [3]. This is typically completed using some properties of the RTP protocol. Most of the time uses the existing video codec for encapsulate the data's. Carrying the videos over IP within a conjugated network do have some special challenges compared to the most non-time-critical IP traffic. Many of these problems can be characterize to those problems that are encountered in a voice over IP, in case of video needs a much higher level of engineering requirements [4]. A must needed property is the assurance of quality of service (QoS) for professional broadcasting.

V. BANDWIDTH (BW) OPTIMIZATION

Bandwidth optimization is a technique that can measure and utilize the capacitive bandwidth within the whole system. Now a days bandwidth management is a problem as the number of internet users are increased day by day. Even using local LAN, a limit of bandwidth must be controlled among all the users those are provided through the PBX system, or the bandwidth will be wasted among the caller and called party and also may be wasted between the link and the end point. By thinking of it an algorithm is established.

It is a controlled leaky Bucket algorithm with some specific features. This algorithm shapes the packets between the caller and the called party also between the link and the end point of the throughout system. The packets are shaped with the help of specific time buffer. The buckets are not fixed as normally as leaky algorithm uses. Increase number of video calls and number of packets the time delay is decreased and the data's are optimized. The discarded packets from the overflow data's are saved in an optimal bucket until the process of the previous step [5]. Then the data's are sent sequentially. Overall Process ensures an optimization scheme within the whole network area.

VI. TRAFFIC ENGINEERING

Traffic engineering is a trend of finding possible ways to optimize the performance and usability of data's within a communication network through performing analysis, measurement, prediction and regulation of data's random or dynamic behavioral activities transmitted over the conjugated network [6]. This technique is an enormous way to optimize or utilize the data's among different networks like PSTN, LAN, WAN, MPLS, VLSM LAN's etc.

A few but very popular algorithms that categorizes traffic engineering used for shaping bursts, data sizes and data queues they are given below: 1) Leaky Bucket and 2) Token bucket. Thinking all the possible ways a technique is implemented that have least complexity than the individuals with the help of mixing traffic shaping, scheduling and congestion avoidance in the field of traffic engineering.

The process ensures an optimized scheme over the whole network. Here a modified leaky bucket, modified FIFO scheduling and modified policy making is used that shapes the burst size over the network nodes [7].

VII. M/M/1 QUEUING

If in a service station, customers arrive at a single-server in accordance with a Poisson process having rate λ. That is, the times between successive arrivals are independent exponential random variables having mean $1/\lambda$. Each customer, upon arrival, goes directly into service if the server is free and, if not, the customer joins the queue. When the server finishes serving a customer, the customer leaves the system, and the next customer in line, if there is any, enters service [8]. The successive service times are assumed to be independent exponential random variables having mean $1/\mu$, then the preceding is called the M/M/1queue.

Properties:

- Arrival rate is a Poisson distribution, a memory less property
- Arrival rate of customer is unlimited
- Service time is an exponential distribution, is a Poisson model
- All arrivals are waited to be served at a constant time interval, steady-state property
- The λ is a constant, Sometimes, $\mu > \lambda$ (average service rate > average arrival rate)
- A single server serves customers one at a specific time from the first of the queue, according to FCFS discipline. When the service is complete the customer leaves the queue and the number of customers in the system shortens by one. Buffers are of infinite in size, Number of customers that can be contained is limitless.

In our proposed method we define customers as data packets and modified the buffer size as specific within a time interval. Overall process itself contains a memory less property scheme.

VIII. TIME AND MEMORY COMPLEXITY

Since a construction of the algorithm is made to shape the time interval on an average, so the time is constant, and the complexity is O (1).

This means that the algorithm requires the same fixed number of steps regardless of the size of the task. The algorithm uses insert and remove operations for a queue [9]. Space (memory) complexity is measured by using polynomial amounts of memory, with an infinite amount of time.

The difference between space complexity and time complexity is that space (memory) can be reused. Ex-Amount of computer memory required during the program execution, as a function of the input size. But we fix the time interval as an average using the CPU benchmark synthetic properties. The property uses Dhrystone benchmark, which counts the program execution process in MIPS (million instructions per second). This occupies whatever the data size is, the data size composed of several packets will end up the queue using FIFO property. All the process occupies 1 sec to complete within the server. If packet overflows for larger assumptions it will peak 2 sec, but the difference will be the same 1 sec in every aspect and there occupies no loss of packets [10]. For the results we have checked several algorithms and proved that our algorithm performs better. So it is a nondeterministic time memory that uses O (n) space.

The algorithm is tested on core-i7 4220 3.2 Ghz processor ---

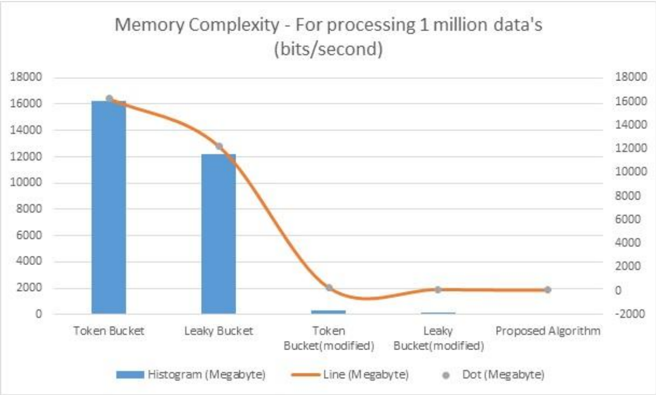


Figure. 4. Memory complexity

IX. DISTRIBUTED NETWORKS

A distributed computing architecture is typically a set of processors which are interconnected by a communication network where each processor has its own local memory and other peripheral components and the communication generates between any two processors of the system through passing messages over the communication network medium. A distributed network refers as a type of network that follows the network node architecture of a distributed system. In this proposed analogy distributed architecture is built to ensure the upmost usage of the system. For a distributed system it is much beneficial to add more nodes or clients in a network to perform more to serve more users [11]. If a system interruption occurs within an area other areas won't affect cause of this distribution architecture. The architecture itself having less cost to build and also can construct easily if a failover occurs.

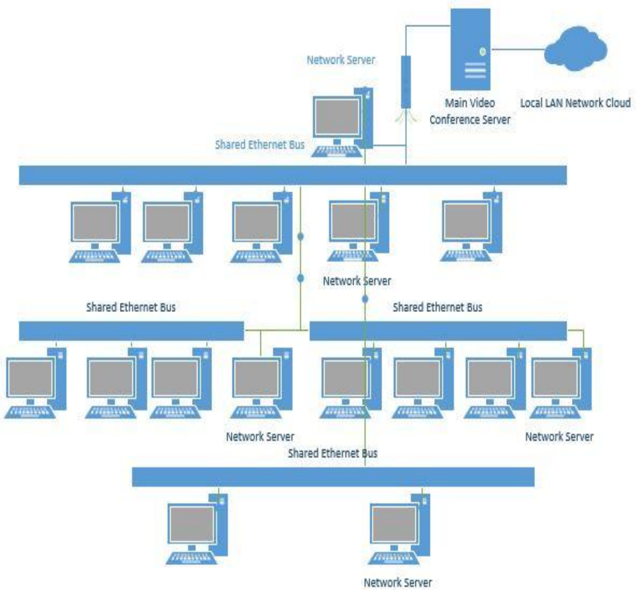


Figure. 5. Distributed System Networks (Minicomputer model architecture)

X. NETWORK TOPOLOGY

Topology is a trend of architectural way to mount the network nodes within an efficient and favorable manner. By constructing proper topology for a network architecture estimates the best measurement throughout the network length and also attain maximum throughput. In network topology, there are different kinds of topology or architecture that uses for network construction. But mainly there are seven topologies that can be defined for unique criteria's.

The topologies are –
Point to point, Bus, Hybrid, Ring, Mesh, Tree, Star

A. *Best topology criteria*

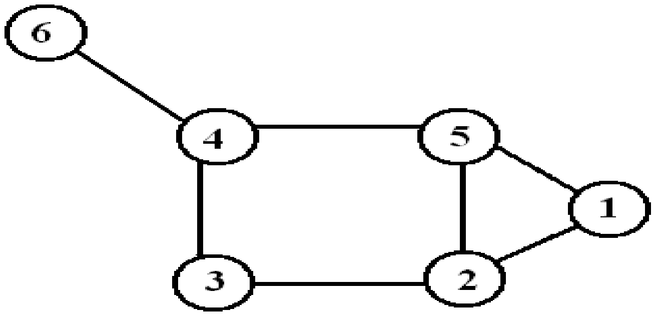
- Maximum attainable throughput
- Minimum average path length
- Maximum number of nodes.

B. *Topology choosing factors*

- Type and number of computers and peripheral devices being used
- The anticipated speed of data transfer
- Types of application running on network
- The required response time of the network
- Network cost

XI. GRAPH THEORY

A graph is a collection of nodes and edges which can be denoted by G = (V, E), where
V = nodes (vertices, points).
E = edges (links, arcs) between pairs of nodes.
Degree is an inherent property denoted by K
Degree of a node = the number of its connections
The graph size parameters are: $n = |V|$, $m = |E|$.



Figur. 6. Graph nodes

In the above graph figure
N: = 6, m: =7

Vertices (V): = {1, 2, 3, 4, 5, 6}

Edge (E): = {1,2},{1,5},{2,3},{2,5},{3,4},{4,5},{4,6}}

N (4): = Neighborhood (4) = {6, 5, 3}

The originality of graph theory can be described through Euler's work in case of the Konigsberg bridges problem (1735), which led him to the concept to establish an Eulerian graph. The further study of cycles among graphs described by polyhedra is stated on the work of Thomas P. Kirkman (1806 - 95) and William R. Hamilton (1805-65) which also led them on the concept to establish a Hamiltonian graph [12].
Graphs can be used to model pair wise relations between objects or nodes. Usually a network can be describes thoroughly by a graph in an efficient manner. Other real-life practical problems can easily be shown in terms of graph theory.

XII. MOORE GRAPH

Moore graph is a special kind of graph property in a graph theory that can describe node estimation also estimates the length of the edges between nodes and eliminates path redundancy [13]. In a more graph the minimum distance branching factor b and root b-1; the parameter g is odd, when

g=2d+1, d >=1

No vertex z where g contains two distinct z-x paths of length at most d. Since g has a cycle length at most 2d at least delta vertices at distance 1 from x, for second time at least delta (delta-1)

For a node, if g is odd

$$n_0(g,\delta) = 1 + \frac{\delta}{\delta-2}\{(\delta-1)^{\frac{g-1}{2}} - 1\}$$

If g is even

$$n_0(g,\delta) = \frac{2}{\delta-2}\{(\delta-1)^{\frac{g}{2}} - 1\}$$

A. *Moore network node estimation property*

By Metcalf's law, the value of a telecommunication network is proportional to the square of the number of users in the network.

If users are denoted by U and Value is denoted by V, then

$$V \propto U^2$$

If the number of connection is N then, for a point to point

$$N = n(n-1)/2$$

In moore network p, b are variables and g is the parameter where p and b are defined to investigate the maximum attainable performance of the best possible topology. From p, b and g we can calculate the value of S, a, c and N. The number of switches is N/A where N is fixed. So for reducing the number of switches need maximization of the value of a.
To calculate S, parameters are needed .The parameters are N, a, c and g. By getting the value of N, a, c and g ; S can be calculated.

$$N = Sa$$
$$S = N/a$$

For the number of ports of the servers

$$a + b = c$$

In more graph for a single-source minimum distance spanning tree has a branching factor of b at the root and (b-1) at every other nodes.
For a fully populated number of nodes,

$$S = (1 + b\sum_{i=0}^{p}(b-1)^i)$$

For geometric sum,

$$S = (b-1)^p - 1/b - 2$$

For b>2,

$$S = b(b-1)^p - 2/b - 2$$

The above equation is defined as moore limit for an optimal topology construction.Considering no use of path redundancy the bandwidth of average path length times for single server,

$$B = g\bar{p}a$$

The average path length is related to the attainable throughput and average data packet latency from the fully populated node equation considering the geometric sum and triangular sum mixing both and rewriting them as double sum, the equation stated,The average path length value divided by the number of servers,

$$\bar{p} = \frac{b(b-1)^p(p-\frac{1}{b-2}) + \frac{b}{b-2}}{b(b-1)^p - 2}$$

$$a = g.\frac{b(b-1)^p - 2}{(b-1)^p(p-\frac{1}{b-2}) + \frac{1}{b-2}}$$

The parameters that are defined in case of proposed perspective

c = Trivial relationship between server ports.

g = Range of the network nodes(area).

N = The number of connections that are made.

S = Number of servers.

a = Relationship between the server ports on first occurrence.

Network cost efficiency , E=a/c

Considering all the merits, demerits and criteria of the above stated topology an estimation of an architecture is found. The network nodes matches with it composes the best topology for a network architecture to establish video calling and conferencing. The topology works against path redundancy, expensive cost and also dedicated nodes. This is a hybrid topology mixed with both star and bus criterias.It also supports M/M/1 queuing theory as our algorithm is constructed for a single server distributed networks as well as it is the best topology to serve more users to distribute communication among peoples in a distributed network.Star-bus topology is chosen because if node servers fails the main server bus remains intact and distribute data's among users; in case bus server fails then each node server can distribute data's among each node that occupies each user. More servers reduces the network performance and also may cause startup errors. The proposed hybrid network topology:

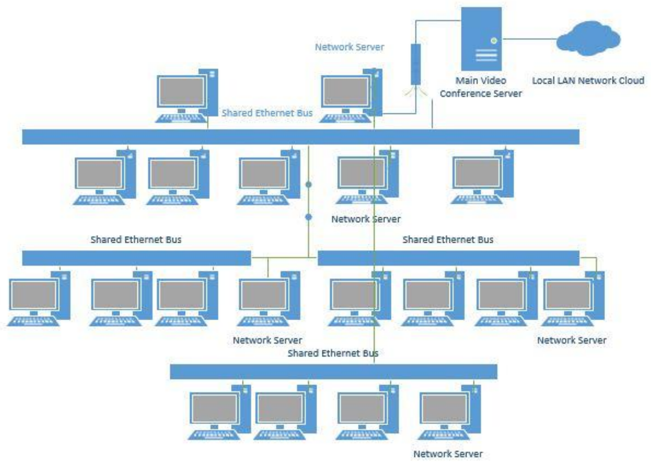


Figure.7. Proposed Star-Bus hybrid (4x4=16 network nodes, where N=4, S=1, a=4) Topology for real time (video conference) high-speed data transfer following M/M/1 queuing theory

Table.1. Distributed perspective among three attributes

Mechanism	Server 1	Server 2	Server 3	Server 4
Average throughput(BW)	12000 b/s	10000 b/s	9000 b/s	8000 b/s
Time Interval(Average Waiting Time)	1s	2s	3s	5s
Memory Usage(Video Processing)	0.25%	0.12%	0.8%	0.6%

XIII. CONCLUSION

It is always a challenging task to establish network architecture. Designing a proper architecture may ensure service integration much more than previous times or if improper architecture established that may lead to high service disruption and also lead an enterprise to a huge data loss. This topology is mainly based on client-server distributed architecture. After comparing the above calculations it can be said that it is good, efficient and better administrative topological architecture for real-time data processing in a distributed environment. It can also be constructed as an enterprise level system. Moreover, the topology redirects a new way towards the network communication and device junction technology. Though the network topology consumes mid-level cost among the all basic topologies but ensures better analytical performance architecture for distributed network communication.

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