A Voice/Data Integration MAC Protocol for Ku Band CDMA VSAT Networks

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Abstract- **A medium access control (MAC) protocol that integrates both voice and data traffic in TDM/CDMA VSAT networks is proposed and investigated. Architecture of the VSAT network is described. For voice traffic, speech activity detector (SAD) is used, and for data traffic, WWW browsing traffic is considered. Dynamic code allocation strategy with QoS consideration is the key feature of this MAC protocol. Given a VSAT, data packets can be transmitted in the silence period of voice traffic. If talkspurt continues, the channel should be obliged to transmit voice instead of data. Simulation results are given for pure voice call and voice/data integration traffic.**

I. INTRODUCTION

Very-small-aperture-terminals (VSATs) can be defined as a class of small, intelligent satellite earth stations suitable for easy on-premise installation and usually operating in conjunction with a larger hub earth station acting as a network management center. These can be used as nodes of a satellite network supporting a wide range of two-way integrated services. A number of recent innovations that have resulted in a decrease of the station size and cost make VSAT system more attractive service providers in the future [1,2].

In VSAT network, the link from the hub station to the VSATs (outbound) is usually configured using conventional time division multiplexing (TDM). However the multiple access link from the VSATs to the hub (inbound) has been subjected to a greater degree of variation. Debate continues about the most suitable and efficient choice for particular traffic demands. Generally, there are several candidates for multiple access of inbound links, including CDMA. For the CDMA, each VSAT channel is assigned a unique PN address code used to spread the spectrum of transmitted inbound messages. The advantages of using CDMA as candidate multiple access strategy of inbound links have been described in [1,3]. Based on these considerations, the network architecture is in a hub and spoke topology. The outbound links utilize a TDM format to simplify the receiver design of VSAT, and the inbound links operate with CDMA. For VSAT networks, an immediate consequence of this coding separation is that a separate receiver tuned to each spreading code is required in the CDMA hub station for each possible transmitter in the network. If the total number of potential VSATs in the network is much larger than the number of active terminals at any given time, this requirement can introduce considerable complexity into the design of a CDMA system. This calls for a suitable medium access control protocols to provide dynamic code assignment.

Very few people studied implementation and performance of CDMA VSAT MAC protocol. [1] studied how to integrate multi-service for next generation VSAT network. But how to design MAC protocol for future CDMA VSAT network is not mentioned. [3] proposed an implementation structure for C band TDM/CDMA VSAT system. But SAD is not used for voice traffic and dynamic code assignment method is not mentioned. What's more, system performance is not studied.

This paper proposes a code reservation based MAC protocol to support both voice and data traffics in VSAT networks. A voice terminal should transmit only during talkspurts by means of an activity detection scheme. Moreover, in order to increase the resource exploitation of VSATs, it is crucial that the bandwidth unused by voice traffic be assigned to data traffic that has not stringent delay requirements.

This paper is organized as follows: Section II gives a general description of our VSAT networks and traffic model. Section III proposed reservation based voice/data integration MAC protocol in VSAT networks. Simulation results and analysis are shown in section IV. Conclusion is given in section V.

II. Voice/data integration VSAT networks

A. VSAT architecture

The network consists of a central hub station and hundreds of VSAT stations. At the user earth station, there are telephone or PBX, facsimile and computer as user terminals. The central station interfaces with a host computer, telephone lines as shown in Fig.1. Using this network, various data communication and dialed telephone communication can be completed. The system operated at the *Ku* band. Outband only includes one channel which transmits either voice or data signal over the same carrier. These synchronous TDM signals are coded and spread by PN code to form a BPSK RF signal. The data to be sent to VSATs is asynchronously time division multiplexed into data stream with 153.6 kbps. It is also via FEC and PN spreading to form a BPSK RF signal. The PN codes for vocoder and data are of the same..

The inbound consists of a lot of CDMA signals from VSATs. Each VSAT signal also comes from one channel, i.e. either vocoder channel for 9.6 kbps vocoder transmission or a TDM data channel for 9.6 kpbs data transmission. After FEC and orthogonal PN spreading, the BPSK RF signals are sent out.

In receiving, for each VSAT, the vodcoder signals or data flows are selected after dispreading, demodulation, decoding and de-multiplexing. Then one vocoder is fed to telephone, or the desired data packets are fed to data terminals via proper interfaces. For the central station, every VSAT's vocoder or data are separatedly despread, demodulated, decoded and then fed into the circuit switching and the packet switching. After that, it is fed into PSTN, host computer or other equipment.

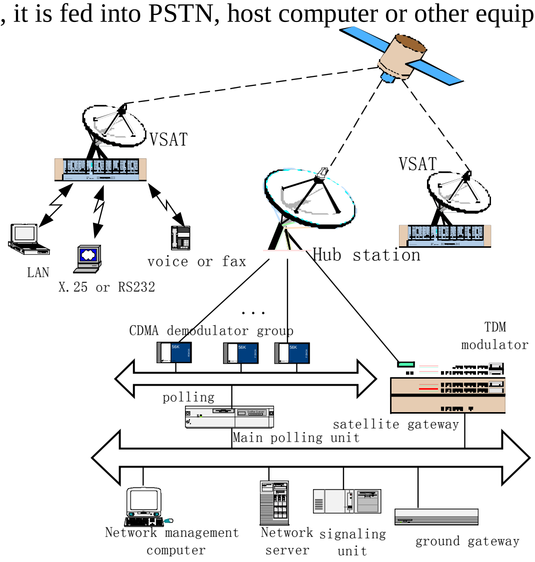

Fig.1. Architecture of VSAT networks

B. *Voice/ Data traffic source model*

A low-bit-rate vocoder with speech activity detection (SAD) will help to make a more efficient use of the channel. [1] discussed reusing speech silences for data transmission.

If SAD is used for voice traffic, talkspurts and silent phases are assumed exponentially distributed with mean values t_1=1 s and t_2=1.35 s, respectively. The voice activity factor is $\psi_v=(t_1/(t_1+t_2))\approx 0.425$. The voice traffic can be described by the σ_v (probability that a silence gap ends in T_f) and γ_f (probability that a talkspurt ends during T_f). Their expressions can be founded in [4].

Data can be divided into signaling data, SNMP data and traffic data. Signaling data carries information of voice signaling, access request, release notification, etc. SNMP data is used to transmit network management information. Traffic data includes LAN, X.25, RS232 data, etc. If a VSAT wants to send data, a peer-to-peer connection should be established before packets are sent out. Here, WWW browsing traffic model is considered. It is assumed the traffic messages arrive after transmission request is acknowledged by hub station.

Let L_d ($L_{d,bit}$) denote the mean message length in *packets* (*bits*, information part). We have

$$L_{d,bit} = L_d L_{pkt}\left[\frac{bits}{msg}\right] \qquad (1)$$

The data performance of the system is measured by average packet delay and throughput in which delay means the mean time from the packet arrival to the buffer to the instant when this packet is correctly received, and throughput denotes the mean number of correctly received packet each frame duration.

A VSAT generating WWW browsing traffic produces packets calls separated by a reading time during a browsing session [5,6]. A geometrically distributed number of datagrams with mean N_{pc} is generated per packet call and the datagram inter-arrival time is exponentially distributed with mean rate $\mu_{pkt.}$ The reading time is exponentially distributed with mean rate μ_{rd}. VSAT data traffic active factor[1] φ_w is

$$\varphi_w=(N_{pc}/\mu_{pkt})/((N_{pc}/\mu_{pkt})+1/\mu_{rd})=25/(25+8q) \qquad (2)$$

The mean datagram arrival rate is $\lambda=\mu_{pkt}\varphi_{w.}$ The datagram length has the same truncated and discrete Pareto distribution is shown in [5].

III. *Voice/data integration MAC protocol*

Different classes of traffic have different performance requirements. It is therefore necessary to develop an access scheme that accommodates all traffic classes in an efficient way. One way of treating this problem is to develop a combined random-access/channel reservation protocol and try to make it adaptive to a changing traffic mix. For Stream-type traffic (voice calls, file transfer) usually require a collision-free allocated channel, while random-access transmission could work well with small data messages.

A. Frame design

Suppose an active VSAT generates one packet per frame. Each packet encompasses $L_{pkt}=R_sT_f$ information bits. There are $N=T_f/T_s$ slots per frame. Each access request can be transmitted in one slot, while information packets are transmitted frame by frame.

Voice and data traffic have different QoS requirement. Voice packets have stringent delay constraint but can tolerate certain loss while data packet is not sensitive to delay but cannot suffer any loss. For telephone transmissions, CCITT recommendation G.114 stipulates that the propagation time between subscribers must not exceed 400 *ms*. If we allow 30 ms for the sum of the propagation delays in the end networks, for a geostationary Earth orbit satellite with 278 ms propagation delay, frame durations T_f must satisfy $T_f \leq (400-278-30)/2 = 46ms\cdot$

Considering that smaller packets reduce the throughput due to constant overheads for small and large packets, there should be some compromise between satisfying the QoS requirement and maximizing the throughput. In this system, we choose T_f=26*ms*. The other parameters are listed in Table 1.

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[1] The burstiness degree is the peak-to-mean traffic ratio. A WWW browsing VSAT has a burstiness $Bw=1/\varphi_w$; the traffic burstiness of a WWW browsing increase with q.

Table 1. Some fixed parameters in simulation

Variable	Value
Processing gain (G)	31
Total codes in code pool (C)	33
Channel rate (R)	9.6 Kbps
Frame length (L_{pkt})	255 bits
Frame duration (T_f)	26 ms
Slot duration （T_s）	2.6 ms
Mean values of talkspurts (t_1)	1 s
Mean values of silence (t_2)	1.35 s
Mean call duration ($T_{v\tau}$)	100 s
Mean message length ($L_{d,bit}$)	2550 bits/msg
Round trip delay Time(τ_{rd})	278 ms

From Table 1, we can see that each round trip consists of 108 slots and each frame consists of 10 slots. In each slot of access channel, there would be 15 information bits transmitted as access request, which is enough to identify each VSAT. When voice/data comes, its access request would randomly choose one of the 10 slots in next frame for transmitting. When collision occurs in access channel, retransmission (maximum three times) is conducted.

B. Traffic control and interference suppression

CDMA is an interference-limited system, so throughput will decrease if there are too much simultaneous users. Assume:

- Power control is perfect. And radio channels are ideal without fading, shadowing or multipath.
- The synchronization of the packets is made by marking the time slots on a separate broadcast channel.

For example, Gaussian Approximation is applied to determine the bit error probability, for (255,131,18) BCH code data, with processing gain of G=31, E_b/N_0=3dB, the packet error probability is obtained in Fig.2. When the traffic is heavy, packet success probability will decrease greatly due to too much MAI. Proper choice of the maximum codes limitation L has significant influence on the packet error probability and packet error probability is an important factor that influences the system throughput. Another performance issue is access channel delay. Considering the long round trip satellite propagation delay (278 ms), the main requirement of the access channel is to provide high success probability of the first access attempt so that the call setup time is minimized.

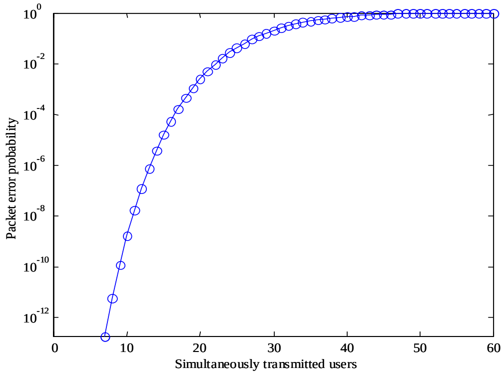


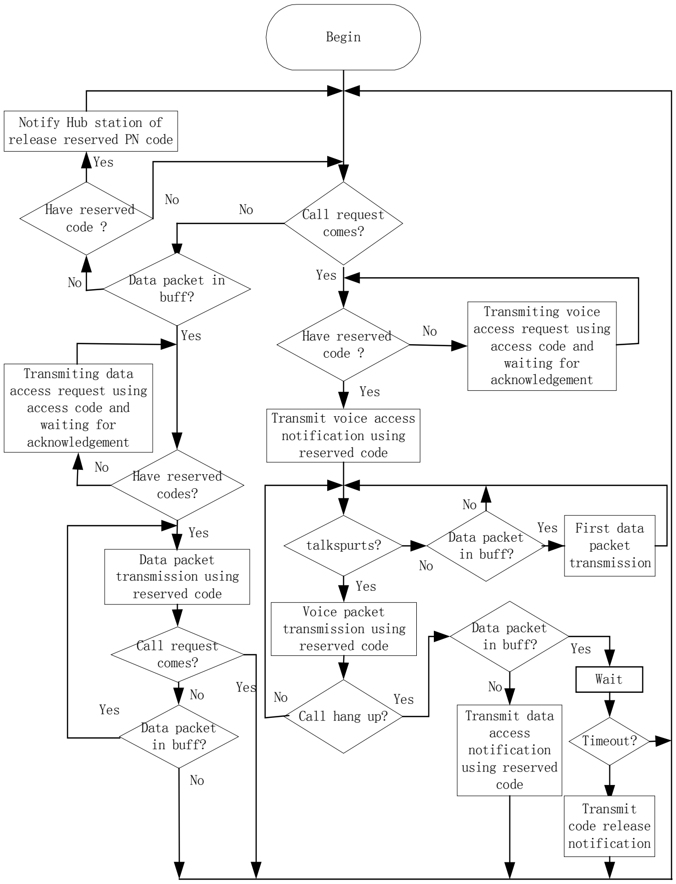
Fig. 2. Relation between packet error probability and simultaneous transmitted users in AWGN with E_b/N_o=3dB

C. Medium Access Protocol

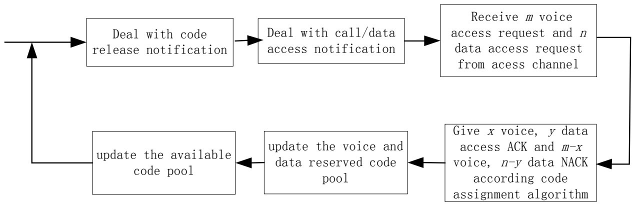
Ku band portable VSAT is a power limited system and only one branch of signal (either voice or data) is allowed to transmit at the same time due to power constraint. To meet the QoS requirement of voice traffic, we propose only voice traffic can be served if both voice and data traffic coming simultaneously. So, if voice comes, we transmit voice packet no matter whether data traffic exist. That is, if data packet is in transmitting, we will interrupt data traffic and transmit voice using this channel. Data traffic can only be transmitted in absence of voice traffic or during voice silence period.

Total code set $\{S\}$ can be divided into reserved code pool $\{S_r\}$ and available code pool $\{S_a\}$. $\{S_r\}$ is further divided into reserved voice code pool $\{S_{rv}\}$ and reserved data pool $\{S_{rd}\}$.

The code assignment method of voice and data traffic is illustrated in Fig.3. From Fig.3(a), if traffic comes and no code is reserved for this VSAT, reservation request will be sent out to hub station to get the assigned code. Meanwhile, if all type of traffic ends, the reserved code has to be released so that this code can be used by other VSATs. For all kinds of traffic, access PN code can't be used if the VSAT has special reserved PN code. This strategy can reduce the collision of access channel.



(a) VSAT flowchart each slot



(b) HUB flowchart each slot

Fig.3 VSAT and its hub station flowchart in MAC layer

From Fig.3(b), the hub station polls the received code-release notification and updates the received code from reservation pool to available pool every slot. And when the hub station received the code access notification which means call comes during data transmission period, it moves the code from $\{S_{rd}\}$ to $\{S_{rv}\}$. At the end of each slot, the hub will gather all successfully received voice/data code request and give ACK/NAK to voice request first. After that, acknowledgements for data requests will be sent to corresponding VSATs if there are still codes available.

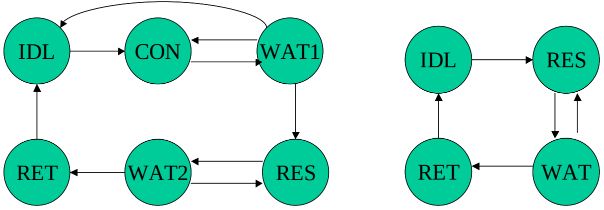
Suppose there are r codes in reservation pool in which s demodulators are receiving traffic, the decisions is based on the following rules in Fig.4.

```
    while (Current slot ends) {
            m voice and n data request successfully received;
            if $m+n+s<L$
              $x=m, y=n$;
            elseif $m+n+s>L$ and $m+s<L$
              $x=m, y=L-s-m$;
            elseif $m+s>L$
              $x=L-s, y=0$;
            end;
            update the code pool;
}
```

Fig.4. Code assignment rules in hub station

In this MAC protocol, state transition of a VSAT that initiates a voice call to other VSATs or PSTN is shown in Fig.5(a). Fig.5(a) can also illustrate the state transition of VSAT that positively initiate or passively receive a data connection. State transition of a VSAT that receive a call from other VSAT or PSTN is shown in Fig.5(b). The main difference of these two cases is that in the latter case, hub station has already assigned a available PN code for the passive VSAT station as well as the slot for voice call and tell this VSAT through voice signaling message in downlink; but in the former case, VSATs have to apply for PN code themselves.

In Fig.5(a), a VSAT is in *idle* (IDL) state if no voice/data traffic comes. After voice/data comes, it goes into *contend* (CON) state to apply for traffic code using signaling code. Then it waits for the acknowledgement (WAT1) before timeout. If NACK comes or WAT1 is still timeout after 3 attempts, it will go back to IDL state. If ACK comes, it has a unique traffic code to transmit its traffic. Then it enters *reserve* (RES) state. After traffic ends, the VSATs is in WAT2 state waiting so that if further traffic comes, it doesn't need to apply for PN code another turn. Notes that WAT1 has no traffic PN code while WAT2 has its special code. If WAT2 is timeout, then this VSAT has to return its PN code (RET state) so that this code can be used by other VSAT. In Fig.5 (b), when the passive VSAT is called by other VSAT or PSTN, the signaling message has already contained PN code to be used. So the VSAT would directly go to RES state.



(a) active call (b) passive call

Fig.5. State diagram of a VSAT

IV. COMPARISONS AND SIMULATION RESULTS

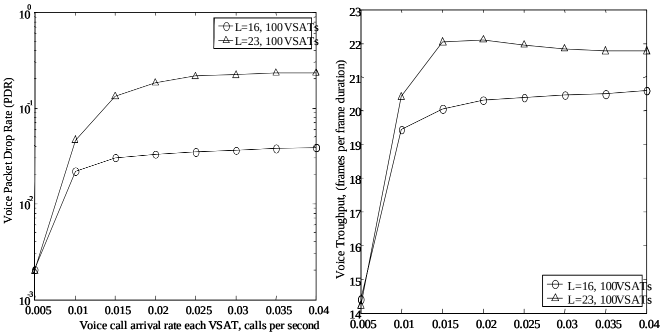
First, we simulated the performance of voice traffic in this VSAT system. Then WWW data traffic is added. Each simulation has been very long (1000000 slots) and repeated ten times to obtain reliable results.

A. Performance of Voice calls in absence of data

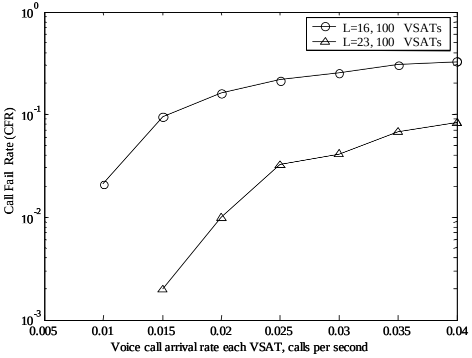
Assume that the arrival process of voice calls each VSAT is independent Poisson processes, with parameters λ_1. Suppose that the voice packet is exponentially distributed with means μ_1^{-1}. D_p is the sum of channel propagation delay, transmission delay of message in satellite channel and signaling processing delay. It is supposed that total call duration $T_{v\tau} = \mu_1^{-1} + D_p$ is 100s (see Tab.1).

Arriving calls are cleared after three times failure of access attempt. Let WAT1 timeout timer equal to roundtrip (τ_{rd}), and WAT2 timeout timer equal to $2\tau_{rd}$. We obtain simulation results of VSAT system for different maximum transmitted code limitation in Fig.6 using parameters in Tab. 1.

From Fig.6, with the increase of call arrival rate each VSAT, the Packet Drop Rate (*PDR)* rises. But when the voice Call Arrival Rate (*CAR*) is larger than 0.02, the increase rate of *PDR* isn't so sharp as that of smaller *CAR*. Due to SAD used, the number of VSATs that can be accessed to the system will be larger than 23 when some VSAT is in silence period. But when the VSATs in silence are back to talkspurts and transmit its traffic, the MAI would increase more than predetermined limitation in some cases. So from Fig.6(a), the *PDR* is far from allowable value (0.01) when *L* equals to 23 and *CAR* exceeds 0.015 calls per second. Fig.6(b) shows that when *CAR* is larger than 0.15, the throughput almost keep constant (around 22). That is the effect of our code limitation strategy, in which the simultaneously transmitted users always maintain under 23 and thus the *FER* is almost constant when traffic increase. Fig.6(c) shows that Call Failure Rate (*CFR*) also tends to increase as *CAR* becomes greater. But with the *CAR* ranging from 0.005 to 0.04, in general, *CFR* does not exceed the unacceptable limitation (usually 0.1).



(a) Packet drop rate (*PDR*) (b) Voice packet throughput



(c) Voice call failure rate (*CFR*)

Fig.6. Voice traffic performance vs. call arrival rate (calls per second)

B. Performance of Voice/Data integration

In voice call reservation period, the data packets can be sent during the detected silent intervals of voice call using the same PN code. When the talkspurts come back, this transmission interrupts until another turn of silence comes or voice/call hang up. WWW browsing traffic models [5] are adopted in simulation. The data performance is measured by the average packet delay and throughput. Packet delay is measured by the mean time from the instant when the message reaches data buffer to the instant when this message is completely sent out. Assuming that data packets can be generated only if it obtains the PN code successful, the data delay will be not influenced by heavy call block rate.

Performance of voice and WWW data integration VSAT is shown in Tab. 2 given 100 VSATs provided. Simulation parameters are shown Tab. 1. In this table, For a VSAT, given a voice Call Arrival Rate (*CAR*), Data Generation Rate (*DGR*) and maximum code limitation *L*, we get voice/data performance of the VSAT network. Voice Throughput (*VT*, number of frames per frame duration), Voice packet Drop Rate (*VDR*), Call Failure Rate (*CFR*) and Data Throughput (*DT*, frames per frame durations), data delay (number of frame duration) are used to measure the performance.

Table 2. Performance of Voice/data integration VSAT

CAR	*q*	*L*	*DT*	*Delay*	*VT*	*VDR*	*CFR*
0.01	0.002	23	18.319	18.592	0.034	1.97E-04	0.140
0.01	0.0025	23	18.605	18.824	0.019	3.53E-04	0.344
0.01	0.003	23	15.975	23.034	0.027	7.38E-04	0.224
0.02	0.003	23	1.3903	11.084	0.056	4.02E-05	0.001
0.01	0.002	30	17.835	19.240	0.038	5.10E-03	0.096
0.01	0.002	32	13.225	28.602	0.038	1.47E-02	0.071

From Table 2, with data traffic increase (*q* from 0.002 to 0.003), the *DT* increase, then decreases. And data delay will increase. This means that retransmissions increase due to too much MAI. The *VDR* also has such trends. *CFR* may increase or decrease with data traffic increasing. And *VT* will change when *CFR* differs. When the *CAR* increases (from 0.01 to 0.02), *DT* decreases sharply. But the delay doesn't necessarily increase. The reason is that data access calls have less superiority than that of voice, and it has less opportunity to access the system.

The maximum code limitation (*L*) also has significant impact on system performance. When code limitation increases, more VSATs will be allowed to transmit its traffic simultaneously. Therefore, *CFR* will decrease and too much MAI will induce at the same time. Too much MAI causes decreases of data/voice throughput and increase of delay.

Proper choices of *L* should be made to guarantee the QoS requirement. In most case, *VDR* below 0.01 and *CFR* below 0.1 are necessary for voice calls. If both *VDR* and *CFR* requirement have to be satisfied, *L*=30 is best choices for *CAR*=0.01 and *q*=0.002 under parameters listed in Table 2.

V. CONCLUSION

A voice/data integration MAC protocol for CDMA VSAT network is proposed. Code limitation strategy can decrease the MAI level and frame error rate, which improves the system performance when the background noise is very high. But too tight *L* will increase the call fail rate under heavy voice traffic. So code limitation should be properly determined by channel property.

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